Please use the Itinerary Planner at https://ativ.me/ybe/Mobile App (download instructions at https://acousticalsociety.org/new-orleans-2025/for the up-to-date schedule of presentations

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TECHNICAL PROGRAM CALENDAR
188th Meeting of the Acoustical Society of America joint with the 25th International Congress on Acoustics
18-23 May 2025

Please refer to the Itinerary Planner/Mobile App for Updated Schedule

Monday Morning

7:00 1aAAa **Architectural Acoustics, Noise, ASA Committee on Standards, Practitioners and Industry:** Day of ASHRAE Part I - Emerging Trends in HVAC Noise and Noise Control. Galerie 3

7:55 1aAAb **Architectural Acoustics:** Acoustics of Sustainable Building Assemblies and More I. Galerie 2

7:55 1aAB **Animal Bioacoustics:** Progress on Bioacoustics of Fish I. Galerie 4

8:40 1aAO **Acoustical Oceanography:** Topics in Acoustical Oceanography I. Studio 7/8

8:35 1aBA **Biomedical Acoustics and Signal Processing in Acoustics:** Super Resolution Ultrasound Imaging I. Balcony K

8:00 1aCA **Computational Acoustics:** Computational Methods I. Studio 6

11:10 1aID **Interdisciplinary:** Plenary Lecture: The Past Has Ears at Notre-Dame: Acoustic Research at the Intersection of Virtual Reality, Cultural Heritage, and Experimental Archaeology. Bissonet/Carondelet

8:20 1aMU **Musical Acoustics:** General Topics in Musical Acoustics I. Galerie 6

8:00 1aPAa **Physical Acoustics and Biomedical Acoustics:** Acoustic Holograms and Wavefront Modulation Techniques. Balcony I

8:15 1aPAb **Physical Acoustics, Computational Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Viscothermal Effects in Phononic Crystals and Acoustic Metamaterials. Balcony J

8:00 1aPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session I (Poster Session). Studio Foyer

7:55 1aSP **Signal Processing in Acoustics and Acoustical Oceanography:** Acoustic Array Processing and Sound Field Reconstruction I. Galerie 1

7:40 1aUW **Underwater Acoustics:** General Topics in Underwater Acoustics. Studio 9/10

Monday Afternoon

1:00 1pAAa **Architectural Acoustics:** Acoustics of Sustainable Building Assemblies and More II. Galerie 2

12:55 1pAAb **Architectural Acoustics, Noise, ASA Committee on Standards, Practitioners and Industry:** Day of ASHRAE Part II - Sound Standards and Codes. Galerie 3

1:35 1pAB **Animal Bioacoustics:** Progress on Bioacoustics of Fish II. Galerie 4

12:55 1pAOa **Acoustical Oceanography:** Acoustical Oceanography at Deep Water Abrupt Topography I. Studio 7/8

3:20 1pAOb **Acoustical Oceanography:** Topics in Acoustical Oceanography II. Studio 7/8

1:40 1pBA **Biomedical Acoustics and Signal Processing in Acoustics:** Super Resolution Ultrasound Imaging II. Balcony K

1:15 1pEA **Engineering Acoustics, Biomedical Acoustics, and Physical Acoustics:** Acoustic Holography: Advances and Applications. Balcony N

1:20 1pMU **Musical Acoustics:** General Topics in Musical Acoustics II. Galerie 6

12:55 1pNS **Noise:** Assessment of Low-Frequency Sound in Noise Criteria. Galerie 5

1:00 1pPAa **Physical Acoustics and Biomedical Acoustics:** Acoustofluidics. Balcony I

1:00 1pPAb **Physical Acoustics and Education in Acoustics:** It’s Not Physics. Balcony J

12:55 1pPP **Psychological and Physiological Acoustics:** The Hartmann Effect: Bill Hartmann’s Influence on Monaural and Binaural Hearing Research. Salon F/G

1:00 1pSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics. Balcony M

1:20 1pSC **Speech Communication:** Speech Communication Poster Session. Studio Foyer

1:00 1pSPa **Signal Processing in Acoustics and Acoustical Oceanography:** Acoustic Array Processing and Sound Field Reconstruction II. Galerie 1

1:00 1pSPb **Signal Processing in Acoustics:** Signal Processing Potpourri I. Salon D

1:00 1pUW **Underwater Acoustics:** Measurements of Natural and Man-Made Underwater Sounds. Studio 9/10

Tuesday Morning

7:00 2aAAa **Architectural Acoustics, Noise, Psychological and Physiological Acoustics, and Speech Communication:** At the Intersection of Speech and Architecture I. Galerie 2

7:00 2aAAb **Architectural Acoustics:** Student Design Competition. Studio Foyer

7:55 2aAB **Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Distributed Acoustics Sensing (DAS) in Ocean Acoustics I. Galerie 4

7:20 2aAO **Acoustical Oceanography and Underwater Acoustics:** Acoustical Oceanography at Deep Water Abrupt Topography II. Studio 7/8

7:55 2aBAa **Biomedical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Double, Double, Toil and Trouble - Towards a Cavitation Dose I. Balcony L

8:35 2aBAb **Biomedical Acoustics, Computational Acoustics, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics:** Technological Developments and Emerging Biomarkers in Elasticity Imaging I. Balcony K

7:55 2aCA **Computational Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Computational Methods for Nonlinear Problems in Acoustics and Vibration. Studio 6

7:35 2aED **Education in Acoustics:** Acoustics Around the World - Part 1: Education Programs at Universities. Balcony N

11:10 2aID **Interdisciplinary:** Plenary Lecture: Acoustics and Wave Physics in Modern Applications of Ultrasound in Therapy. Bissonet/Carondelet

8:20 2aMU **Musical Acoustics:** Musical Instruments in Jazz I. Galerie 6

7:55 2aNS **Noise, Physical Acoustics, and Computational Acoustics:** From Boom to Zoom: Department of Defense and Noise I. Galerie 5

7:50 2aPAa **Physical Acoustics, Education in Acoustics, and Engineering Acoustics:** Celebrating Steven L. Garrett's Fifty Years of Contributions in Acoustics I. Balcony J

7:55 2aPAb **Physical Acoustics and Biomedical Acoustics:** Acoustic Radiation Force and Its Applications. Balcony I

8:00 2aPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Best Student Poster Award Session (Poster Session). Studio Foyer

7:55 2aSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials and Phononic Crystals I. Balcony M

9:00 2aSC **Speech Communication:** Speech Production Poster Session I. Studio Foyer

7:55 2aSP **Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics:** Machine Learning in Underwater Acoustics I. Galerie 1

7:00 2aUW **Underwater Acoustics, Acoustical Oceanography, Computational Acoustics, and Signal Processing in Acoustics:** Ambient Sound Measurements and Models I. Studio 9/10

Tuesday Afternoon

1:00 2pAAa **Architectural Acoustics, Noise, Psychological and Physiological Acoustics, and Speech Communication:** At the Intersection of Speech and Architecture II. Galerie 2

1:00 2pAAb **Architectural Acoustics, Noise, ASA Committee on Standards, Practitioners and Industry:** Day of ASHRAE Part III - Research, Education, Certification, and Remediation. Galerie 3

1:20 2pAB **Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Distributed Acoustic Sensing (DAS) in Ocean Acoustics II. Galerie 4

1:00 2pAO **Acoustical Oceanography and Underwater Acoustics:** Acoustical Oceanography at Deep Water Abrupt Topography III. Studio 7/8

1:00 2pBAa **Biomedical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Double, Double, Toil and Trouble - Towards a Cavitation Dose II. Balcony L

1:20 2pBAb **Biomedical Acoustics, Computational Acoustics, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics:** Technological Developments and Emerging Biomarkers in Elasticity Imaging II. Balcony K

1:00 2pCA **Computational Acoustics:** Computational Methods II. Studio 6

1:00 2pEA **Engineering Acoustics:** Acoustic Transducers and Sensors. Balcony N

2:00 2pMUa **Musical Acoustics:** Musical Instruments in Jazz II. Galerie 6

5:00 2pMUb **Musical Acoustics:** Musical Instruments in Jazz III - Concert. Galerie 6

1:35 2pNS **Noise, Physical Acoustics, and Computational Acoustics:** From Boom to Zoom: Department of Defense and Noise II. Galerie 5

1:00 2pPAa **Physical Acoustics, Education in Acoustics, and Engineering Acoustics:** Celebrating Steven L. Garrett's Fifty Years of Contributions in Acoustics II. Balcony J

12:55 2pPAb **Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics:** Acoustic Manipulations of Objects: Theories and Applications. Balcony I

12:55 2pPP **Psychological and Physiological Acoustics:** Auditory Cognition in Interactive Virtual Environments. Salon F/G

12:55 2pSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials and Phononic Crystals II. Balcony M

1:20 2pSC **Speech Communication:** Speech Perception Poster Session I. Studio Foyer

1:00 2pSPa **Signal Processing in Acoustics and Acoustical Oceanography:** Acoustic Array Processing and Sound Field Reconstruction III. Galerie 1

1:00 2pSPb **Signal Processing in Acoustics:** Signal Processing Potpourri II. Salon D

1:00 2pUW **Underwater Acoustics, Acoustical Oceanography, Computational Acoustics, and Signal Processing in Acoustics:** Ambient Sound Measurements and Models II. Studio 9/10

Wednesday Morning

9:00 3aAAa **Architectural Acoustics:** Architectural Acoustics Potpourri. Galerie 3

9:00 3aAAb **Architectural Acoustics:** At the Intersection of Speech and Architecture III. Galerie 2

9:00 3aAB **Animal Bioacoustics:** Arthropod Biotremology and Bioacoustics. Galerie 4

9:55 3aAO **Acoustical Oceanography:** Bioacoustic Attenuation Spectroscopy. Studio 9/10

10:00 3aBA **Biomedical Acoustics:** Biomedical Acoustics Best Student Paper Award Poster Session. Studio Foyer

9:00 3aCA **Computational Acoustics, Physical Acoustics, Underwater Acoustics, and Acoustical Oceanography:** Parabolic Equation Methods Across Acoustics. Studio 6

9:00 3aEA **Engineering Acoustics:** Recording and Processing of Higher-Order Spatial Audio. Balcony N

11:00 3aED **Education in Acoustics:** Education in Acoustics Prize Lecture. Salon F/G

8:00 3aIDa **Interdisciplinary:** Plenary Lecture: Selective Listening in Music: From Psychoacoustical Principles to Hearing Device Evaluation. Bissonet/Carondelet

10:55 3aIDb **Interdisciplinary:** Hot Topics in Acoustics. Bissonet/Carondelet

9:15 3aNS **Noise and Archives and History:** History of Acoustics and Evolution of Sound Measurement. Galerie 5

9:00 3aPAa **Physical Acoustics and Structural Acoustics and Vibration:** Mesoscopics in Acoustics and Elasticity I. Balcony I

9:15 3aPAb **Physical Acoustics and Computational Acoustics:** Session in Honor of Richard Raspet I. Balcony J

8:55 3aSA **Structural Acoustics and Vibration and Musical Acoustics:** Friction Acoustics. Balcony M

8:55 3aSC **Speech Communication and Psychological and Physiological Acoustics:** Speech Perception Beyond Intelligibility I. Salon H

8:55 3aSP **Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics:** Physics-Inspired Neural Networks (PINNs) in Underwater Acoustics. Galerie 1

8:55 3aUW **Underwater Acoustics and Acoustical Oceanography:** Boundary Interactions Including Shear Wave Effects in Underwater Acoustics. Studio 7/8

Wednesday Afternoon

1:20 3pAA **Architectural Acoustics:** At the Intersection of Speech and Architecture IV. Galerie 2

1:00 3pAOa **Acoustical Oceanography:** Acoustical Oceanography Prize Lecture. Studio 9/10

2:20 3pAOb **Acoustical Oceanography:** Decadal Survey for Ocean Acoustics. Studio 9/10

1:00 3pBA **Biomedical Acoustics:** General Topics in Biomedical Acoustics: Cavitation. Balcony K

1:00 3pNS **Noise:** General Topics in Noise: Community Noise. Galerie 5

12:55 3pPAa **Physical Acoustics, Education in Acoustics, and Structural Acoustics and Vibration:** Nonlinear Waves in Architected Solids. Balcony I

12:55 3pPAb **Physical Acoustics and Computational Acoustics:** Session in Honor of Richard Raspet II. Balcony J

1:00 3pPP **Psychological and Physiological Acoustics:** William and Christine Hartmann Prize in Auditory Neuroscience Lecture. Salon F/G

12:55 3pSA **Structural Acoustics and Vibration:** Constrained Layer Damping. Balcony M

1:20 3pSC **Speech Communication and Psychological and Physiological Acoustics:** Speech Perception Beyond Intelligibility II. Salon H

1:00 3pSP **Signal Processing in Acoustics:** Signal Processing Poster Session. Studio Foyer

1:00 3pUW **Underwater Acoustics:** Acoustics of Marine Sediments. Studio 7/8

Thursday Morning

7:00 4aAAa **Architectural Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Data-Driven Room Acoustics I. Galerie 3

7:00 4aAAb **Architectural Acoustics and Structural Acoustics and Vibration:** The Intersection of the Acoustic and Structural Domains in Sound Transmission in Buildings. Galerie 2

7:20 4aAB **Animal Bioacoustics:** Acoustic Ecology and Biological Soundscapes. Galerie 4

7:55 4aBAa **Biomedical Acoustics:** Bubbles and Ultrasound - Physiological Considerations I. Balcony K

7:55 4aBAb **Biomedical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Wave Propagation and Aberration in Complex Media: From Theory to Applications I. Balcony L

7:40 4aEA **Engineering Acoustics:** General Topics in Engineering Acoustics. Balcony N

8:40 4aED **Education in Acoustics:** General Topics in Education in Acoustics. Salon H

11:10 4aID **Interdisciplinary:** Plenary Lecture: Inclusive Speech Technology: Developing Automatic Speech Recognition for Everyone. Bissonet/Carondelet

8:20 4aMU **Musical Acoustics and Computational Acoustics:** String Instruments I. Galerie 6

9:00 4aNS **Noise:** General Topics in Noise: Community Noise Perception and Psychoacoustics. Galerie 5

8:00 4aPAa **Physical Acoustics and Computational Acoustics:** Infrasound I. Balcony J

8:00 4aPAb **Physical Acoustics and Structural Acoustics and Vibration:** Mesoscopics in Acoustics and Elasticity II. Balcony I

8:00 4aPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics Poster Session II (Poster Session). Studio Foyer

9:20 4aSA **Structural Acoustics and Vibration:** Aerospace and Structural Acoustics. Balcony M

9:00 4aSC **Speech Communication:** Speech Perception Poster Session II. Studio Foyer

7:55 4aSPa **Signal Processing in Acoustics and Underwater Acoustics:** Universal and Doubly Adaptive Methods for Signal Processing I. Studio 7/8

8:20 4aSPb **Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics:** Machine Learning in Underwater Acoustics II. Galerie 1

7:00 4aUW **Underwater Acoustics, Acoustical Oceanography, Computational Acoustics, and Signal Processing in Acoustics:** Directional Sensing: Applications and Methods I. Studio 9/10

Thursday Afternoon

1:00 4pAAa **Architectural Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Data-Driven Room Acoustics II. Galerie 3

12:55 4pAAb **Architectural Acoustics, Noise and Underwater Acoustics:** Memorial Session Honoring David Lubman. Galerie 2

1:20 4pAB **Animal Bioacoustics:** General Topics in Animal Bioacoustics. Galerie 4

1:00 4pBAa **Biomedical Acoustics:** Bubbles and Ultrasound–Physiological Considerations II. Balcony K

1:00 4pBAb **Biomedical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Wave Propagation and Aberration in Complex Media: From Theory to Applications II. Balcony L

1:15 4pED **Education in Acoustics and Musical Acoustics:** Acoustics Education Research for Newbies: The Science of Teaching and Learning. Salon H

1:20 4pMU **Musical Acoustics and Computational Acoustics:** String Instruments II. Galerie 6

1:20 4pNS **Noise:** Exposure Response and Community Tolerance Level. Galerie 5

1:00 4pPAa **Physical Acoustics and Computational Acoustics:** Infrasound II. Balcony J

1:00 4pPAb **Physical Acoustics:** General Topics in Physical Acoustics I. Balcony I

12:55 4pPPa **Psychological and Physiological Acoustics:** Virtual Thunder: Top Presentations from the P&P Trainee Lightning Round competition. Salon F/G

3:15 4pPPb **Psychological and Physiological Acoustics and Signal Processing in Acoustics:** Cadenza Machine Learning Challenge (CAD2): Improving Music for People with Hearing Loss. Salon F/G

12:55 4pSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Active and Tunable Acoustic Metamaterials. Balcony M

1:20 4pSC **Speech Communication:** Speech Production Poster Session II. Studio Foyer

1:00 4pSPa **Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics:** Machine Learning in Underwater Acoustics III. Galerie 1

1:00 4pSPb **Signal Processing in Acoustics and Underwater Acoustics:** Universal and Doubly Adaptive Methods for Signal Processing II. Studio 7/8

1:00 4pUWa **Underwater Acoustics, Acoustical Oceanography, Computational Acoustics, and Signal Processing in Acoustics:** Directional Sensing: Applications and Methods II. Studio 9/10

3:20 4pUWb **Underwater Acoustics:** Underwater Acoustic Communications. Studio 9/10

Friday Morning

7:00 5aAA **Architectural Acoustics:** Theatres, Auditoria, and Other Gathering Spaces. Galerie 3

8:00 5aAB **Animal Bioacoustics:** Animal Vocal Communication and Physiology. Galerie 4

7:40 5aBA **Biomedical Acoustics:** General Topics in Biomedical Acoustics: Quantitative Ultrasound. Balcony L

11:10 5aID **Interdisciplinary:** Plenary Lecture: Acoustic Ecology of Marine Mammals in the Era of Blue Economy: Navigating Development and Ocean Noise Challenges. Bissonet/Carondelet

8:40 5aMU **Musical Acoustics:** General Topics in Musical Acoustics III. Galerie 6

9:00 5aNS **Noise:** General Topics in Noise: Measurement and Processing I. Galerie 5

8:00 5aPAa **Physical Acoustics, Underwater Acoustics, and Acoustical Oceanography:** Meteorological Acoustics. Balcony I

8:00 5aPAb **Physical Acoustics:** Physical Acoustics Best Student Paper Award Poster Session. Studio Foyer

9:30 5aPAc **Physical Acoustics:** General Topics in Physical Acoustics II (Poster Session). Studio Foyer

8:00 5aSP **Signal Processing in Acoustics:** Signal Processing Potpourri III. Galerie 1

7:00 5aUW **Underwater Acoustics and Acoustical Oceanography:** Bill Kuperman (1943-2024): Contributions to the Field of Underwater Acoustics I. Studio 9/10

Friday Afternoon

1:00 5pAA **Architectural Acoustics:** Materials for Sound Absorption, Diffusion, and Transmission Loss. Galerie 3

1:00 5pBA **Biomedical Acoustics:** General Topics in Biomedical Acoustics: Tissue Characterization. Balcony K

12:55 5pMU **Musical Acoustics, Architectural Acoustics, Psychological and Physiological Acoustics, and Noise:** Discrimination Tests: Methodologies and Applications. Galerie 6

1:00 5pNS **Noise:** General Topics in Noise: Measurement and Processing II. Galerie 5

1:00 5pPAa **Physical Acoustics:** General Topics in Physical Acoustics III. Studio 7/8

1:00 5pPAb **Physical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Topological Aspects of Acoustic Waves. Galerie 1

1:00 5pUWa **Underwater Acoustics and Acoustical Oceanography:** Bill Kuperman (1943-2024): Contributions to the Field of Underwater Acoustics II. Studio 9/10

3:40 5pUWb **Underwater Acoustics:** Underwater Acoustic Propagation. Studio 6

SCHEDULE OF COMMITTEEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Sun, 18 May, 10:00 a.m.	ICA Board Meeting	Lafayette
Mon, 19 May, 9:00 a.m.	Executive Council	St. Charles
Mon, 19 May, 1:00 p.m.	Technical Council	St. Charles
Mon, 19 May, 6:00 p.m.	WESPAC	St. Charles
Tue, 20 May, 7:00 a.m.	Member Engagement	Lafayette
Tue, 20 May, 7:30 a.m.	Editorial Board	St. Charles
Tue, 20 May, 8:30 a.m.	CIRDI	Lafayette
Tue, 20 May, 1:30 p.m.	Meetings	Lafayette
Tue, 20 May, 5:00 p.m.	Newman Fund	Lafayette
Tue, 20 May, 7:30 a.m.	Panel on Public Policy	St. Charles
Wed, 21 May, 7:00 a.m.	Regional and Student Chapters	St. Charles
Wed, 21 May, 7:30 a.m.	Finance	Lafayette
Wed, 21 May, 9:30 a.m.	Acoustical Society Foundation Fund	Lafayette
Wed, 21 May, 11:00 a.m.	Medals and Awards	St. Charles
Wed, 21 May, 12:00 p.m.	Public Relations	Napolean
Thu, 22 May, 7:30 a.m.	Practitioners and Industry	Napoleon
Thu, 22 May, 9:30 a.m.	ICA Board	Bacchus
Fri, 23 May, 8:00 a.m.	Technical Council	St. Charles
Fri, 23 May, 11:00 noon	Executive Council	St. Charles

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 20 May, 5:30 p.m.	Acoustical Oceanography	Studios 7/8
Tue, 20 May, 6:00 p.m.	Animal Bioacoustics	Galerie 4
Tue, 20 May, 6:00 p.m.	Architectural Acoustics	Galerie 2
Tue, 20 May, 5:30 p.m.	Engineering Acoustics	Balcony N
Tue, 20 May, 5:30 p.m.	Signal Processing in Acoustics	Salon D
Tue, 20 May, 5:30 p.m.	Speech Communication	Studios 9/10
Wed, 21 May, 7:30 p.m.	Biomedical Acoustics	Balcony K
Wed, 21 May, 7:30 p.m.	Education in Acoustics	Salon /F/G
Wed, 21 May, 7:30 p.m.	Structural Acoustics and Vibration	Balcony M
Thu, 22 May, 5:30 p.m.	Computational Acoustics	Studio 6
Thu, 22 May, 5:30 p.m.	Musical Acoustics	Galerie 6
Thu, 22 May, 5:30 p.m.	Noise	Galerie 5
Thu, 22 May, 7:30 p.m.	Physical Acoustics	Balcony J
Thu, 22 May, 5:30 p.m.	Psychological and Physiological Acoustics	Salon F/G
Thu, 22 May, 7:30 p.m.	Underwater Acoustics	Studios 9/10

STANDARDS COMMITTEES AND WORKING GROUPS

Tue, 20 May, 8:00 a.m.	Standards Plenary including TAGs	Napolean
Tue, 20 May, 9:15 a.m.	ASC S12 Noise	Napolean
Tue, 20 May, 10:30 a.m.	ASC S2 Mechanical Vibration and Shock	Napolean
Tue, 20 May, 12:15 p.m.	ASC S3 Bioacoustics	Napolean
Tue, 20 May, 1:30 p.m.	ASC S3/SC1 Animal Bioacoustics	Napolean
Tue, 20 May, 2:45 a.m.	ASC S1 Acoustics	Napolean

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Sun, 18 May 2:00 p.m.–5:00 p.m.	Registration	2nd floor foyer
Sun, 18 May 5:00 p.m.–6:00 p.m.	Opening Ceremony	Carondelet
Sun, 18 May, 6;00 p.m.–7:00 p.m.	Opening Reception	Bissonet
Mon, 19 May 7:00 a.m. –5:00 p.m.	Registration	2nd floor foyer
Tue–Thu, 20–22 May 7:30 a.m. – 5:00		
Fri, 21 May 7:30 a.m.–12:00 noon		
Mon–Fri, 19–23 May 7:00 a.m.–5:30 p.m.	A/V Preview	Bonaparte
Mon–Fri, 19–23 May 7:00 a.m.–5:00 p.m.	Mothers' Room	Audobon
Mon, Tue, Thu, Fri, 19, 20, 22, 23 May 10:00 a.m.–11:20 a.m. 9:00 a.m.–10:20 p.m.	Morning Coffee Breaks	Acadia Foyer
Tue, 20 May 2:20 p.m.–3:20 p.m.	Afternoon Coffee Break	Acadia
Wed, 21 May 10:00 a.m.–11:20 a.m.	Morning Coffee Breaks	Acadia Foyer
Mon, 19 May 8:00 a.m.–10:00 a.m.	Accompanying Persons	Lafayette
Mon, 19 May 5:30 p.m.–7:00 p.m.	Exhibit Opening Reception	Acadia
Mon, 19 May 5:00 p.m.–5:30 p.m.	Student & First Time Attendee	Balcony M
Mon, 19 May 5:45 p.m. –7:30 p.m.	Student Meet and Greet	Riverview I, II, Prefunction
Tue, 20 May 9:00 a.m.–5:00 p.m. Wednesday, 21 May 9:00 a.m.–12 noon	Exhibit	Acadia
Tue, 20 May 3:00 p.m.–4:00 p.m.	Women's Roundtable Discussion	St. Charles
Tue, 20 May 6:00 p.m.–8:00 p.m.	Student Reception	Riverview I, II, Prefunction
Wed, 21 May 9:00 a.m.–12:00 p.m.	ICA General Assembly	Salons A–E
Wed, 21 May 11:45 a.m.–1:30 p.m.	Women in Acoustics Luncheon	Riverview II
Wed, 21 May 3:45 p.m.–6:00 p.m.	Plenary Session/Awards Ceremony	Salons A–E
Wed, 21 May 6:00 p.m.–7:30 p.m.	Social Hour	Carolendete/ Bissonet
Wed, 21 May 8:00 p.m.– 12:00 a.m.	ASA Jam	Salons A–E
Thu, 22 May 3:00 p.m.–4:00 p.m.	Student Grant Panel	St. Charles
Thu, 22 May 2:00 p.m.–4:00 p.m.	Strategic Plan Champions	Acadia
Fri, 23 May 5:00 p.m.–6:00 p.m.	ICA Closing Ceremony/ Reception	Bissonet

188th Meeting of the Acoustical Society of America
25th International Congress on Acoustics (ICA2025 New Orleans)

The joint 188th meeting of the Acoustical Society of America and the 25th International Congress on Acoustics (ICA2025 New Orleans) will be held Sunday through Friday, 18-23 May 2025 at the New Orleans Marriott Hotel, New Orleans, Louisiana, USA.

SECTION HEADINGS

1. REGISTRATION
2. TECHNICAL SESSIONS
3. TECHNICAL SESSION DESIGNATIONS
4. PLENARY SESSION
5. OPENING CEREMONY AND OPENING RECEPTION
6. EXHIBIT AND EXHIBIT OPENING RECEPTION
7. PRIZES AND PRIZE LECTURES
8. TECHNICAL COMMITTEE OPEN MEETINGS
9. WOMEN IN ACOUSTICS ROUNDTABLE
10. PLENARY SESSION AND AWARDS CEREMONY
11. CLOSING CEREMONY
12. INTERNATIONAL SYMPOSIUM ON MUSICAL ACOUSTICS AND ROOM
13. ANSI STANDARDS COMMITTEES
14. COFFEE BREAKS
15. A/V PREVIEW ROOM
16. MOTHERS ROOM
17. SOCIAL
18. STUDENT EVENTS: NEW STUDENTS/FIRST-TIME ATTENDEE ORIENTATION, MEET AND GREET, STUDENT CAREER MIXER, STUDENT RECEPTION
19. WOMEN IN ACOUSTICS LUNCHEON
20. JAM SESSION
21. ACCOMPANYING PERSONS
22. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)
23. TECHNICAL PROGRAM ORGANIZING COMMITTEE
24. MEETING ORGANIZING COMMITTEE
25. PHOTOGRAPHING AND RECORDING
26. ABSTRACT ERRATA
27. GUIDELINES FOR ORAL PRESENTATIONS
28. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS
29. DATES OF FUTURE ASA MEETINGS

1. REGISTRATION

Registration is required for all attendees and accompanying persons. Registration badges must be worn in order to participate in technical sessions and other meeting activities.

Registration will open on Sunday, 18 May, at 2:00 p.m. in the Foyer on the second floor of the New Orleans Marriott Hotel.

Visa, MasterCard and American Express credit cards and checks in US dollars drawn on a bank in the US will be accepted for payment of registration. Meeting attendees who have pre-registered may pick up their badges and registration materials at the pre-registration desk.

The registration fees (in USD) are $950 Regular Registrants, $400 for One-day Registrants, $250 for ASA Student members, and $150 for Accompanying Persons. Emeritus registration is not available on-site at the meeting.

One-day registration is for attendees who attend the meeting on only one day either to present a paper and/or to attend sessions. Accompanying Persons registration is for persons who will not participate in the technical program of the meeting either as presenters or to attending sessions without presenting.

Special note to students who pre-registered online: You will also be required to show your student id card when picking-up your registration materials at the meeting.

2. TECHNICAL SESSIONS

The technical program includes over 1300 abstracts.

Session Chairs have been instructed to adhere strictly to the printed time schedule, both to be fair to all speakers and to permit attendees to schedule moving from one session to another to hear specific papers. If an author is not present to deliver a lecture-style paper, the Session Chairs have been instructed either to call for additional discussion of papers already given or to declare a short recess so that subsequent papers are not given ahead of the designated times.

Several sessions are scheduled in poster format, with the display times indicated in the program schedule.

3. TECHNICAL SESSION DESIGNATIONS

Abstract code examples: 1aAA1, 2pBAb4, 1eID1

The first character is a number indicating the day the session will be held, as follows:

1-Monday, 19 May
2-Tuesday, 20 May
3-Wednesday, 21 May
4-Thursday, 22 May
5-Friday, 23 May

The second character is a lower case "a" for a.m., "p" for p.m., or "e" for evening corresponding to the time of day the session will take place. The third and fourth characters are capital letters indicating the primary Technical Committee that organized the session using the following abbreviations or codes:

AA Architectural Acoustics
AB Animal Bioacoustics
AO Acoustical Oceanography
BA Biomedical Acoustics
CA Computational Acoustics
EA Engineering Acoustics
ED Education in Acoustics
ID Interdisciplinary
MU Musical Acoustics
NS Noise
PA Physical Acoustics
PP Psychological and Physiological Acoustics

SA Structural Acoustics and Vibration
SC Speech Communication
SP Signal Processing in Acoustics
UW Underwater Acoustics

In sessions where the same group is the primary organizer of more than one session scheduled in the same morning or afternoon, a fifth character, either lower-case "a," or "b," is used to distinguish the sessions. Each paper within a session is identified by a paper number following the session-designating characters, in conventional manner. As hypothetical examples: paper 2pEA3 would be the third paper in a session on Tuesday afternoon organized by the Engineering Acoustics Technical Committee; 3pSAb5 would be the fifth paper in the second of two sessions on Wednesday afternoon sponsored by the Structural Acoustics and Vibration Technical Committee.

Note that technical sessions are listed both in the calendar and the body of the program in the numerical and alphabetical order of the session designations rather than the order of their starting times. For example, session 3aAA would be listed ahead of session 3aAO even if the latter session begins earlier in the same morning.

4. PLENARY LECTURES

A Plenary Lecture will be presented each day of the meeting at 11:00 8:00 a.m. on Monday, Tuesday, Thursday, and Friday and at 11:10 a.m. on Wednesday.

5. OPENING CEREMONY AND OPENING RECEPTION

The Opening Ceremony will be held on Sunday, 18 May, 5:00 p.m. in the Carondelet Room. Dr. Michael White of the University of New Orleans will review the history of New Orleans Jazz followed by followed by a performance by his Original Liberty Jazz Band.

The Opening Reception will follow at 6:00 p.m. in the Bissonet Room with light refreshments.

6. EXHIBIT AND EXHIBIT OPENING RECEPTION

An instrument and equipment exhibition will be located in the Acadia Room on the 3rd floor and will open on Monday, 19 May, with an evening reception serving a complimentary drink. Exhibit hours are Monday, 19 May, 5:30 p.m. to 7:00 p.m., Tuesday, 20 May, 9:00 a.m. to 5:00 p.m., and Wednesday, 21 May, 9:00 a.m. to 12:00 noon.

7. PRIZES AND PRIZE LECTURES

The Auditory Neuroscience Prize Lecture will be presented on Wednesday, 21 May, in session 3pPP at 1:30 p.m. in Salon F/G. The Acoustical Oceanography Prize Lecture will be presented on Wednesday, 21 May, in session 3pAOa at 1:00 p.m. in Studios 9/10. The Acoustics Education Prize Lecture will be presented on Wednesday, 21 May, in session 3aED at 11:00 a.m. in Salon F/G.

8. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday. The schedule and rooms for each Committee meeting are given on page A7

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussions.

9. WOMEN IN ACOUSTICS ROUNDTABLE

A Roundtable sponsored by the Women in Acoustics Committee will be held on Tuesday, 20 May, 3:00 p.m. in the St. Charles Room. Topics will include work/life balance, the 2-body problem, advocating for your career, finding a mentor, and understanding employee benefits. Volunteers will lead separate discussions on these and other topics. You may join one conversation or jump between the groups as needed. Bring your questions and challenges and an open mind for discussion.

10. PLENARY SESSION AND AWARDS CEREMONY

A plenary session and awards ceremony will be held Wednesday, 21 May, at 3:30 p.m. in Salons A-E. ASA scholarship recipients will be introduced and newly-elected Fellows will be announced, and ASA Prizes and Awards will be presented.

All attendees are welcome and encouraged to attend. Please join us to honor and congratulate these medalists and other award recipients.

11. CLOSING CEREMONY AND CLOSING RECEPTION

A Closing Ceremony and Reception will be held on Friday, 23 May, at 3:00 p.m. in the Carondelet and Bissonet Rooms. Light refreshments will be available.

12. INTERNATIONAL SYMPOSIUM ON MUSICAL AND ROOM ACOUSTICS (ISMRA)

The ISMRA joint symposium will take place 25-27 May, Loyola University campus in New Orleans, a short distance from the New Orleans Marriott.

Two optional workshops are scheduled for Saturday May 24.

The symposium will feature plenary sessions and keynote speakers addressing topics that encompass both musical and room acoustics including parallel sessions, both oral and poster, on technical topics of interest to the respective musical acoustics and room acoustics communities.

Registration fees after 21 April are $450 for Regular registration and $225 for Students.

13. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be held at the meeting as noted in the schedule of meetings and other events.

People interested in attending and in becoming involved in working group activities should contact the ASA Standards Manager for further information about these groups, or about the ASA Standards Program in general, at the following address: Nancy Blair-DeLeon, ASA Standards Manager, Standards Secretariat, Acoustical Society of America, asastds@acousticalsociety.org

14. COFFEE BREAKS

Morning coffee breaks will be held daily from 9:00 a.m. to 10:20 a.m. and an afternoon break will be held on Tuesday from 2:20 p.m. to 3:20 p.m. in the Acadia Room on Monday, Tuesday, and Wednesday and in the Acadia Foyer on Thursday and Friday.

15. A/V PREVIEW ROOM

The A/V preview room will be set up in the Bonaparte Room and will be available on Monday through Friday from 7:00 a.m. to 5:00 p.m.

16. MOTHERS ROOM

A Mothers Room for ASA meeting attendees will be available Monday to Friday, 19-23 May, in Audobon room. The hours are Monday to Friday, 8:00 a.m. to 5:00 p.m.

17. SOCIAL

A Social will be held on Wednesday evening, 6:00 p.m. to 7:30 p.m. in the Carondelet/Bissonet Room on the third floor.

The Social provides a relaxing setting for meeting attendees to meet and mingle with their friends and colleagues as well as an opportunity for new members and first-time attendees to meet and introduce themselves to others in the field.

18. STUDENT EVENTS: NEW STUDENTS/FIRST-TIME ATTENDEE ORIENTATION, MEET AND GREET, STUDENT CAREER MIXER, STUDENT RECEPTION

Follow the student twitter throughout the meeting @ASAStudents.

A New Students/First-Time Attendee Orientation will be held on Monday, 19 May, from 5:30 p.m. to 6:00 p.m. in Balcony M followed by the Student Meet and Greet from 6:00 p.m. to 7:30 p.m. in the Riverview Room on the 41st floor where refreshments and a cash bar will be available.

The Student Career Mixer will be held on Thursday, 22 May, 11:30 a.m. to 2:00 p.m. in Riverview. Students will mingle with representatives from potential employers in industry, government institutions, and consulting firms during an informal lunch. The purpose of this event is to expose students to career opportunities with an emphasis on cross-technical committee awareness.

The Students' Reception will be held on Tuesday, 20 May, from 6:00 p.m. to 8:00 p.m. in Riverview 1 and 2. This reception will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

19. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at Wednesday, 21 May, at 11:45 a.m. in Riverview 1 and 2 on the 41st floor. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Wednesday, 21 May. The fee is USD $35 for non-students and USD $15 for students.

20. JAM SESSION

You are invited to Salons A-E on Wednesday night, 21 May, from 8:00 p.m. to midnight for the ASA Jam. Bring your axe, horn, sticks, voice, or anything else that makes music. Musicians and non-musicians are all welcome to attend. A full PA system, backline equipment, guitars, bass, keyboard, and drum set will be provided. All attendees will enjoy live music, a cash bar with snacks, and all-around good times. Don't miss out.

21. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the meeting. The on-site registration fee for accompanying persons is USD $150. A hospitality room for accompanying persons will be open in the Lafayette Room (41st floor) 8:00 a.m. to 10:00 a.m. Monday, 19 May. Accompanying Persons entitles you access to the accompanying persons room, the Tuesday social, the Jam Session, and the Plenary Session on Wednesday afternoon. It does not provide access to attend technical sessions.

22. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The meeting will have a published proceedings and submission is optional. The proceedings will be a separate volume of the online journal, "Proceedings of Meetings on Acoustics" (POMA). This is an open access journal, so that its articles are available in pdf format for downloading without charge to anyone in the world. Authors who are scheduled to present papers at the meeting are encouraged to prepare a suitable version in pdf format that will appear in POMA. It is not necessary to wait until after the meeting to submit one's paper to POMA. Further information regarding POMA can be found at the site http://acousticsauthors.org. Published papers from previous meeting can be seen at the site http://asadl/poma.

23. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Michael Haberman, Preston Wilson, Technical Program Chairs; Christopher Bassett, Acoustical Oceanography; Heloise Fruin-Mouy, Animal Bioacoustics; Brandon Cudequest, Kaitlin Hunt, Architectural Acoustics; John Cormack, James Kwan, Biomedical Acoustics, Amanda Hanford, Computational Acoustics; Ahmed Allam, Michael Haberman, Engineering Acoustics; Kimberly Riegel, Education in Acoustics; Mark Rau, Gary Scavone, Musical Acoustics; Alexandra Loubeau, Hales Swift, Noise; Raphael Hermann, Joel Lonzaga, Physical Acoustics; Gregory Ellis, Nirmal Srinivasan, Christopher Stecker, Psychological and Physiological Acoustics; William Jenkins, Signal Processing in Acoustics; Kelly Berkson, Megan Clayards, Lisa Redford, Benjamin Tucker, Speech Communication; Anthony Bonomo, Stephanie Konarski, Structural Acoustics and Vibration; David Dall'Osto, Underwater Acoustics; Brijonnay Madrigal, Student Council.

24. MEETING ORGANIZING COMMITTEE

Congress Chair: Mark Hamilton (University of Texas at Austin), Congress Vice Chair: Joel Mobley (University of Mississippi), Technical Program Chairs: Michael Haberman (University of Texas at Austin) and Preston Wilson (University of Texas at Austin), ASA Liaison: Susan Fox (ASA), Proceedings Managers: Megan Ballard (Editor, Proceedings of Meetings on Acoustics) Webmaster: Chirag Gokani (University of Texas at Austin), Meeting Planner: Debra Nolan (AMC Source)

Technical Committees/Technical Specialty Group Chairs (ASA)

Acoustical Oceanography: David Barclay, Animal Bioacoustics: Xavier Mouy, Architectural Acoustics: David Woolworth, Biomedical Acoustics: Julianna Simon, Computational Acoustics: Jennifer Cooper, Education in Acoustics: Kimberly Riegel, Engineering Acoustics: Joseph Vignola, Musical Acoustics: Jonas Braasch, Noise: Michelle Vigeant-Haas, Physical Acoustics: Christopher Kube, Psychological and Physiological Acoustics: Christopher Stecker, Signal Processing in Acoustics: Paul Hursky, Speech Communication: Benjamin Tucker, Structural Acoustics and Vibration: Micah Shepherd, Underwater Acoustics: David Dall'Osto.

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25. PHOTOGRAPHING AND RECORDING

Photographing and recording during regular sessions are not permitted without prior permission from the Acoustical Society.

26. ABSTRACT ERRATA

This meeting program is Part 2 of the November 2024 issue of *The Journal of the Acoustical Society of America*. Corrections, for printer's errors only, may be submitted for publication in the Errata section of the Journal.

27. GUIDELINES FOR ORAL PRESENTATIONS

Preparation of Visual Aids

- See the guidelines for computer projection in section 41 below.
- Allow at least one minute of your talk for each slide (e.g., PowerPoint). No more than 12 slides for a 15-minute talk (with 3 minutes for questions and answers).
- Minimize the number of lines of text on one visual aid. 12 lines of text should be a maximum. Include no more than 2 graphs/plots/figures on a single slide. Too little information is better than too much.
- Presentations should contain simple, legible text that is readable from the back of the room.
- Characters should be at least 0.25 inches (6.5 mm) in height to be legible when projected. A good rule of thumb is that text should be 20 point or larger (including labels in inserted graphics). Anything smaller is difficult to read.
- Make symbols at least 1/3 the height of a capital letter.
- For computer presentations, use all of the available screen area using landscape orientation with very thin margins. If your institutions logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide. Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- DVDs should be in standard format.
- Presentation
- Organize your talk with introduction, body, and summary or conclusion. Include only ideas, results, and concepts that can be explained in the allotted time. Four elements to include are:
- Statement of research problem
- Research methodology
- Review of results
- Conclusions
- No more than 3–5 key points can be covered adequately in a 15-minute talk so keep it concise.
- Rehearse your talk so you can confidently deliver it in the allotted time. Session Chairs have been instructed to adhere to the time schedule and to stop your presentation if you run over.
- An A/V preview room will be available for viewing computer presentations before your session starts. It is advisable to preview your presentation because in most cases you will be asked to load your presentation onto a computer which may have different software or a different configuration from your own computer.

- Arrive early enough so that you can meet the session chair, load your presentation on the computer provided, and familiarize yourself with the microphone, computer slide controls, laser pointer, and other equipment that you will use during your presentation. There will be many presenters loading their materials just prior to the session so it is important that you check that all multi-media elements (e.g., sounds or videos) play accurately prior to the day of your session.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They won't have time to figure it out for themselves. A popular myth is that a technical audience requires a lot of technical details. Less can be more.
- Turn off your cell phone prior to your talk and put it away from your body. Cell phones can interfere with the speakers and the wireless microphone.

28. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

The poster should be centered around two or three key points supported by the title, figures, and text. The poster should be able to "stand alone." That is, it should be understandable even when you are not present to explain, discuss, and answer questions. This quality is highly desirable since you may not be present the entire time posters are on display, and when you are engaged in discussion with one person, others may want to study the poster without interrupting an ongoing dialogue.

- To meet the "stand alone" criteria, it is suggested that the poster include the following elements, as appropriate:
- Background
- Objective, purpose, or goal
- Hypotheses
- Methodology
- Results (including data, figures, or tables)
- Discussion
- Implications and future research
- References and Acknowledgment

Design and layout

- A board approximately 8 ft. wide × 4 ft. high will be provided for the display of each poster. Supplies will be available for attaching the poster to the display board. Each board will be marked with an abstract number.
- Typically, posters are arranged from left to right and top to bottom. Numbering sections or placing arrows between sections can help guide the viewer through the poster.
- Centered at the top of the poster, include a section with the abstract number, paper title, and author names and affiliations. An institutional logo may be added. Keep the design simple and uncluttered. Avoid glossy paper.

Lettering and text

- Font size for the title should be large (e.g., 70-point font)
- Font size for the main elements should be large enough to facilitate readability from 2 yards away (e.g., 32-point font). The font size for other elements, such as references, may be smaller (e.g., 20–24 point font).
- Sans serif fonts (e.g., Arial, Calibri, Helvetica) are much easier to read than serif fonts (e.g., Times New Roman).
- Text should be brief and presented in a bullet-point list as much as possible. Long paragraphs are difficult to read in a poster presentation setting.

Visuals

- Graphs, photographs, and schematics should be large enough to see from 2 yards (e.g., 8 × 10 inches).
- Figure captions or bulleted annotation of major findings next to figures are essential. To ensure that all visual elements are "stand alone," axes should be labeled and all symbols should be explained.
- Tables should be used sparingly and presented in a simplified format.

Presentation

- Prepare a brief oral summary of your poster and short answers to questions in advance.
- The presentation should cover the key points of the poster so that the audience can understand the main findings. Further details of the work should be left for discussion after the initial poster presentation.
- It is recommended that authors practice their poster presentation in front of colleagues before the meeting. Authors should request feedback about the oral presentation as well as poster content and layout.

Other suggestions

- You may wish to prepare reduced-size copies of the poster (e.g., 8 1/2 × 11 sheets) to distribute to interested audience members.

29. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, E-mail: asa@acousticalsociety.org.

189th Meeting – joint with the Acoustical Society of Japan, Honolulu, Hawaii, 1-5 December 2025

190th Meeting, Philadelphia, Pennsylvania, 11-15 May 2026.

ANNUAL GIVING TO THE ACOUSTICAL SOCIETY FOUNDATION FUND – 2024

The Acoustical Society of America Foundation Board is deeply grateful for all contributions received in 2024. To help express this gratitude, the list of donors to the Foundation Fund is published for all donations received in 2024.

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Contributors – up to $99
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*Balachandran, Balakumar
Barath, Jim D.
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Willson, Abigail
Yurk, Harald

Session 1aAAa

Architectural Acoustics, Noise and ASA Committee on Standards: Day of ASHRAE Part I—Emerging Trends in HVAC Noise and Noise Control

Derrick P. Knight, Cochair
Trane Technologies, 2313 20th Street South, La Crosse, WI 54601

Jerry G. Lilly, Cochair
JGL Acoustics, Inc., 5266 NW Village Park Drive, Issaquah, WA 98027

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Chair's Introduction—7:00

Invited Papers

7:05

1aAAa1. Acoustical impacts of operating air-cooled chillers as heat pumps. Scott Hausmann (Trane Technologies, 2313 20th St. South, La Crosse, WA 54601, shausmann@trane.com) and Derrick P. Knight (Trane Technologies, La Crosse, WI)

With the increased application of heat pumps across wider geographic regions, it is useful to understand how the operating point of an air-cooled chiller shifts when functioning as a heat pump and what impact this has on its acoustics. This paper will review how compressor and fan operating points change when in heating mode and the trends in sound power level with these changes. Measured reverberant room results of compressors at typical heating and cooling conditions will be presented. The impact of unit control strategies in the different modes will be discussed and their impact on sound will be summarized.

7:25

1aAAa2. Complaints versus compliance: Engineering noise and airflow solutions for air-cooled chillers in data centers. Viken Koukounian (Parklane Mech. Acoust., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@parklanemechanical.com) and Matthew Downey (Parklane Mech. Acoust., Oakville, ON, Canada)

Air-cooled chillers (ACC) are essential to data center operations but present challenges in balancing acoustical performance, airflow efficiency, and structural constraints. This session examines the interplay between compliance and community complaints, focusing on downstream risks such as schedule delays, legal exposure, and cost overruns. It explores engineering approaches to mitigate noise, address airflow limitations—including static pressure effects and re-entrainment—and optimize thermal and energy performance. Using computational fluid dynamics (CFD) modeling, we evaluate airflow patterns, including dispersion and diffusion, to assess their impact on system performance and noise propagation. The session highlights structural considerations for equipment selection, site layouts, and retrofits, demonstrating how proactive design strategies can mitigate permitting delays and reduce complaints. Attendees will gain insight into integrating noise control measures with airflow and thermal performance requirements to de-risk projects while achieving compliance and reliability in demanding data center environments.

7:45

1aAAa3. Equipment sound data—What we want and how to get it. Andy Carballeira (ArchMech, Acentech, 33 Moulton St., Cambridge, MA 02138, acarballeira@acentech.com), Jack Taylor, and Bill Yoder (ArchMech, Acentech, Cambridge, MA)

Data center construction and commissioning continues to grow at a fast pace. Noise control engineers and equipment vendors have an important role to play in ensuring that these facilities can function at the cutting edge, as good neighbors. Large mechanical equipment items like air-cooled chillers are the primary source of data center noise, and accurate knowledge of their noise emission characteristics is critical in the design of appropriate engineering controls. The total sound power level of large outdoor air-conditioning equipment can be quantified using the AHRI 370 standard, which provides methods to estimate sound power using pressure and intensity field measurements. While both field methods can estimate the total sound power level, the intensity method can also quantify the power on each radiating plane, for later computer modeling of source directivity. Additional useful information can be determined from intensity measurements using holography techniques that map the local intensity to a visual image. This paper will present a "wish list" for vendor equipment sound data to address questions of source directivity, low-frequency noise, and one-third octave-band assessment methods. We will share our recent experience conducting AHRI 370 measurements using pressure, intensity, and near-field acoustic holography methods.

8:05

1aAAa4. Noise control strategies for Data Centers near residential communities. Joshua Cassarino (Trinity | Cerami, 1001 Ave. of the Americas, 4th Fl., New York, NY 10018, jcassarino@ceramiassociates.com), Gregory A. Miller (Trinity | Cerami, Chicago, IL), and Lucas Schwartz (Trinity | Cerami, New York, NY)

As growing demand for computing power and data storage grows, the proliferation of large commercial Data Centers has brought new and growing noise concerns to residential communities around the United States. Drawing on the authors' experience over the past several years in Data Center noise control, this presentation will discuss some of the most objectionable noise sources (such as cooling and emergency power systems), means of controlling noise from these sources, and issues related to establishment of noise control criteria with Authorities Having Jurisdiction (AHJ's). Discussions will include the application of local noise ordinances written to address less robust noise sources, differences between land proffers and noise ordinances (including ways that well-crafted proffers can be negotiated with AHJ's to protect both developers and residents), and challenges of enforcement of agreed-upon criteria even with motivated AHJ's. As much of the work with AHJ's involves the input of local residents, discussion will also address means of educating residents regarding potential acoustic criteria.

8:25

1aAAa5. A new low-noise heat pump for single family homes. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Because of the effects of climate change, there is a recent trend to move away from natural gas and oil furnaces for heating homes and move toward all-electric systems. Split system heat pumps provide an efficient and effective way to both heat and cool single family residences. Typicallly, the furnace is located inside the garage and the outdoor unit is located on the side of the house so it is not visible from either the front or the back yard. The noise generated by the outdoor units located on the side of the house will often exceed the local noise ordinance sound level limit at the property line. This is particularly important in colder climates for heat pumps because the noise ordinance limit is usually lower during the nighttime hours when the heat pump will be operating in the heating mode. This presentation will discuss a new heat pump series that has noise levels 10–15 dBA lower than previous designs of comparable capacity. Using these low-noise heat pumps in single family residential areas will often eliminate the need for additional noise barriers around the equipment. A case study will be presented where a 4-ton heat pump was able to meet the 45-dBA nighttime noise ordinance limit with the unit located on the side of the house 10 feet from the property line.

Contributed Paper

8:45

1aAAa6. Field measurements to determine potential conversion factor for terminal units' combined path sound pressure data to sound power spectra. Spencer Zack (Resonance Acoust., 364 Bush St., Fl. 2, San Francisco, CA 94104, szack@resonanceac.com) and Randy Waldeck (Resonance Acoust., San Francisco, CA)

Accurately predicting mechanical noise levels requires reliable sound power spectra of mechanical equipment. However, some manufacturers of electric fan-powered units publish only sound pressure level data, commonly citing the JIS B8616 standard without strictly adhering to its procedural requirements. This practice combines radiated, inlet, and discharge noise into a single dataset. It becomes necessary to determine the sound power level for each path (i.e., discharge, inlet, and radiated) from this single sound dataset. Converting sound pressure spectra to sound power by applying basic acoustic principles can yield inconsistent results lacking precision. This study uses field-measured HVAC data to establish relationships between back-calculated sound power levels and manufacturer-provided "combined" sound pressure data.

Invited Papers

9:05

1aAAa7. Chilled beam systems—Quiet, but need fresh air. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

Chilled beam systems can have several formats, most of which are very quiet, but there can be accompanying noise concerns. In some systems, small tubes carrying cooled water are attached to metal panels, which act to transfer thermal energy from rooms; this is very quiet, plus the metal panels can be perforated and provided with sound-absorptive backings for room treatments. Some systems use fins that both carry the water and transfer the thermal energy. However, occupied spaces need fresh air, whose delivery can result in unexpected levels or undesirable qualities of noise. Several examples of chilled beam systems will be discussed, including "active" chilled beams, which integrally also provide fresh air.

9:25–9:45 Break

9:45

1aAAa8. Survey of exhaust and relief air noise control in performing arts and academic atria. Robert D. Miller (Threshold Acoust., LLC, 141 W. Jackson Blvd, Ste. 2080, Chicago, IL 60604, rmiller@threshold.llc), Shane J. Kanter (Threshold Acoust., LLC, Chicago, IL), Carl P. Giegold (Threshold Acoust., LLC, Evanston, IL), and Scott Pfeiffer (Threshold Acoust., LLC, Chicago, IL)

Simple smoke hatches are increasingly deemed not adequate for the evacuation of smoke in public gathering areas. Project design requirements for CFD simulations using highly combustible sources have pushed mechanical engineers in many cases to design mechanically ventilated systems. But how do these perform in the real world and why are they often quieter than performance data may indicate? What additional benefits for interior acoustics may be afforded by these large shafts? Examples from performing arts and academic atria will be provided.

10:05

1aAAa9. Designing from first principles to control floor vibration from mechanical equipment. Christina Young (Papadimos Group, PO Box 1025, Menlo Park, CA 94026, cyoung@papadimosgroup.com), Roman Wowk, and Jenny Solheim (Papadimos Group, San Rafael, CA)

This presentation outlines a method for predicting floor vibration due to mechanical equipment using the structural frequency response function (FRF) and dynamic forces produced by the mechanical equipment. The goal is to develop more reliable and cost-optimized designs, in contrast to conventional approaches that prescribe equipment vibration isolators based on equipment type, horsepower, and structural column spacing. Accurate structural FRF predictions are possible using finite element analysis (FEA), provided that the analysis methodology is validated against measured data from comparable existing structures. At this time, dynamic force data are generally unavailable from equipment manufacturers. However, rough estimates are possible by measuring mechanical vibration and the structural FRF of an existing installation. While this is currently a field method with several limitations, it should ultimately be developed into a standardized laboratory method manufacturers use to provide force data. With increasing construction costs and a trend toward placing mechanical equipment near vibration sensitive areas, there is now an opportunity to use experimental and analytical methods to optimize designs rather than relying on generic approaches that can be overly conservative or miss important details.

10:25

1aAAa10. Noise control solutions: Understanding why to create netter solutions. Erik Miller-Klein (Tenor Eng. Group, 11514 Dayton Ave N, Seattle, WA 98133, erik.mk@tenor-eng.com)

This session will show and discuss how advanced testing and commissioning techniques can help create more effective and efficient acoustical designs. The examples will focus on the uses of structure-borne noise/vibration assessments and photo-enabled sound intensity to develop holistic solutions. Through testing and remediation engineering, these tools have shown opportunities for prototyping and mock-up validation that can be used to optimize noise control solutions.

Session 1aAAb

Architectural Acoustics: Acoustics of Sustainable Building Assemblies and More I

Jonathan M. Broyles, Cochair
Civil, Environmental and Architectural Engineering, The University of Colorado Boulder, 13582 Via Varra, Broomfield, CO 80020

Arthur W. van der Harten, Cochair
Acoustics, Acoustic Distinctions / Open Research in Acoustical Science and Education, 400 Main St. Ste. 600, Stamford, CT 06901

Chair's Introduction—7:55

Invited Papers

8:00

1aAAb1. What's happening to acoustics in LEED v5? Kristen Murphy (Acentech, 33 Moulton St., Cambridge, MA 02144, kmurphy@acentech.com)

The Leadership in Energy and Environmental Design (LEED) green building certification, now pursued on hundreds of thousands of projects across the globe, is transitioning to the new LEED v5. Although acoustic design has been recognized by LEED for several of these iterations, the proposed LEED v5 appears to bring several changes compared to previous versions. This presentation will examine the evolution of acoustic criteria in LEED and will provide a detailed discussion of the expected changes to acoustics in LEED v5. This talk will provide examples where previous versions of LEED have led to positive acoustical design outcomes and challenges in architectural projects, discuss the new acoustical requirements in context of the broader needs and goals in the development of LEED v5, and introduce proposed strategies in working with the new requirements on architectural projects. Let's continue to learn how to use our green building standards to help projects deliver high performance for building occupants.

8:20

1aAAb2. Aligning acoustics, sustainability, and well-being through Standards and Guidelines. Viken Koukounian (Parklane Mech. Acoust., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@parklanemechanical.com)

Acoustics play a critical role in sustainable and human-centered design, yet the processes for integrating them into early design stages are often overlooked. This session explores how standards and guidelines—such as those by ISO, ASHRAE, and USGBC—are being leveraged to deliver more robust frameworks, providing structured approaches for benchmarking performance and evaluating indoor environmental quality (IEQ). By aligning acoustical design with broader sustainability and well-being goals, the discussion highlights strategies to address occupant health, productivity, and environmental performance. Emphasis will be placed on integrating acoustical mapping, life-cycle analysis, and performance metrics into sustainability-focused projects, demonstrating how standards and guidelines can inform decision-making. Attendees will leave with insights into applying these tools to streamline acoustical design, improve IEQ performance, and support emerging priorities around ESG and the UN Sustainable Development Goals (SDGs).

8:40

1aAAb3. Perceived and predicted acoustic comfort in university classrooms in the framework of the overall indoor environmental quality monitoring. Virginia I. Fissore, Tugana Aydin (Dept. of Energy, Politecnico di Torino, Turin, Italy), Pietro Chiavassa (Dept. of Control and Comput. Eng., Politecnico di Torino, Turin, Italy), Giuseppina E. Puglisi (SAIL Dept. University Sustainability, Res. Infrastructures and Labs., Politecnico di Torino, Turin, Italy), Gustavo A. Ramirez Espinosa (Dept. of Electronics, Pontificia Universidad Javeriana, Bogotá, Colombia), Louena Shtrepi (Dept. of Energy, Politecnico di Torino, Torino, Italy), Antonio Servetti, Bartolomeo Montrucchio (Dept. of Control and Comput. Eng., Politecnico di Torino, Turin, Italy), Franco Pellerey (Dept. of Mathematical Sci., Politecnico di Torino, Turin, Italy), Anna Pellegrino (Dept. of Energy, Politecnico di Torino, Turin, Italy), and Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Corso DC degli Abruzzi, 24, Torino 10129, Italy, arianna.astolfi@polito.it)

Indoor environmental quality (IEQ) monitoring is state of the art in many new buildings to optimize their energy efficiency and improve the comfort and performance of the occupants. This study focuses on the IEQ monitoring and perceived comfort in four newly built university classrooms of the same size but with different façade orientation. IEQ monitoring was performed in summer and winter periods by means of devices mounted on the internal walls of the classrooms, while a questionnaire collected students' feedback. Objective and subjective data were processed to obtain a percentage of predicted and perceived comfort, respectively, for each domain. For acoustic comfort, A-weighted sound pressure level (SPL) was collected during the lessons and a minimum threshold was set to ensure a speech-to-noise ratio of 15 dB. A correction based on experimental data was applied to obtain the SPL at the center of the room from

the SPL flush to the wall. Results revealed a good agreement between perceived and predicted acoustic comfort and that students were more satisfied with the acoustic and visual domains than with thermal and air quality. Perceived thermal comfort resulted strongly dependent on the classroom façade orientation, despite no difference was highlighted between the objective data.

9:00

1aAAb4. Investigating architectural acoustics modeling of Warner Theatre's rehearsal room. Dante N. Christian (Phys., Pomona College, 5794 Epping Ct, Douglasville, GA 30135, chefdelroy1@gmail.com), Jonah Sacks, Khaleela Zaman, and Robert Connick (Acentech, Cambridge, MA)

Dante Christian participated in a 10-week internship at Acentech. The research project that Dante landed on was an auralization of the Warner Theater's rehearsal room—a structure in Erie, PA. Two of his mentors from Acentech visited to make reverberation time measurements and to ensure that the rehearsal room was behaving as it was intended to. Thus, these measurements served as the foundation of his project's structure. For Dante's summer research, he architecturally modeled in SketchUp and in Trebel (a geometric and wave-based solver) to gather and simulate impulse responses for the Warner Theater's Rehearsal room. Through making this model, he calibrated it to the actual space to observe why reverberation times were as low as they were during data acquisition. He worked iteratively to assign absorption and scattering coefficients to the model that best fit the measured experimental results. This is because, especially, absorption coefficients of every material in the room were not specified, rendering it more difficult for Dante to calibrate his model to the space. Overall, this research project concluded in Dante finding that the space was not performing as intended and, regardless of the materials, the composition of such materials matters for good acoustics.

9:20–9:40 Break

Contributed Papers

9:40

1aAAb5. Acoustic assessment of shared workspaces. konca saher (Interior Architecture and Environ. Design, Kadir Has Univ., kadir has caddesi, Cibali, Istanbul, Fatih 34083, Turkey, konca.saher@khas.edu.tr), dilara kelle (Art and Design Faculty, Kadir Has Univ., Istanbul, Turkey), and Ozge Ustundag Ganic (Kolektif House, Istanbul, Turkey)

With increasing mobility and rising costs, shared workspaces have become popular alternatives to traditional fixed-desk open-plan offices (OPOs). This study examines differences in acoustic attributes, specifically noise levels and noise disturbance, between shared workspaces with temporary and flexible usage patterns and fixed-desk OPOs. Additionally, it explores task types and their mental workload in shared workspaces. The research was conducted in two shared open-plan offices of similar size (350 m^2) in Istanbul: one labeled "normal" and the other "silent." Occupied noise measurements were complemented by subjective user feedback from 35 participants using the Gêne Acoustique dans les Bureaux Ouverts (GABO) questionnaire, as specified in BS ISO 22955-2021, expanded with the NASA-Task Load Index (TLX). Results revealed that, while noise levels were similar to traditional OPOs, the "silent" office exhibited higher noise levels than the "normal" office. Interestingly, louder activities, such as phone calls and video meetings, were more frequently conducted in the "silent" office. These findings highlight the complex relationship between acoustic environments, user behavior, and task characteristics, providing insights for improving shared workspace design.

10:00

1aAAb6. Experimental investigation of the acoustics and visual comfort inside a library building, KFUPM. Sara Yousaf (Architectural Eng., King Fahd Univ. of Petroleum and Minerals, Academic belt Rd., Dhahran, Eastern Province 32227, Saudi Arabia, bibisara576@gmail.com)

This study evaluates the lighting and acoustics of King Fahd University of Petroleum and Minerals' (KFUPM) major library, a vital academic area for the well-being and productivity of students. Subjective questionnaires and objective measures were used in two floors of research to assess interior comfort in study halls, conference rooms, entrance lobbies, and stairwells. According to survey data, people were generally satisfied with the acoustics, but not with the lighting. Carpeting was a good way to cut down on noise, but some mechanical systems made it harder to focus. Although sections close to equipment needed improvement, sound levels were within permissible bounds. The architectural layout of the library, which was meant to improve natural light and ventilation, resulted in uneven lighting, little sunshine penetration, and glare problems. The lack of proper illumination on bookshelves and numerous broken lighting fixtures reduced the visual comfort. Improved ergonomics to increase brightness, brighter furniture and wall colors, and better lighting maintenance are among the suggestions. Updates to the decor and more spotlights on the bookshelves could improve the atmosphere even further. In line with BS 8233:2014, BB93 (2015), and ISO 8995:2005 standards, these findings highlight the significance of optimal lighting and acoustics in academic settings and provide useful advice for library design enhancements that promote student productivity and well-being.

10:20

1aAAb7. Sustainable metamaterial: An integrated approach to building noise control and natural ventilation. Junhyeok Choi (School of Architecture, Virginia Tech, 1400 Seneca Dr., Blacksburg, VA 24060, junhyeok@vt.edu)

Natural ventilation in buildings is essential for improving the indoor air quality and thermal comfort; however, openings for natural ventilation in façades can allow outdoor noise to interfere with indoor acoustic comfort. This pilot study aims to reconcile this inverse relationship. Metamaterials have been developed as promising acoustic-ventilation materials in mechanical engineering and material science, but their application in architecture remains underexplored. This research proposes a ventilated acoustic metamaterial (VAM) to be integrated in building facades. Ventilation efficiency will be analyzed using Rhino Computational Fluid Dynamics on 3-D digital samples modeled in Rhino and Grasshopper. Samples demonstrating adequate ventilation performance will be fabricated through additive manufacturing. Acoustic performance, including sound transmission loss (STL), will be measured using an impedance tube. By proposing a ventilated acoustic metamaterial, this research advances the potential of metamaterials in architectural applications, enhancing acoustics and indoor air quality. Future studies will focus on scaling the material to panel size and establishing criteria like visual appeal, durability, and stiffness to ensure feasibility as a façade material.

10:40

1aAAb8. Renewable energy sources and noise control—Assessment of noise emissions during construction and operation of renewable energy facilities in urban areas. Alexander Müller (Architecture and Civil Eng., Tech. Univ. pf Appl. Sci. Würzburg-Schweinfurt, Würzburg, Germany) and Normen Langner (Architecture and Civil Eng., Tech. Univ. pf Appl. Sci. Würzburg-Schweinfurt, Röntgenring 8, Faculty of Architecture and Civil Eng., Würzburg 97070, Germany, normen.langner@thws.de)

To achieve climate protection goals, it is essential to adopt renewable energy sources. However, specific projects and measures for the generation and distribution of electricity and thermal energy often face local opposition

due to noise emissions from the facilities. Shifting energy and heat production from large, remote facilities to decentralized generation units closer to urban or within residential areas may result in noise emissions that lead to conflicts with nearby residents. Based on a literature review, this study examines potential noise impacts associated with various renewable energy sources (e.g., wind turbines, geothermal systems, heat pumps, biogas plants) during both the construction and operational phases. For instance, in the case of wind turbines, aerodynamic noise generated by the rotor blades during operation is predominant, whereas geothermal systems can cause significant noise emissions during drilling activities. Against this backdrop, the paper identifies technical and planning measures (e.g., sound insulation, optimized facility placement) that can mitigate noise development. The findings of this study demonstrate that climate protection objectives and high standards for noise control can be achieved together. This requires adherence to established and well-known principles of noise protection in the design, planning, and construction of renewable energy facilities.

MONDAY MORNING, 19 MAY 2025 GALERIE 4, 7:55 A.M. TO 11:00 A.M.

Session 1aAB

Animal Bioacoustics: Progress on Bioacoustics of Fish I

Kelly S. Boyle, Cochair
Biological Sciences, University of New Orleans, Department of Biological Sciences, University of New Orleans, 2000 Lakeshore Drive, New Orleans, LA 70148

John S. Allen, Cochair
Mechanical Engineering, University of Hawaii Manoa, Holmes 302, 2540 Dole Street, Honolulu, HI 96822

Chair's Introduction—7:55

Invited Papers

8:00

1aAB1. Vessel noise impacts on sciaenid fishes of the northern Gulf of Mexico coast. Kelly S. Boyle (Biological Sci., Univ. of New Orleans, Dept. of Biological Sci., University of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, ksboyle@uno.edu), Ariel N. Alonso (Biological Sci., Univ. of New Orleans, New Orleans, LA), Gina A. Badlowski (Biological Sci., Florida Int. Univ., Miami, FL), and Bennett H. Price (Biological Sci., Univ. of New Orleans, New Orleans, LA)

Many fish species in the family Sciaenidae produce sounds during reproduction. Variability among sciaenid species in the season of peak spawning, sound frequency spectra, and hearing range and sensitivity may result in species-specific vessel noise impacts, as vessel traffic is more prevalent in the summer and may impart hearing loss differentially among frequencies. We have examined the potential of noise to disrupt communication for sciaenid fishes in the northern Gulf of Mexico with a multidisciplinary approach using passive acoustic monitoring (PAM) and laboratory experiments. PAM observations of fish choruses in association with vessel noise indicate that Red Drum (*Sciaenops ocellatus*), Spotted Seatrout (*Cynoscion nebulosus*), and Silver Perch (*Bairdiella chrysoura*) exhibit modest reduction in chorus intensity, consistent with a prediction of disturbance in the presence of vessel noise. Lab experiments with Atlantic Croaker (*Micropogonias undulatus*) indicate a reduction in hearing sensitivity following noise exposure. Hair cells of the sensory epithelium of the saccule in Atlantic Croaker ears proliferate with growth, but following noise exposure in the lab, hair cell proliferation rates do not increase. Together, these results indicate that acoustic communication in sciaenid fishes may be impacted by reducing signal-noise, lowering calling activity, and decreasing acoustic sensitivity of listening fish.

8:20

1aAB2. Identifying unknown fish choruses from the Atlantic Ocean off North Carolina USA with a historical soundscape comparison. Joseph J. Luczkovich (Biology, East Carolina Univ., Dept of Biology, East Carolina University, Greenville, NC 27858, luczkovichj@ecu.edu) and Mark W. Sprague (Phys., East Carolina Univ., Greenville, NC)

We deployed a wave glider with a passive acoustic recorder off North Carolina and recorded the soundscape for 8 days in August 2017. These recordings were made 74 years after the geophysicist Milton Dobrin reported on fish sounds that he and the US Navy Ordinance Laboratory recorded in the same coastal area and at the same time of year during World War II. Here we will review the information we collected and compare it with the historical records provided by Dobrin and the published Smithsonian tape recordings from the US Navy. Dobrin attributed the peak at 2400 Hz to the bastard trout *Cynoscion nothus* (Sciaenidae). We think this identification is

incorrect based on the frequencies reported, which are due to other species (striped cusk-eel *Ophiodon marginatum* and an unknown species that could be the Atlantic midshipmen *Porichthys plectrodon*). We identified a mixed chorus of multiple species of Sciaenidae in the frequency range of 300–600 Hz, which we attributed to weakfish (*Cynoscion regalis)* and Atlantic croaker (*Micropogonias undulatus*). However, we also recorded another fish chorus we termed the "unknown grunt" at 500 Hz. We don't know which species produced this sound, but putative sound-producing species will be discussed.

8:40

1aAB3. The Estuarine Soundscape Observatory Network in the Southeast—Ten plus years of listening to fish in estuaries of South Carolina, USA. Eric W. Montie (Natural Sci., Univ. of South Carolina Beaufort, One University Boulevard, Bluffton, SC 29909, emontie@uscb.edu) and Alyssa Marian (Natural Sci., Univ. of South Carolina Beaufort, Bluffton, SC)

The Estuarine Soundscape Observatory Network in the Southeast (or ESONS) monitors biological sound and human-made noise in four estuaries of South Carolina (SC) that vary in human activity. The long-term goal is to "eavesdrop" on key behaviors of marine animals that can change rapidly or gradually in response to environmental changes and human impacts, thus providing a measure of resilience or shifting baselines in a globally changing environment. While tracking the sounds of fish, we observed the temporal synchrony of courtship behavior in black drum, oyster toadfish, silver perch, spotted seatrout, and red drum across estuaries in SC. We found that in years with warmer springs, some species chorused earlier and had longer chorusing seasons than in the years with cooler temperatures. Inversely, cooler temperatures during late summer led to earlier and longer spawning seasons for red drum. Through our seining program, we detected the appearance of young-of-the-year (YOY) approximately 1 month after initiation of the chorusing season. Additionally, we found positive correlations between chorusing and YOY abundance. Current work is exploring chorusing variability among years, to assess the impacts of temperature anomalies and extreme events, and across estuaries, to assess the effects of vessel noise on fish reproduction.

9:00

1aAB4. Identifying fish sounds underwater with spatial audio and 360° video. Marc Dantzker (FishEye Collaborative, FishEye Collaborative, Arlington, VA, mdantzker@fisheyecollaborative.org), Matthew Duggan (Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Erika Berlik (FishEye Collaborative, Arlington, VA), Symeon Delikaris-Manias, Vasileios Bountourakis, Ville Pulkki (D Information and Communications Eng., Aalto Univ., Espoo, Finland), and Aaron N. Rice (Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Despite the proliferation of passive acoustic monitoring (PAM) recordings in marine and freshwater habitats around the world, identifying which species produce which sounds has been a pervasive and fundamental challenge, especially for fishes. The overwhelming majority of sounds from these recordings remain unidentifiable to a taxonomic level beyond "fish" and are best categorized only by sound characteristics. Most techniques for sound I-D in nature have a high probability of misattribution or have not proven to be scalable. We have developed a scalable method for *in situ* fish sound identification in nature by overlaying visualizations of spatial audio onto 360 deg video recordings. In Curaçao, we have developed an extensive collection of identified sounds from reef-associated Atlantic fishes, and a starting point for an open-access library of fish sounds (fisheyecollaborative.org/library). We intend to expand this approach globally to enhance taxonomic resolution for PAM recordings, providing critical validated information for developing the detection models required for large scale analysis. Identified sounds can help enable PAM to become a transformational tool for fish biology.

9:20–9:40 Break

Contributed Papers

9:40

1aAB5. Approaches to understanding anthropogenic particle motion and substrate-borne vibration effects on fishes and invertebrates. Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, 45600 Woodland Rd., Ste. #455-C33, Sterling, VA 20166, guan@cua.edu), Arthur N. Popper (Univ. of Maryland, College Park, MD), and Louise Roberts (Univ. of Liverpool, Liverpool, United Kingdom)

The past decade has seen a substantial increase in awareness of potential impacts from underwater anthropogenic sound on fishes and aquatic invertebrates. However, many efforts addressing these issues focused on species and research questions that are not relevant to environmental or regulatory concerns. Additionally, some of the experimental designs and settings used in earlier studies are not really appropriate to answer these questions properly or fully. Accordingly, this presentation makes several recommendations on research approaches to best understand potential impacts of anthropogenic particle motion and substrate-borne vibration on ecologically or economically important fishes and aquatic invertebrates. In particular, focus is on species of special concern in terms of regulatory issues. Among these recommendations, three broad perspectives that encompass several key research questions are highlighted, along with a preliminary analysis of four general experimental settings that could be used to address these research questions. Our analyses show that open-water experimental setting with real sound sources is the most effective approach to address hearing and hearing effects that are critical to regulatory questions and that open-water setting with free-ranging animals is the best way to answer questions related to behavioral and physiological effects.

10:00

1aAB6. Tomographic imaging using an experimental standing wave tube-like setup to explore moving fish hearing structures. Isabelle P. Maiditsch (Systematic Zoology, Ludwig-Maximilians-Univ. Munich, Großhaderner Strasse 2, Planegg, Bavaria 82152, Germany, maiditsch@bio.lmu.de), Tanja Schulz-Mirbach (Systematic Zoology, Ludwig-Maximilians-Univ. Munich, Munich, Germany), Friedrich Ladich (Dept. of Behavioral and Cognit. Biology, Univ. of Vienna, Vienna, Austria), Marco Stampanoni (Photon Sci. Dept., Paul Scherrer Institut, Zürich, Switzerland), Martin Hess (Systematic Zoology, Ludwig-Maximilians-Univ. Munich, Munich, Germany), and Christian M. Schlepütz (Photon Sci. Dept., Paul Scherrer Institut, Villigen, Switzerland)

Modern bony fishes show significant variation in the morphology of their hearing structures, which have been studied for almost two centuries. However, the exact processes of sound transmission, interaction, and their contributions to fish hearing are still matters of assumption. Characterizing the sound-induced *in situ* motion of fish auditory structures and their interplay while avoiding invasive damage is very challenging. A new synchrotron radiation-based tomography setup characterizes the motion of these

structures (swimbladder, auditory ossicles, inner ear otoliths), allowing *in situ* morphofunctional studies with unprecedented resolution. We developed a miniature standing wave tube to meet the needs of tomography and tank acoustics, enabling control over the acoustic field. Sound Pressure Levels can be determined and adjusted during tomographic measurements, generating sound pressure or sound-induced particle motion at the test subject, and the setup reliably produces frequencies of up to 2 kHz. We tested six otophysan species, including the glass catfish (*Kryptopterus vitreolus*) and zebrafish (*Danio rerio*), and were able to visualize the dynamic interactions and movements of fish auditory structures in response to pure tone stimuli. Our approach will enhance our understanding of fish sound detection and set a new standard for non-invasive, high-resolution imaging techniques in the field of aquatic sensory biology.

10:20

1aAB7. Fish chorusing patterns and their environmental drivers in California National Marine Sanctuaries. Ella B. Kim (Scripps Inst. of Oceanogr., Univ. of California San Diego, 8635 Kennel Way, Ritter Hall, La Jolla, CA 92037, ebkim@ucsd.edu), Annebelle Kok (Univ. of Groningen, Groningen, Netherlands), Emma Beretta (California Polytechnic State Univ. San Luis Obispo, San Luis Obispo, CA), Emily Donahue (California State Univ. Monterey Bay, Monterey Bay, CA), Leila Hatch (National Oceanic and Atmospheric Administration, Office of National Marine Sanctuaries, Scituate, MA), John Joseph, Tetyana Margolina (Naval Postgrad. School, Monterey, CA), William Oestreich (Monterey Bay Aquarium Res. Inst., Moss Landing, CA), Lindsey Peavey Reeves (National Oceanic and Atmospheric Administration, Office of National Marine Sanctuaries, Silver Spring, MD), John P. Ryan (Res., MBARI, Moss Landing, CA), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Marine soundscapes are dominated by fish choruses, when many fish vocalize concurrently, often for mating purposes. Passive acoustic monitoring (PAM) allows us to analyze spatiotemporal patterns of fish chorusing, identifying breeding grounds, species distributions, and mating seasons. By integrating PAM with environmental data (temperature, salinity, etc.), we can assess the environmental drivers of fish chorusing. Through the Sanctuary Soundscape Monitoring project, we collected PAM data across nine sites in Monterey Bay (MBNMS), Chumash Heritage (CHNMS), and Channel Islands National Marine Sanctuaries (CINMS), each recording for ~2 years for a cumulative 17.9 years. We identified: (1) WHO: five fish choruses, including plainfin midshipman, bocaccio rockfish, white seabass, and two unidentified species; (2) WHERE: spatial variation in chorus types and occurrence; (3) WHEN: predominantly nocturnal, seasonal chorusing aligned with reproductive cycles; and (4) DRIVERS: environmental variables associated with different water masses drive chorusing presence, and fish tended to chorus more during marine heatwaves. Through non-invasively listening to fish, we gain critical insights into their reproductive behavior and environmental drivers, to better inform effective management and conservation, particularly under changing environmental conditions.

10:40

1aAB8. Looking for the elusive fish farts: Can we monitor Atlantic herring in the Gulf of Maine with passive acoustics? Xavier Mouy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, xavier.mouy@whoi.edu), Cameron Thompson (Northeastern Regional Assoc. of Coastal Ocean Observing Systems, Portsmouth, NH), Samantha Tolken, Tammy Silva (NOAA Stellwagen Bank National Marine Sanctuary, Scituate, MA), Amanda Holdman, Jessica McCordic, J Michael Jech (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA), Jennifer Miksis-Olds, Ian T. Jones (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, Durham, NH), Jooke Robbins (Ctr. for Coastal Studies, Provincetown, MA), Laura Howes, Pete DeCola, Alice Stratton, Michael Thompson (NOAA Stellwagen Bank National Marine Sanctuary, Scituate, MA), Jackie Motyka (Northeastern Regional Assoc. of Coastal Ocean Observing Systems, Portsmouth, NH), Sofie Van Parijs (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA), and Leila Hatch (NOAA Office of National Marine Sanctuaries, Scituate, MA)

Herring (*Clupea pallasii* and *Clupea harengus*) produce Fast Repetitive Tick (FRT) sounds by expelling air bubbles in the water through their anal duct. Theoretical and experimental studies have shown that FRT sounds from compact schools of herring have the potential to be detectable up to 1 km away from an acoustic recorder in calm conditions. While FRT sounds have been recorded in the wild in the Northeast Pacific and in the Northeast Atlantic, they have never been detected in the Northwest Atlantic where herring are also present. In this study, we investigated whether passive acoustics can be used to detect the presence of Atlantic herring FRT sounds in the Gulf of Maine to help inform fisheries management measures in the region. We examined passive acoustic data collected in the Gulf of Maine since 2014, focusing on the detection of individual FRT sounds, changes in the overall soundscape, and feeding sounds from herring predators. To manage the large dataset, we narrowed down our temporal and spatial search using data from NOAA bottom trawl surveys, long-term echosounder recordings from bottom landers, herring landings from commercial fishing vessel trip reports, visual sightings of whales, and observations of marine mammals feeding on herring.

Session 1aAO

Acoustical Oceanography: Topics in Acoustical Oceanography I

Derek Olson, Chair
Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Road, Monterey, CA 93943

Contributed Papers

8:40

1aAO1. Improving MAMBAT: Toward fully automated model-based tracking of multiple marine mammals. Eva-Marie Nosal (Ocean & Resources Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, nosal@hawaii.edu), Pina Gruden (Pina Gruden s.p., Kranj, Slovenia), and Elizabeth Henderson (Whale Acoust. Reconnaissance Project, NIWC Pacific, San Diego, CA)

We developed MAMBAT (Multiple-Animal Model-Based Acoustic Tracking) to fully automate the process of model-based passive acoustic methods for tracking multiple marine mammals. We previously demonstrated MAMBAT on one- and four-animal datasets of clicking sperm whales recorded on bottom-mounted hydrophones at the Atlantic Undersea Test and Evaluation Center in the Bahamas. This presentation will review the motivation, theory, and application of MAMBAT and discuss the improvements that we made to MAMBAT to track different species in increasingly challenging scenarios (e.g., more animals and lower signal-to-noise situations). Tracking results using bottom-mounted hydrophones from the Pacific Missile Range Facility in Hawaii will be used to demonstrate the improvements. [Work supported by the ONR Marine Mammals and Biology program.].

9:00

1aAO2. Acoustic analysis of ice fracturing on a shallow freshwater lake. John Case (Graduate Program in Acoust., Penn State, 414 ECoRE Bldg., 556 White Course Dr., University Park, PA 16802, jac7175@psu.edu), Daniel C. Brown (Penn State Univ., State College, PA), and Andrew Barnard (Michigan Tech, Houghton, MI)

Ice sheet fracturing is a process that can indicate the intensity of ice breakup and overall ice safety. Thus, ice fracturing is of interest to a wide range of groups, including climate researchers, fisheries management, and cultural centers. Changes in weather conditions cause stresses within the ice sheet. When these stresses exceed the ultimate tensile or shear strength of the ice, fractures may occur, generating acoustic emissions into the water column and elastic waves through the ice. To study this process, a week-long experiment was conducted on a shallow freshwater lake. Hydrophones, microphones, and geophones were used to record acoustic data across a wide range of weather conditions. In addition to ambient measurements, a series of force hammer blows were used to quantify ice-sensor responses.

9:20

1aAO3. Comparing the acoustic response of bubbles adhered to artificial seagrass blades with models for freely oscillating bubbles. Mel C. Chen (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 9289 S Scholars Dr., La Jolla, CA 92093, amc005@ucsd.edu), Allison Bickford, Hailey Gilman, Langdon Tarbell (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), Tracy Mandel (Dept. of Mech. Eng., Univ. of New Hampshire, Durham, NH), and Gabriel R. Venegas (Cntr. for Acoust. Res. and Edu. and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH)

Seagrass beds are threatened coastal ecosystems that play a significant role in carbon sequestration despite covering a small fraction of Earth's oceans by area. During photosynthesis, oxygen bubbles fill channels within the seagrass blades, nucleate onto the blades' surface, and rise freely through the water column. Since bubbles interact strongly with acoustic fields, acoustics have been used to monitor seagrass photosynthesis. However, many acoustics models for bubbly liquids only consider freely oscillating bubbles. To determine the extent of these models' applicability to seagrass monitoring, we conducted a controlled laboratory experiment isolating the effect of bubbles adhered to seagrass blades on the propagation and backscatter of acoustic signals. After oxygenating a tank containing a bed of artificial seagrass, we used an ES200-7CD transducer, RS-4034 hydrophone, and EK80 transceiver to measure acoustic backscatter, phase speed, and attenuation at several points in time after oxygenation. A calibrated Olympus Tough TG-6 digital camera and custom softbox were used to concurrently measure the radius distributions of bubbles adhered to the blades. We used these distributions to predict the acoustic response using models for freely oscillating bubbles. A comparison between these predictions and the measured acoustic response will be discussed. [Work supported by ONR and NSF.]

9:40–10:00 Break

10:00

1aAO4. Sub-bottom profiling using an autonomous underwater vehicle equipped with a sound source and a towed hydrophone array. Paige Pfenninger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 32 Vassar St., Marine Robotics Group, Cambridge, MA 02139, ppfenni@mit.edu) and Ying-Tsong Lin (SIO, Scripps Inst. of Oceanogr., La Jolla, CA)

Theoretical and experimental studies were conducted to investigate the feasibility of using the arrival times of acoustic signals from an AUV source to a short, 16-element towed hydrophone array to determine the sound speed and layer thickness of the seabed through Bayesian geoacoustic inversion. This method provides range-dependent geoacoustic parameters with a resolution on the order of 10 m. Numerical studies indicate that, for timing data with low variance, arrival times can be used to accurately estimate seabed properties. However, the performance of the Bayesian inversion model deteriorates as the variance of the timing data increases. Experimental data were collected during the Seabed Characterization Experiment at the New England Mud Patch. The mean and variance of the direct path, bottom, and sub-bottom timing returns were calculated using Gaussian Process Regression. The experimental data exhibit a high level of variance in the sub-bottom timing returns, likely due to the presence of scatterers in the sediment layer. Furthermore, the results show that layer thickness and sound speeds are highly coupled. Additional prior information is required to decouple the ambiguity and uniquely determine seabed properties. [Work supported by the Office of Naval Research.]

10:20

1aAO5. Pulse propagation in shallow water under conditions of horizontal refraction initiated by nonlinear internal waves. Denis Manulchev, Alexander Kaplun (Marine Geosciences, Univ. of Haifa, Haifa, Israel), and Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave. Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il)

The paper considers the variability of characteristics of a signal propagating in a shallow-water waveguide under conditions of horizontal refraction. The theory of vertical modes and horizontal space-time rays (Burridge

and Weinberg) is used to analyze changes of the signal shape, its spectrum, and features of the phase and amplitude fronts. Possible effects, such as changes in the spectrum of pulses corresponding to normal modes and rotation of the amplitude front, are demonstrated using model problems as an example. An analysis of the propagation of chirp signals (270–330 Hz) is carried out for the Shallow Water 2006 experiment, in a situation where a moving train of intense nonlinear internal waves crosses the acoustic track, and multipath effects are observed in the horizontal plane (Baiey *et al*, 2011). The modeling results are compared with experimental data. [Work supported by ISF, Grant 946/20.]

MONDAY MORNING, 19 MAY 2025 BALCONY K, 8:35 A.M. TO 11:00 A.M.

Session 1aBA

Biomedical Acoustics and Signal Processing in Acoustics: Super Resolution Ultrasound Imaging I

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via sommarive 9, Trento, 38123, Italy

Pengfei Song, Cochair
Biomedical Engineering, Duke University, 100 Science Dr, Hudson Hall Annex 276, Durham, NC 27708

Chair's Introduction—8:35

Invited Papers

8:40

1aBA1. Ultrasound Localization Microscopy (ULM) for the characterization of kidneys and breast cancer. Celine Porte, Zuzanna Magnuska (Experimental Molecular Imaging, RWTH Aachen Univ., Aachen, Germany), Thomas Lisson, jannine Salewski (Chair for Medical Eng., Dept. of Elec. Eng. and Information Technol., Ruhr Univ. Bochum, Bochum, Germany), Susanne Fleig (Dept. of Nephrology and Immunology, Univ. Hospital Aachen, Aachen, Germany), Matthias Kohlen (Dept. of Gynecology and Obstetrics, Univ. Hospital Aachen, Aachen, Germany), Uta Kunter (Dept. of Nephrology and Immunology, Univ. Hospital Aachen, Aachen, Germany), Stefanie Dencks (Chair for Medical Eng., Dept. of Elec. Eng. and Information Technol., Ruhr Univ. Bochum, Bochum, Germany), Elmar Stickeler (Dept. of Gynecology and Obstetrics, Univ. Hospital Aachen, Aachen, Germany), Georg Schmitz (Chair for Medical Eng., Dept. of Elec. Eng. and Information Technol., Ruhr Univ. Bochum, Bochum, Germany), and Fabian Kiessling (Experimental Molecular Imaging, RWTH Aachen Univ., Forckenbeckstrasse 55, n.a., Aachen 52074, Germany, fkiessling@ukaachen.de)

ULM has shown great promise in preclinical studies for characterizing tissue through its microvasculature. Proof of principle has also been demonstrated in patients, but there is still a long way to go before it will become a robust routine clinical tool. In this presentation, we report on our efforts to clinically translate ULM for the characterization of chronic kidney disease and breast cancer. We explain our approach to optimize and validate the measurement protocol and the degree of reconstruction. We then present two clinical studies: In the first, ULM was used to characterize transplant kidneys and we show the visualization of glomeruli and the correlation of ULM parameters with the renal resistive index. In the second study, we evaluate ULM for its ability to discriminate between responders and non-responders to neoadjuvant chemotherapy in breast cancer patients and show a better discriminatory power than histology. Thus, ULM is ready for implementation in clinical trials and initial data highlight its high diagnostic value for various indications.

9:00

1aBA2. Super-resolution ultrasound imaging of microvasculature: Initial experiences in preclinical and clinical applications. Chengwu Huang (Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, Huang.Chengwu@mayo.edu), U-Wai Lok, Jingke Zhang, Ryan DeRuiter, and Shigao Chen (Radiology, Mayo Clinic, Rochester, MN)

Ultrasound localization microscopy (ULM) enables imaging of the microvasculature at spatial resolutions surpassing the limits of conventional ultrasound, while maintaining comparable penetration depth. This super-resolution technique relies on the localization and tracking of individual microbubbles (MBs) administered into the bloodstream, providing detailed visualization of microvascular structures and functions. However, clinical implementation of ULM faces significant challenges, such as low signal-to-noise ratio (SNR), tissue motion artifacts, and prolonged acquisition times. In this presentation, we will discuss advancements aimed at achieving robust and reproducible clinical implementation of ULM through a series of technical innovations and optimizations. These include signal enhancement techniques to boost MB SNR, advanced algorithms for accurate MB localization and tracking at clinically relevant dosages, optimized data acquisition strategies tailored to clinical workflows, and the development of 3-D imaging capabilities. Through pilot preclinical and clinical studies, we demonstrate the feasibility and efficacy of ULM in various applications, including chronic kidney disease (CKD), kidney transplant, liver disease evaluation, and beyond, in both animal models and human subjects under clinical scanning

conditions. These findings underscore the transformative potential of super-resolution ultrasound imaging as a diagnostic tool, paving the way for widespread clinical adoption.

9:20

1aBA3. Assessing microvasculature changes using super-resolution ultrasound imaging in preclinical and clinical study. Qiyang Chen (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Zahra Hosseini (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Zhiyu Sheng, Samit Ghosh, Roderick Tan (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), and Kang Kim (Bioengineering, Univ. of Pittsburgh, 623A Scaife, 3550 Terrace St., Cardiology/Medicine, Pittsburgh, PA 15213, kangkim@upmc.edu)

Super-resolution ultrasound (SRU) imaging is an emerging technology that visualizes microvessels with unprecedented high resolution. Acute kidney injury (AKI) is a risk factor for the development of chronic kidney disease (CKD). One mechanism for this phenomenon is renal microvascular rarefaction and subsequent chronic impairment in perfusion. CKD is also common among individuals with sickle cell disease (SCD) and is a major contributor to early mortality in this population. Renal microvascular rarefaction and peritubular vascular congestion are known to be associated with progressive deterioration of renal health in SCD. SRU imaging can, therefore, be a promising diagnostic tool for kidney diseases by evaluating the changes of microvasculature. In this presentation, the feasibility and accuracy of SRU is shown in preclinical studies using mouse models of a ischemia–reperfusion injury and a heme induced CKD under an approved animal protocol. The potential clinical translation is demonstrated using a clinical ultrasound probe by assessing significant reductions in vessel density in CKD individuals compared to non-CKD subjects under the approval of institutional review board. Current technical limitations with 2-D SRU are also discussed and an optimized imaging sequence and algorithm for 3-D SRU using a row-column array (6 MHz) to overcome such limitations are introduced.

9:40–10:00 Break

Contributed Papers

10:00

1aBA4. Ultrasound matrix imaging for transcranial *in vivo* localization microscopy. Flavien Bureau (Institut Langevin, CNRS, ESPCI, PSL Univ., Paris, France), Louise Denis, Antoine Coudert (Laboratoire d'Imagerie Biomedicale, CNRS, Paris, France), Mathias Fink (Langevin Inst., ESPCI Paris, Paris, France), Olivier Couture (Laboratoire d'Imagerie Biomedicale, CNRS, Paris, France), and Alexandre Aubry (Institut Langevin, CNRS, ESPCI, PSL Univ., 1 rue Jussieu, Paris 75005, France, alexandre.aubry@espci.fr)

Transcranial ultrasound imaging is usually limited by skull-induced attenuation and high-order aberrations. By using contrast agents such as microbubbles in combination with ultrafast imaging, not only can the signal-to-noise ratio be improved, but super-resolution images down to the micrometer scale of the brain vessels can be obtained as well. However, ultrasound localization microscopy (ULM) remains impacted by wave-front distortions that limit the microbubble detection rate and hamper their localization. In this work, we show how matrix imaging, which relies on the prior recording of the reflection matrix, can provide a solution to those fundamental issues [F. Bureau, J. Robin, W. Lambert, M. Fink, and A. Aubry, Nature Commun. **14**, 6793 (2023)]. As an experimental proof-of-concept, an *in vivo* reconstruction of deep brain microvessels is performed on three anesthetized sheeps [F. Bureau, L. Denis, A. Coudert, M. Fink, and O. Couture, A. Aubry, arXiv:2410.14499 (2024)]. The compensation of wave distortions is shown to drastically enhance the contrast and resolution of ULM. This experimental study thus opens up promising perspectives for a transcranial and non-ionizing observation of human cerebral microvascular pathologies, such as stroke.

10:20

1aBA5. Ultrasound Localization Microscopy with reduced frame rates and short acquisition times using bidirectional 2-D interpolation. Sajjad Afrakhteh (Univ. of Trento, Trento, Trentino, Italy), Giulia Tuccio (DISI, Univ. of Trento, via Sommarive, 5, Povo, Trento 38123, Italy, giulia.tuccio@unitn.it), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy)

Ultrasound Localization Microscopy (ULM) enables high-resolution imaging of microvasculature but faces significant challenges due to its dependency on high frame rates (in kHz) and long acquisition times, both of which limit its clinical applicability. To address these challenges, we propose a novel bidirectional Radial Basis Function (RBF)-based interpolation technique that reconstructs missing data by leveraging spatiotemporal information from orthogonal x-t and z-t planes. Results from in silico and *in vivo* datasets, including rat brain and kidney scans, show the approach preserves microvascular depiction and resolves velocity maps. Specifically, the technique enables a tenfold reduction in frame rate (from 1 kHz to 100 Hz) and a 3- to 4-fold compression of acquisition time without significantly compromising super-resolution image quality. Fourier Ring Correlation (FRC) and Dice similarity validate the effectiveness of the proposed technique. From 100-Hz frame rate, the proposed technique achieves a FRC and Dice score (for Density maps) of 18.9 μm and 0.82, respectively. For comparison, images generated at 100 Hz (before interpolation) achieve FRC and Dices score of 22 μm and 0.19, respectively (FRC at 1 kHz: 11.5 μm). By enabling low-frame-rate and short-acquisition ULM, our technique significantly enhances ULM's clinical feasibility. Future work will integrate motion correction to further optimize dynamic profiles.

10:40

1aBA6. Functional ultrasound localization microscopy on freely moving rats. Yike Wang (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801, yikew2@illinois.edu), Matthew Lowerison (Biomedical Eng., Duke Univ., Durham, NC), Bing-Ze Lin (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), Zhe Huang (Biomedical Eng., Duke Univ., Durham, NC), YiRang Shin (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL), and Pengfei Song (Biomedical Eng., Duke Univ., Durham, NC)

Functional ultrasound localization microscopy (fULM), which utilizes the neurovascular coupling effect, has become a critical tool for investigating neural activities in the rodent brain. This technique relies on intravenously injected microbubbles (MBs) as contrast agents, enabling exceptional spatial resolution. However, prior studies have predominantly applied fULM in anesthetized animals, a limitation that markedly reduces sensitivity to neural activity. Recent advancements in functional ultrasound imaging of freely moving rodents have demonstrated not only enhanced sensitivity but also an expanded application to behavioral studies, which is unattainable in anesthetized or head-fixed awake imaging setups. Nevertheless, the application of ULM in freely moving animals remains unexplored. In this study, we introduce a novel fULM imaging approach on freely moving rats. By utilizing indwelling jugular vein catheterization and a miniaturized ultrasound probe, this method achieves high spatial resolution and precise mapping of dynamic neural processes *in vivo*. We validate the system by visualizing cerebral blood flow and microvascular changes during visual stimulation, providing new insights into neurovascular coupling. This platform addresses the traditional limitations of immobilized and anesthetized imaging, offering a transformative tool for neuroscience research in naturalistic settings.

Session 1aCA

Computational Acoustics: Computational Methods I

Amanda Hanford, Chair
Penn State University, PO Box 30, State College, PA 16802

Contributed Papers

8:00

1aCA1. Road Traffic Noise Assessment Method Hong Kong (RONOSS-HK) and GPU Computation Software. Tsz Shan Viviana Tong (Environ. Protection Dept., The Government of Hong Kong Special Administrative Region of the People's Republic of China, 26/F, Southorn Ctr., 130 Hennessy Rd., Wan Chai, Hong Kong, Hong Kong, vivianatstong@epd.gov.hk), Ho Kun Vince MAK, and Chee Kwan Lee (Environ. Protection Dept., The Government of Hong Kong Special Administrative Region of the People's Republic of China, Hong Kong, Hong Kong)

Hong Kong, with its dense and complex urban landscape, has long encountered unique challenges in modeling road traffic noise for planning and assessment purposes. In response, the Environmental Protection Department (EPD) of the Hong Kong Special Administrative Region (HKSAR) Government developed an enhanced prediction methodology called the Road Traffic Noise Assessment Method—Hong Kong (RONOSS-HK). RONOSS-HK aims to improve the accuracy of noise predictions by addressing factors such as source geometry, source terms, noise propagation, considerations for new-energy vehicles, and the baffle type acoustic windows, all tailored to the region's intricate spatial noise environment. To support the adoption of RONOSS-HK, the EPD has introduced Geographic Information System (GIS) assessment tool that could utilize GIS data from publicly available spatial information for calculation. In order to speed up processing time, tool with incorporating Graphics Processing Unit (GPU) was also being explored. These tools enables users to conduct precise road traffic noise predictions using the RONOSS-HK methodology. This paper outlines the RONOSS-HK prediction methodology and the latest advancements in its associated assessment tools.

8:20

1aCA2. Transforming child speech data into clinical-grade artificial intelligence pipelines for speech-language impairment detection. Marisha Speights (Commun. Sci. and Disord., Northwestern Univ., Frances Searle Bldg., 2240 Campus Dr., Evanston, IL 60208, marisha.speights@northwestern.edu), Vishal Shrivastava (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Akangkshya Pathak, Hannah Ma, Bharath Yedla, and Peer Herholz (Northwestern Univ., Evanston, IL)

Diagnosing speech-language impairments in children using AI requires an end-to-end audio processing pipeline capable of handling heterogeneous datasets, speech variability, and clinical-grade accuracy. This study presents a framework that converts raw child speech into AI-ready datasets through rigorous standardization, quality assurance, and explainability. The process begins with automated file restructuring and metadata-tagging to seamlessly integrate audio, video, and transcripts. Advanced preprocessing techniques—such as spectral noise reduction, silence normalization, and adaptive segmentation—produce clean datasets while preserving critical linguistic and acoustic features. The Montreal Forced Aligner synchronizes speech and transcripts at the phonetic level, enabling detailed speaker diarization and annotation. At its core, the AI pipeline employs fine-tuned models like Whisper for ASR and neural network classifiers trained on high-dimensional acoustic and prosodic embeddings. A custom bias analysis framework ensures fairness across diverse demographics, while explainable AI-powered phonetic grading and speech analysis deliver actionable insights for clinicians. Automated orchestration via GitHub Actions minimizes manual effort, enhancing scalability and operational efficiency. By prioritizing clinical interpretability and data security, this framework sets a new benchmark for detecting speech-language impairments, advancing speech pathology and AI research.

8:40

1aCA3. Inclusive automatic speech recognition: A framework for equitable speech recognition in children with disorders. Vishal Shrivastava (Commun. Sci. and Disord., Northwestern Univ., 1024 Noyes St., Evanston, IL 60201, shrivastava_vishal@outlook.com), Marisha Speights (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), and Akangkshya Pathak (Northwestern Univ., Evanston, IL)

Automatic Speech Recognition (ASR) systems often fail to accommodate the diverse speech patterns of children with speech disorders, leading to inaccuracies that undermine their usability in critical diagnostic, therapeutic, and educational applications. This challenge arises due to biases in existing models, which are primarily trained on adult and non-disordered speech, limiting their generalization capabilities and fairness. To address these limitations, we propose a robust technological framework centered on domain-agnostic feature extraction and adversarial training. The feature extractor is designed to learn universal audio representations that transcend domain-specific biases, enabling accurate processing of diverse speech inputs. Adversarial debiasing serves as a key mechanism, optimizing the system to minimize label prediction errors while actively discouraging reliance on domain-dependent features. To further enhance performance, divergence-aware data augmentation generates enriched training datasets, ensuring the model effectively handles variations in speech patterns. Additionally, advanced strategies such as synaptic intelligence and experience replay ensure the retention of critical learned knowledge during iterative model updates. This innovative approach holds the potential to transform ASR systems into equitable and reliable tools, empowering educators, therapists, and caregivers to better support children with speech disorders while advancing the inclusivity of speech recognition technologies.

9:00–9:20 Break

9:20

1aCA4. Feature selection for machine-learned crowd reactions at collegiate basketball games. Jason D. Bickmore (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, jdb387@byu.edu), Mitchell C. Cutler, Katrina Pedersen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shannon Proksch (Psych., Augustana Univ., Sioux Falls, SD), Mark K. Transtrum, and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Crowds at basketball games react acoustically to events on the court in many ways, including applauding, chanting, cheering, and making distracting noises. Acoustic features can be extracted from recordings of crowd noise at Brigham Young University basketball games to train machine learning models for classifying crowd reactions. In this study, feature selection

using random forests and LASSO logistic regression was used to identify the most useful acoustic features for identifying and classifying crowd reactions. Features related to specific 1/3-octave band shapes, sound level, and tonality are found to be highly relevant. Including feature histories can increase classifier accuracies by up to 12%. Interestingly, some features are better predictors of future crowd reactions than current reactions. Reduced feature sets are human-interpretable on a case-by-case basis for the crowd reactions they predict.

9:40

1aCA5. Principal component analysis applied to feature selection at collegiate basketball and football games. Jason D. Bickmore (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, jdb387@byu.edu), Mitchell C. Cutler, Eli Farrer (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shannon Proksch (Psych., Augustana Univ., Sioux Falls, SD), Christian N. Anderson, Katrina Pedersen, Mark K. Transtrum, and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A supervised machine learning model trained acoustic features has been used to previously identify crowd noise reactions during collegiate basketball games. Three of these features come from principal component analysis on one-third octave band spectra [M. C. Cutler *et al*., J. Acoust. Soc. Am. **155**, 962–970 (2024)]. Principal component analysis was repeated on a new dataset of crowd noise from college football games. Despite differences in venue, crowd size, and reactions dependent on event type, the principal component vectors were similar. In both cases, the top three principal components relate to overall sound pressure level, distinguishing crowd noise from public address (PA) noise, and distinguishing crowd noise from music played over the PA system. The results suggest the viability of this approach in identifying crowd reactions in recordings with other noise sources present.

MONDAY MORNING, 19 MAY 2025 — BISSONET/CARONDELET, 11:10 A.M. TO 12:00 NOON

Session 1aID

Interdisciplinary: Plenary Lecture: The Past Has Ears at Notre-Dame: Acoustic Research at the Intersection of Virtual Reality, Cultural Heritage, and Experimental Archaeology

Michael Vorlaender, Chair
IHTA, RWTH Aachen University, Kopernikusstr. 5, Aachen 52074, Germany

Chair's Introduction—11:10

Invited Paper

11:15

1aID1. The Past Has Ears at Notre-Dame: Acoustic research at the intersection of virtual reality, cultural heritage, and experimental archaeology. Brian F. Katz (Sorbonne Université, CNRS, d'Alembert, Sorbonne Univ, CNRS, Paris 75012, France, brian.katz@sorbonne-universite.fr)

The aftermath of the devastating 2019 fire at Notre-Dame Cathedral has now culminated in its restoration and grand reopening to the public in December 2024. As part of the restoration effort, scientific working groups were formed, including one dedicated to Acoustics. In parallel, two projects on architectural acoustic heritage were launched: the European initiative "The Past Has Ears" and the French interdisciplinary study "The Past Has Ears at Notre-Dame." Through these concerted efforts, we elaborated a digital acoustic twin of the pre-fire cathedral, adapted to predict both future and past states, integrating architectural, decorative, and archival data. This simulation model was then employed to explore questions about conditions over time, addressing inquiries related to musicology, religious practice, and organology and examining how the acoustics of Notre-Dame varied through the centuries. Beyond passive renderings, experimental virtual archaeological-acoustics techniques placed trained musicians in historically relevant settings that emerged during the cathedral's construction, shedding light on the dynamic relationship between musical style, performance, and architecture. Further studies focused on Notre-Dame's historical organs, particularly the position of the early medieval organ and how placement influenced acoustics. Responding to widespread public interest, the research team also conducted various outreach activities, including a radio-fiction series, a spatial audio-guide, and an immersive virtual concert.

Session 1aMU

Musical Acoustics: General Topics in Musical Acoustics I

Mark Rau, Chair
Music and Theater Arts & Electrical Engineering and Computer Science, Massachusetts Institute of Technology, 77 Massachusetts Ave, Cambridge, MA 02139

Contributed Papers

8:20

1aMU1. Further investigation of the boundary conditions and directivity of glockenspiel bars. Hanna Pavill (Brigham Young Univ., N284 ESC, Provo, UT 84602, pavillh@byu.edu) and Micah Shepherd (Brigham Young Univ., Provo, UT)

The glockenspiel, a percussion instrument with pitched metal bars mounted in a keyboard-like arrangement, produces sound through mallet strikes. In the standard mounted configuration, the bars are supported by four parallel rails. Two rails hold the top row of bars, while the other two support the bottom row. One end of each bar is secured with a screw, while the other usually rests freely on the rail. The bars' mounting conditions and vibrational modes significantly influence their acoustic radiation and directivity. While free-free beam theory provides a foundational understanding of individual bar behavior, it does not fully capture the dynamics when bars are played within their mounted configuration. This work continues to investigate the directivity of glockenspiel bars in their standard setup using a directivity measurement system and a scanning laser Doppler vibrometer. This study explores how the bars' mounting conditions affect their acoustic output by analyzing the vibrational modes and radiated sounds. Comparisons between theoretical models and experimental data reveal how the structure and mounting affect sound radiation. These findings advance our understanding of the glockenspiel's unique acoustic characteristics, offering new insights into the interaction between vibrational modes and real-world playing conditions.

8:40

1aMU2. An investigation of the contribution of a non-linear harmonic to the sound of a Balinese gamelan gong. Dallin T. Harwood (Brigham Young Univ., N247 ESC, Brigham Young University, Provo, UT 84602, dallinharwood@gmail.com), Hanna Pavill, and Micah Shepherd (Brigham Young Univ., Provo, UT)

Balinese gamelan gongs are percussion instruments of special interest because of their unique geometry and sound. Unlike a Chinese tam-tam, the gongs are quite thick, with a protruding dome in the center and long edges that sharply wrap around the circumference of the gong. When struck in the center, the larger gongs are designed to produce a strong beating pattern. Previous studies have shown the cause of this beating phenomenon to be the proximity of the harmonic of the first axis-symmetric mode to the frequency of the second axis-symmetric mode [Krueger *et al*., J. Acoust. Soc. Am. 128(1) (2010)]. No work has yet been done to characterize this first harmonic in terms of its modal deflection shape or directivity pattern either isolated or coupled with the rest of the system when it produces the beating. This paper will present measurements and discussion of the vibrational and directional characteristics of the gong's first harmonic as well as the beating system as a whole.

9:00

1aMU3. Empirical analysis of modal precession in crystalline Himalayan singing bowls. Isaac R. Settle (Phys., Utah Valley Univ., 800 W University Pkwy, Orem, UT 84058, 10800818@uvu.edu), Kathryn Dispennette (Exercise Sci., Utah Valley Univ., Orem, UT), and Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT)

To date, there is little data on the modal precession and acoustic beating effect observed when utilizing crystalline structure Himalayan singing bowls. This study will present data found through multiple experiments, including the use of a multipoint scanning laser Doppler vibrometer, performed on several crystalline bowls. The results from various experiments support the hypothesis that the acoustic beating effect often perceived from Himalayan bowls of any structural type is due to the rotation of the fundamental mode around the circumference of the bowl, and not an interference effect cause by the modulation of multiple frequencies present. In addition, the highly symetric geometry of the crystalline bowls produces very small spread in the bandwidth of frequencies present, including a lack of harmonic content beyond the fundamental frequency, when the crystalline bowl is excited.

9:20–9:40 Break

9:40

1aMU4. An efficient detuning detection method for the tabla. Sreerag Ashok (Design, Indian Inst. of Technol., Kanpur, Kanpur, Uttar Pradesh, India), Akshay Kumar (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, Uttar Pradesh, India), and Nachiketa Tiwari (Design, Indian Inst. of Technol., Kanpur, Indian Inst. of Technol. Kanpur, Kanpur, India, ntiwari@iitk.ac.in)

Tabla, the musical drum, can produce a large number of musical syllables or *bols*. To produce a good musical performance, the tabla membrane has to be symmetrically tuned to the right pitch. Tabla players sense the extent of such symmetry in the membrane tension by playing the /t̪a:/ *bols* at eight different locations along the circumference of the tabla. This is a tedious process. Our findings show that a single strike of a /ʈ/ *bol* can be a good indicator of the presence of asymmetry in membrane tension. Using two stereoscopically arranged high-speed cameras, and an audio data acquisition system, we have investigated /t̪a:/, /ʈ/, and /t̪ʊn/ *bols* in the context of tuning a tabla. When the membrane was asymmetrically tuned, the frequency spectrum of /ʈ/ *bol* showed two close peaks corresponding to the second harmonic, and its out-of-plane deflection pattern revealed rotation in mode ψ_{11} that features one nodal diameter, and one nodal circle corresponding to the rim of the tabla. We noted that such frequency splitting was present even in tablas which had slight asymmetry in the membrane tension. We propose sensing such a frequency-splitting phenomenon to implement an efficient detuning detector for the tabla.

10:00

1aMU5. Exploring the musical pillars of Nellaiyappar Temple in Tamil Nadu. Sedwin T C (Dept. of Mech. Eng., Indian Inst. of Technol. Madras, IIT Madras, Chennai, Tamil Nadu 600036, India, me23m102@smail.iitm.ac.in), Arthis P, Lenin B. M C (School of Mech. Eng., Vellore Inst. of Technol. Chennai, Chennai, India), and Chandramouli Padmanabhan (Dept. of Mech. Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu, India)

The musical pillars in the Arulmigu Nellaiyappar Temple in Tirunelveli, Tamil Nadu, consist of many clusters, varying from 4 to 48 pillars each. These pillars were built in the 7th century and are made of a specific kind of granite rock. By tapping one of the pillars by hand, musical notes are produced. Interestingly, this tapping causes the vibrations to be transmitted to the nearby sub-pillars. To comprehend on how the musical notes are produced and how vibrations are transferred to the clusters, an extensive experimental investigation was carried out. By positioning the accelerometers at a few pillars and tapping at the same pillar or other sub-pillars, the vibrations were measured. Along with the vibrations, a microphone was used to record the sound produced. A metrological laser type 3-D scanner was used to generate 3-D CAD models of the pillars. The free vibration characteristics of the pillars are subsequently determined using finite element methods, and the results are then compared with data that has been measured. A transient analysis has been carried out to understand how the wave is propagated from the stimulated pillar to the neighbouring pillars. Comparisons of simulation results with experimental measurements are presented.

10:20

1aMU6. Immersive recordings in virtual acoustics: Differences and similarities between a concert hall and its virtual counterpart. Gianluca Grazioli (McGill Univ., 555 Sherbrooke St W, Montreal, QC H3A 1E3, Canada, gianluca.grazioli@mail.mcgill.ca), Andrea Gozzi (Université de Sherbrooke, Montreal, QC, Canada), Mehdi Rahimdokht (McGill Univ., Montreal, QC, Canada), Alessandro Braga (École de technologie supérieure - ÉTS Montréal, Montréal, QC, Canada), Richard King, and Wieslaw Woszczyk (Music Res., McGill Univ., Montreal, QC, Canada)

Virtual acoustic systems can artificially alter a recording studio's reverberation in real-time using spatial room impulse responses captured in different spaces. By recreating another space's acoustic perception, these systems influence various aspects of a musician's performance. Traditional methods involve recording a dry performance and adding reverb in post-production, which may not align with the musician's artistic intent. In contrast, virtual acoustic systems allow simultaneous recording of both artificial reverb and the musician's interaction using standard recording techniques—just as it would occur in the actual space. This study analyzes immersive recordings of nearly identical musical performances captured in both a real concert hall and McGill University's Immersive Lab, which features a new dedicated virtual acoustics software, and highlights the similarities and differences between the real space and its virtual counterpart.

MONDAY MORNING, 19 MAY 2025 — BALCONY I, 8:00 A.M. TO 11:00 A.M.

Session 1aPAa

Physical Acoustics and Biomedical Acoustics: Acoustic Holograms and Wavefront Modulation Techniques

Kai Melde, Cochair
Heidelberg University, Im Neuenheimer Feld 225, IMSEAM, Stuttgart 69120, Germany

Noé Jiménez, Cochair
I3M, Universitat Politecnica de Valencia, Camino de Vera s/n, Valencia 46022, Spain

Invited Papers

8:00

1aPAa1. Multifrequency and multifocal acoustic holograms in complex media: Applications to transcranial ultrasound focusing. Jean-Francois Aubry (CNRS, Phys. for Medicine Paris, 10 rue d'Oradour-sur-Glane, Paris 75015, France, jean-francois.aubry@espci.fr)

Advancements in ultrasonic beam shaping using multi-element arrays have significantly improved the ability to focus ultrasound through aberrating media, such as the human skull, enabling rapid growth in transcranial-focused ultrasound therapies. However, the high cost of these technologies presents a barrier to widespread adoption. Acoustic lenses offer a promising, low-cost alternative for correcting skull-induced aberrations. We will show that computed tomography-based simulations allow to design silicone acoustic lenses for single-element transducers. This approach demonstrates a 10-fold increase in energy deposition at the target when lens-based aberration correction is applied. Additionally, we will show that the same acoustic lens can produce multifrequency holograms across a 500 kHz to 1 MHz range, validated both in free-field conditions and through human skulls, provided the phase is unwrapped before the estimation of the thickness of the lens. Finally, we will demonstrate that the unwrapped acoustic lenses allow multifocal refocusing and we will discuss potential applications in transcranial ultrasonic neurostimulation for the treatment of drug-resistant depression. Together, these results emphasize the necessity of aberration correction to mitigate the field distortions induced by the human skull and highlight the potential of tailored acoustic lenses as a cost-effective solution for advancing transcranial ultrasound therapies.

8:20

1aPAa2. Hologram-assisted focused ultrasound for brain therapy. Sergio Jiménez-Gambín (Biomedical Eng., Columbia Univ., 630 W. 168th St P&S 19-418, New York, NY 10032, sj3044@columbia.edu), Noé Jiménez, Francisco Camarena (I3M, Universitat Politecnica de Valencia, València, Spain), and Elisa Konofagou (Biomedical Eng.; Radiology, Columbia Univ., New York, NY)

Acoustic holograms are 3-D-printed structures that can passively modulate the transmitted wavefront from a single-element source and generate arbitrary-shape beams in multi-layered media. This approach has been of great interest in the brain therapy field, where the skull-induced beam aberration leads to undesired off-targeting effects while the complex brain anatomy may limit optimal target coverage. Acoustic holograms have been shown to successfully address these limitations by providing a more simple and cost-efficient alternative as compared to phased arrays. This presentation will cover assessment of acoustic holograms in both animal and human applications, including hologram design technical details. First, an overview of potentially useful acoustic patterns generated transcranially for brain applications will be presented. Second, the basics of hologram design are described, with different approaches of wavefront processing and heights distribution generation depending on single-element transducer geometry. Third, an overview of the preclinical blood–brain barrier (BBB) opening and neuromodulation outcomes in mice will be presented, followed by BBB opening outcomes in non-human primates. Finally, in-silico feasibility of clinical BBB opening and neuromodulation will be reported. Hologram-assisted focused ultrasound (FUS) is a promising and powerful technology which provides new avenues into a novel and simple approach for cost-efficient and rapid FUS application in the brain.

8:40

1aPAa3. A library of acoustic holograms for precise targeting of murine brain regions. Rachel Burstow (Dept. of Surgical & Interventional Eng., King's College London, Becket House, 1 Lambeth Palace Rd., London SE1 7EU, United Kingdom, rachel.burstow@kcl.ac.uk) and Antonios Pouliopoulos (Dept. of Surgical & Interventional Eng., King's College London, New York, NY)

This work aims to produce a library of acoustic holograms, which will precisely target brain regions of varying shapes, sizes, and focal depths through the aberrating murine skull. K-wave was used to simulate an isotropic grid for a focused single-element transducer (H-204, 1.68 MHz, 82-mm aperture; Sonic Concepts) using 6 points per wavelength. A time reversal method was used to design a phase-only holographic lens in clear resin. Binary matrices of a single mouse skull and brain were used as background media to account for tissue inhomogeneity and varying speed of sound. The Allen Brain Atlas was used to identify and select the regions for targeting. Four different brain regions were selected: the substantia nigra, the subthalamic nucleus (both relevant to Parkinson's disease), the central lateral thalamic nucleus (chronic pain), and the hippocampus (Alzheimer's disease). With the top of the mouse head positioned at a distance of 60 mm, these regions required a focal depth of 62.4, 62.1, 60.6, and 60.2 mm, respectively. The successful in-target focusing was 50%, 87.5%, 84%, and 38% of the ROI, respectively. This work demonstrates that acoustic holograms can improve targeting accuracy and reduce out-of-target effects, thereby improving the efficiency of drug delivery to the brain.

Contributed Paper

9:00

1aPAa4. Acoustic holograms for both therapy and monitoring transcranial ultrasound. Noé Jiménez (Instituto de Instrumentación para Imagen Molecular (I3M), Universitat Politècnica de València - Consejo Superior de Investigaciones Científicas, Camino de Vera s/n, Valencia 46022, Spain, nojigon@upv.es), Nathalie Lamothe, Diana Andrés, Alicia Carrión (Instituto de Instrumentación para Imagen Molecular (I3M), Universitat Politècnica de València - Consejo Superior de Investigaciones Científicas, Valencia, Spain), José A. Pineda-Pardo (HM CINAC (Centro Integral de Neurociencias Abarca Campal), Hospital Universitario HM Puerta del Sur, HM Hospitales, Madrid, Spain), Alba Eroles-Simó, Víctor Vegas-Luque, María E. Pérez-Sirvent, Juan J. Rodríguez-García, José L. Alonso-Ramos, and Francisco Camarena (Instituto de Instrumentación para Imagen Molecular (I3M), Universitat Politècnica de València - Consejo Superior de Investigaciones Científicas, València, Spain)

When using holograms for cavitation-based therapeutic ultrasound, the acoustic focal spot may be misaligned with the passive cavitation detector (PCD) axis. We propose the use of 3-D-printed acoustic lenses for both therapeutic and monitoring systems. First, an acoustic hologram is designed to sharply focus the therapeutic beam at an arbitrary target into the skull. Then, a second hologram is designed for passive cavitation beamforming, i.e., to beamform the reception wavefront of the PCD with the target location, and in this way, cavitation emissions can be locally detected. Experiments with an *ex vivo* macaque skull and several blood vessel phantoms with microbubbles were conducted using a 500-kHz focused transducer and a confocal 3.5-MHz piezoelectric PCD. Results show that passive cavitation beamforming improves sensitivity and aligns the response of the PCD with a therapeutic focus located off-axis. The retrieved cavitation doses using this approach show a similar dynamic than for on-axis targeting setup. When compared to off-axis targeting, passive cavitation beamforming using holograms increases sensitivity to cavitation emissions, enabling precise, localized monitoring for therapeutic ultrasound. This method offers a cost-effective way to enhance the performance and monitoring of transcranial therapeutic ultrasound systems.

9:20–9:40 Break

Invited Papers

9:40

1aPAa5. From digital twins to singularities: Unlocking acoustic holograms with automatic differentiation. Tatsuki Fushimi (Inst. of Library, Information and Media Sci., Univ. of Tsukuba, 1-2 Kasuga, Tsukuba 3050821, Japan, tfushimi@slis.tsukuba.ac.jp)

The ability to precisely shape acoustic fields is important for diverse applications such as particle manipulation, material assembly, and ultrasonic haptics. Automatic differentiation (AD) offers a flexible and efficient framework for optimizing acoustic holograms, enabling precise control over acoustic fields. By integrating AD with experimental workflows via digital twin methodologies, it provides a

powerful tool to bridge simulation and real-world implementation, allowing for iterative refinement with a focus on efficiency and accuracy. This talk begins by explaining the principles of AD-driven optimization, illustrating its capability through examples such as phase singularity control and hologram optimization under complex constraints. These examples demonstrate how AD can efficiently produce solutions tailored to specific requirements, serving as an effective tool for practical implementation. Beyond optimization, this talk highlights the importance of understanding what acoustic holograms represent and the broader principles they embody. AD provides a pathway to explore and manipulate specific solutions, but true insight requires analyzing the fundamental properties and constraints of holographic fields to better understand the mechanisms and limits that define acoustic holograms.

10:00

1aPAa6. Making holograms interactive. Athanasios Athanassiadis (Heidelberg Univ. & Max Planck Inst. for Medical Res., Im Neuenheimer Feld 225, IMSEAM - AG Fischer, Heidelberg 69120, Germany, thanasi@uni-heidelberg.de), Lennart Schlieder (Max Planck Inst. for Intelligent Systems, Tübingen, Germany), Rahul Goyal, and Peer Fischer (Heidelberg Univ. & Max Planck Inst. for Medical Res., Heidelberg, Germany)

Acoustic holograms have developed over the last 10 years as powerful tools for ultrasonic wavefront and field shaping. They can provide high-resolution control over wavefronts, allowing them to form much more spatially complex wavefronts than existing array transducers, and with significantly simpler hardware. However, unlike ultrasonic arrays, acoustic holograms are static: the fields they produce cannot be readily changed or adapted in real-time. To address this limitation, our group has developed different approaches to realize dynamic and interactive holograms over the past several years. In this talk, I will describe these recent efforts and highlight some of our latest work in this area, including the combination of holograms and transducer arrays, the integration of machine learning techniques in the hologram design process, and the development of remotely addressable electrical circuits for the dynamic updating of high-density ultrasonic arrays. Our work demonstrates different approaches that can be taken to expand the versatility of holograms. This talk highlights the opportunities for interactive holograms and the advantages of certain methods.

10:20

1aPAa7. Acoustic vortex beams interacting with a metasurface and the acoustic Hall effect. Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38655, zhang@olemiss.edu), Xinyue Gong (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS), and Joao Ealo (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, Cali, Valle del Cauca, Colombia)

Acoustic metasurfaces are capable of modulating the wavefronts of sound beams, governed by generalized laws of reflection and refraction. However, these laws fall short in describing the angular momenta carried by acoustic vortex beams. Such beams, characterized by twisted wavefronts arising from a rotating phase in their cross sections, exhibit unique phenomena with promising applications. Our numerical simulations reveal that an acoustic vortex beam refracted by a metasurface exhibits an asymmetry in its pressure field relative to the incident plane [Fan and Zhang, Phys. Rev. Res. **3**(1), 013251 (2021)]. This asymmetry is analogous to that seen in the refraction of vortex beams propagating through stratified media [Fan *et al*., Phys. Rev. Res. **1**(3), 032014 (2019)]. The asymmetry depends on the twist direction of the beam's rotating phase, a phenomenon referred to as the acoustic Hall effect. The asymmetry can also be observed via a geometric projection of the vortex beam, which interprets angular momentum carried by the beam when obliquely reflected from a flat interface [Zou *et al*., Phys. Rev. Lett. **125**(7), 074301 (2020)]. Here, we experimentally observed the asymmetry of a vortex beam interacting with a metasurface, along with the associated conservation of angular momentum.

Contributed Paper

10:40

1aPAa8. Design and validation of custom acoustic holograms for ultrasonic immersion testing of anisotropic silicon. Lauren Katch (Eng. Sci. and Mech., Penn State Univ., Philadelphia, PA) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., 212 Earth-Engr Sci. Bldg, University Park, PA 16802, aza821@psu.edu)

Acoustic holograms offer powerful capabilities for manipulating ultrasonic fields, yet their application to immersed anisotropic materials remains challenging due to complex wave propagation behavior. This study advances acoustic holography by developing customized lenses that enhance ultrasonic beam focusing in anisotropic silicon wafers immersed in water. Using ray tracing combined with gradient descent optimization, novel lens geometries were designed to constrain wave propagation to a conical profile within [3 1 1] and [1 3 5] silicon orientations. The lenses were fabricated via stereolithographic 3-D printing and systematically compared against conventional spherical lenses through analytical modeling and experimental testing. Angular spectrum approach modeling demonstrated that the custom lenses produced improved focusing, with up to 41% higher amplitude and 49.8% better circularity compared to spherical lenses. Experimental validation using flat-bottom hole defects showed that the custom lenses achieved more refined defect signatures while requiring 10 dB less electrical gain than spherical lenses. The benefits were most pronounced in materials with lower crystal symmetry, where directionally varying focal patterns could be made more uniform through custom lens design. This work establishes a comprehensive framework for optimizing acoustic holograms that account for fluid–solid interfaces and anisotropic wave propagation, advancing capabilities for ultrasonic inspection of complex materials.

Session 1aPAb

Physical Acoustics, Computational Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Viscothermal Effects in Phononic Crystals and Acoustic Metamaterials

José Sánchez-Dehesa, Cochair
Electronic Engineering, Universitat Politecnica de Valencia, Camino de Vera s/n, Valencia ES-46022, Spain

Vicente C. Henriquez, Cochair
Technical University Lyngby, Lyngby, Denmark

S. Hales Swift, Cochair
Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082

Chair's Introduction—8:15

Invited Papers

8:20

1aPAb1. Nonreciprocal transmission through acoustic Tesla valve filled with viscous fluid. Arkadii Krokhin (Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, arkady@unt.edu)

Attenuation of sound is a viscous fluid is a signature of irreducible process accompanied by increase of entropy. Presence of elastic scatterers adds non-dissipative contribution to attenuation. If the scatterers are symmetrically distributed between emitter and receiver (*P* symmetrical system), sound transmission emitter ↔ receiver remains reciprocal, although it is irreducible in each direction. From the pioneering work by Rayleigh, it is known that acoustic pressure is a reciprocal quantity even in a viscous fluid and non-symmetric distribution of scatterers. This property makes sound transmission an exclusion from the general mechanical principle requiring nonreciprocal dynamics if time-reversal symmetry (*T* symmetry) is broken. Here we demonstrate that acoustic intensity is a truly nonreciprocal quantity if *PT* symmetry is broken. As an example of an acoustic system with strongly broken *P* symmetry, we consider a Tesla valve filled with liquids of different viscosity, thus presenting mechanical systems with different levels of broken *T* symmetry. Experimental spectra for intensity confirm our theoretical conclusion about nonreciprocal wave dynamics in the systems with broken *PT* symmetry. Considering viscosity coefficient η as a nonreciprocal parameter, we show that replacement $\eta \rightarrow -\eta$ does not restore reciprocity, i.e., transmission through a viscous fluid does not possess Onsager symmetry. [Work supported by the NSF Grant No. 1741677 and by the AFOSR Grant FA9550-23-1-0630.]

8:40

1aPAb2. New directions for metamaterial optimization including viscous and thermal acoustic losses. Peter R. Andersen (Res. & Exploration, GN, Lautrupbjerg 7, Ballerup 2750, Denmark, prandersen@jabra.com)

Structural optimization methods, such as shape and topology optimization, hold significant promise for enhancing the design of acoustic metamaterials. Prior research emphasizes the necessity of incorporating viscous and thermal losses into the optimization process for resonator-based acoustic metamaterials. While acoustic losses can be integrated into boundary element and finite element frameworks, shape optimization often lacks the design flexibility required for complex metamaterial structures. Consequently, topology optimization is frequently preferred due to its greater design freedom. Resent research has demonstrated that density-based topology optimization can include acoustic losses using an indicator function combined with the boundary layer impedance condition. As an alternative to density-based topology optimization, this presentation will discuss implementation aspects of a cut-elements and level-set based topology optimization approach including viscous and thermal losses. The cut-elements method offers advantages over traditional density-based approaches by providing a clear interface between acoustic and rigid domains, ensuring that each design iteration accurately reflects the underlying physical behavior.

Contributed Papers

9:00

1aPAb3. Characterizing attenuation in 3-D-printed ultrasonic Metamaterials. Alireza Tadibi (Mech. and Mater. Eng., Univ. of Cincinnati, 230 Ludlow Ave. Apt 4, Cincinnati, OH 45220, tadibiaa@mail.uc.edu), Yehia Zakaria, and Ahmed Allam (Mech. and Mater. Eng., Univ. of Cincinnati, Cincinnati, OH)

3-D-printed Metamaterials offer new capabilities for manipulating ultrasonic waves in the MHz regime, promising new devices, such as lenses and phase shifters, in imaging and power applications. The losses present in such devices can be critical to their practical realization, yet the factors contributing to attenuation in metamaterials remain underexplored. This study presents a numerical and experimental analysis to link the total attenuation expressed by 3-D-printed metamaterials to their unit cell design and the 3-D printing process used in their fabrication. First, we develop a lossy unit-cell analysis based on the finite element method to predict the effective attenuation of the metamaterial. We analyze factors such as unit cell size and geometry and viscoelastic losses in the base material to determine their influence on the metamaterial's frequency-dependent attenuation. Then, we experimentally characterize the attenuation of multiple metamaterial realizations using the through-transmission method within a frequency range of 0.5–5 MHz. Comparing the numerical predictions to experimental results reveals the influence of practical 3-D-printing process parameters on attenuation. The results establish a framework for evaluating and optimizing attenuation behavior in ultrasonic metamaterials, accelerating their adoption in practical ultrasonic applications, including imaging and therapeutic ultrasound, and non-destructive testing.

9:20

1aPAb4. Designing phononic crystals to improve viscous absorption at low frequencies. José Sánchez-Dehesa (Electron. Eng., Universitat Politecnica de Valencia, Universitat Politécnica de Valéncia, Camino de Vera s/n, Valencia, Valencia ES-46022, Spain, jsdehesa@upv.es), Martin Ibarias (Electron. Eng., Universitat Politecnica de Valencia, Valencia, Valencia, Spain), Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX), Vicente C. Henriquez, and Frieder Lucklum (Tech. Univ. of Denmark, Lyngby, Denmark)

This work reviews configurations of phononic crystals specially designed to enhance viscous absorption at low frequencies. The phononic crystals are made of two-dimensional distributions of cylindrical rods with circular sections embedded in air. A homogenization theory has been developed to easily study the case of finite structures of large dimensions. The theory provides analytical expressions for the effective size of the structure, effective mass density, effective sound speed, and effective viscosity. Results are reported for semi-infinite slabs, finite clusters, and superstructures denominated as super-crystals consisting of phononic crystals with two different lattice periods. It is shown that the filling fraction is a key factor in determining the maximum absorption of crystals made with cylinders having the same diameter. For super-crystals, the combination of two lattice periods produces a decay coefficient faster than that of the crystal made with smaller cylinders. Experimental data support the results predicted by the theory.

9:40–10:00 Break

10:00

1aPAb5. Acoustic performance prediction of fibrous materials using anisotropic and fundamental parameter distributions. Do Yong Kim (Mech. Eng., Seoul National Univ., 16, Nakseongdaeyeok 10-gil, Gwanak-gu, Seoul, Republic of Korea, Seoul 08799, Korea, younggarisnu@snu.ac.kr) and Yeon June Kang (Mech. Eng., Seoul National Univ., Seoul, Korea)

Eco-friendly and recycled fibrous acoustic materials are emerging as key solutions for sound package interiors. Optimizing these materials for target acoustic performance requires predictive models that account for microstructural parameters. However, existing approaches often neglect inhomogeneity and anisotropy, relying on homogeneous assumptions and inconsistent parameter selections. This study introduces a deep learning model that leverages anisotropic parameter distributions to predict acoustic performance with significantly reduced computational cost. PET felt cross sections were analyzed via micro-computed tomography (micro-CT) to extract anisotropic parameter distributions, which were used to train the model. The proposed framework demonstrates superior predictive performance compared to conventional homogeneous models. Its efficiency in handling large datasets and reducing computation time makes it highly effective for optimizing and designing fibrous acoustic materials. By identifying critical structural parameters influencing acoustic behavior, this model enables data-driven optimization and tailored design. Furthermore, this research lays the foundation for practical applications, extending to diverse fibrous materials and advancing toward generating structural design images that enable multi-objective optimization. By incorporating acoustic performance and constraints like cost and sustainability, these models open possibilities for real-world applications, bridging design, and functionality.

Invited Paper

10:20

1aPAb6. Acoustic eigenvalue analysis of microstructures using an equivalent source method. Yong-Bin Zhang (Hefei Univ. of Technol., No. 193 Tunxi Rd., Hefei, Anhui 230009, China, ybzhang@hfut.edu.cn), Meng-Hui Liang, and Chang-Jun Zheng (Hefei Univ. of Technol., Hefei, China)

Acoustic eigenvalues provide crucial information for the design and optimization of microstructures, including metamaterials and phononic crystals. Computational methods based on the wave equation have been widely used to calculate acoustic eigenvalues; however, these methods typically neglect thermoviscous effects. In microstructures, thermoviscous effects significantly influence acoustic eigenvalues because their dimensions approach the boundary layer thickness. This phenomenon has been extensively observed in studies of acoustic black holes (ABH) and acoustic metamaterials. Therefore, an eigenvalue solver for thermoviscous acoustic problems was developed using the Equivalent Source Method (ESM). By adopting the ESM concept, the solutions of the thermoviscous equations are coupled on the structure surface through isothermal and non-slip conditions. However, the frequency dependence of the transfer matrix in the ESM equation leads to a nonlinear eigenvalue problem (NLEP), introducing additional challenges to eigenvalue analysis. To address this issue, the contour integral method was employed to convert the NLEP into a generalized eigenvalue problem (GEVP). This method effectively identifies complex acoustic eigenvalues commonly encountered in thermoviscous acoustic problems. The effectiveness and accuracy of the proposed method were verified through simulations of an ABH and a simplified MEMS microphone.

Session 1aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session I

Nirmal Kumar Srinivasan, Chair
Audiology, Speech-Language Pathology, and Deaf Studies, Towson University, 8000 York Road, Towson, MD 21252

All posters will be on display from 8:00 a.m. to 11:00 a.m. Authors of odd numbered papers will be at their posters from 8:00 a.m. to 10:30 a.m. and authors of even numbered papers will be at their posters from 10:30 a.m. to 11:00 a.m.

Contributed Papers

1aPP1. Comparing the encoding of vowels via intensity and amplitude-modulation cues. Braden N. Maxwell (Psych., Univ. of Minnesota, 601 Elmwood Ave., Rochester, NY 14642, maxwe318@umn.edu), Brita M. O'Brien, and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Vowel formants may be encoded peripherally via local increases in firing rate (rate-place code) or decreases in neural fluctuations for fibers tuned near formants, relative to fibers tuned between formants. To distinguish between these alternatives, we compared the effects of intensity and amplitude-modulation cues on listeners' ability to identify synthetic vowels. Harmonic tone complexes were generated with a slope of 6 dB/octave, and formants were defined by either (1) increased intensity at formant frequencies or (2) changes in phase relationships between components to manipulate modulation depth at formant frequencies. Vowels were either (1) presented with intensity cues, but with modulation cues in opposition (providing more, rather than less, modulation depth at the formant frequencies), or (2) defined solely by reductions in amplitude modulation at the formant frequencies. Two control conditions used uniform cosine or random phase with standard intensity increases for formants. Preliminary results suggest that listeners can perceive vowels without intensity increments, based only on local changes in modulation depth, that phase changes counteracting intensity cues worsen performance, and that performance with random-phase stimuli is poorer than with cosine-phase stimuli, consistent with an important role for amplitude-modulation cues in the identification of synthetic vowels. [Supported by NIH, Grant R01DC012262.]

1aPP2. Effects of musical experience and sentence difficulty on noisy speech recognition in second language learners. Siheng Li (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, Zhongshan Dadao W. No. 55, Guangzhou 510000, China, eddysihengli@gmail.com), Shuhang Chen, and Yu Li (Appl. Psych. Programme, Dept. of Life Sci., BNU-HKBU United Int. College, Zhuhai, Guangdong, China)

Correctly recognizing second-language speech in a noisy environment is a difficult task for second-language learners. Early works have indicated the beneficial effects of musical training in native language acquisition and development. However, how musical training experiences influence second-language speech-in-noise (SIN) recognition in second-language learners remains unknown. To examine this question, we recruited 78 right-handed young adults who spoke English as their second language and showed different musical training experiences over the past 10 years and employed an English SIN test in which two factors were manipulated, signal-to-noise ratio level (SNR; quiet, + 5 dB, −5 dB) and sentence difficulty (easy, hard). The percentage of words correctly identified from sentences was used in the data analyses. The results revealed a significant interaction between musical training experience, SNR level, and sentence difficulty. Further analyses found that the long musical training group outperformed the short musical training group in identifying hard sentences from noise. Overall, these results provide new evidence for the benefits of music training experiences for SIN recognition in the context of second language learning. [Work supported by the Humanities and Social Sciences Foundation of Ministry of Education of China 20YJCZH079.]

1aPP3. Using nicotine to alleviate tinnitus and improve cognitive and speech performance. Xianhui Wang (Ctr. for Hearing Res. and Dept. of Otolaryngol. – Head and Neck Surgery, Univ. of California, Ohio University, Grover Ctr., W218, Athens, OH 45701, xw659217@ohio.edu), Jaden Pon, Camille Handa, Ayaan Mustafa (Ctr. for Hearing Res. and Dept. of Otolaryngol. – Head and Neck Surgery, Univ. of California, Irvine, CA), Jonathan Ge (Warren Alpert Med. School, Brown Univ., Providence, RI), and Fan-Gang Zeng (Ctr. for Hearing Res. and Dept. of Otolaryngol. – Head and Neck Surgery, Univ. of California, Irvine, CA)

Tinnitus is a phantom perception of sounds without external stimuli, affecting 10%–20% of the general population. Tinnitus is co-morbid with hearing loss, which not only impairs cognitive and speech performance but also causes mental health issues like anxiety and depression. Currently, no cure exists for tinnitus. Previous studies suggested involvement of nicotine in tinnitus generation and development. In a placebo-controlled, double-blinded design, this prospective study investigated the effect of nicotine, delivered via gum at varying doses, on tinnitus, cognition and speech perception in adults with chronic tinnitus. Preliminary results showed that nicotine (4 mg) not only improved selective attention and speech perception in noise but importantly reduced tinnitus loudness. especially in those with severe tinnitus. These findings suggest that nicotine can be potentially used as a therapeutic option for alleviating tinnitus and improving cognitive and speech performance.

1aPP4. The importance and better understanding of psychoacoustic parameters for measuring sound quality. Roland Sottek (Tech. Div., HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, Roland.Sottek@head-acoustics.com), Wade R. Bray (HEAD Acoust., Inc., Brighton, MI), and Andre Fiebig (Eng. Acoust., TU Berlin, Berlin, Germany)

In today's market, the influence of sound quality is becoming more and more important in the face of increasing competition and new acoustic standards. This leads to the need for accurate quality predictions that can be made as easily and quickly as possible. In most cases, the use of psychoacoustic parameters is the first choice for these sound quality metrics. However, the use of psychoacoustic parameters in legislation has been slow to spread. They often contain only guidelines for maximum levels or, at best, level penalties for very tonal noises, for example. To promote the meaningful use of psychoacoustic parameters, their application needs to be more accessible through free tools and easier to understand. For this purpose, the "psychoacoustic thermometer" was introduced, which vividly illustrates the

meaning of the standardized psychoacoustic parameters based on the Sottek Hearing Model. This paper describes the current developments in the calculation and scaling of these psychoacoustic quantities based on the Sottek Hearing Model and the need to systematically determine their just-noticeable differences. This leads to a better estimation of whether sounds differ in their perception, e.g., an important indication in acoustic quality control.

1aPP5. Federated study on auditory characteristics related to selective listening. Shunsuke Kidani (Japan Adv. Inst. of Sci. and Technol., 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, kidani@jaist.ac.jp), Masaharu Kato, and Yoko Shimada (Doshisha Univerisity, Kyotanabe, Japan)

Selective listening is an essential characteristic of smooth speech communication. Many papers report that temporal and frequency resolution of the auditory system and cognitive abilities play important roles in selective listening. However, it is unclear which abilities contribute to selective listening and to what extent. Previous studies have examined individual auditory characteristics, and no studies have examined them federated. This study aims to clarify the contribution of each ability to selective listening. We measured various auditory characteristics in the same participants to clarify the relationship between selective listening and each characteristic. Word intelligibility in noise and word intelligibility in dichotic listening were used as indices of selective listening. The measured auditory characteristics were a masking threshold, gap detection thresholds, amplitude modulation detection thresholds, temporal modulation transfer function, and N-back memory tasks. The results showed that none of the auditory characteristics alone could explain selective listening. This paper also reports on the results of the path analysis conducted to show the contribution pathways.

1aPP6. Influences of noise type and gamification on speech-in-noise performance for those with mild traumatic brain injury. Lauren Charney (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR, charneyl@ohsu.edu), Esteban Sebastian Lelo de Larrea-Mancera (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Boston, MA), Karen Garcia, Conner Corbett (O), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Difficulty hearing speech in noise is a common complaint among patients with mild traumatic brain injury (mTBI). The cause is not well understood, and effective testing and treatment of this complaint is lacking. The goal of this pilot project was to determine whether noise type or gamification affects speech in noise performance for participants with mTBI history. Participants with and without a history of mTBI completed a gamified version of a speech in noise task. First, participants completed a version with "garbled" speech maskers in which the speech spectra and modulations were preserved but the correlations across spectral bands were perturbed. Next, participants completed a version with traditional speech maskers. A subset of the participants also completed a different gamified version of the task with speech maskers only. While the groups did not perform significantly differently, there was an effect of noise type and gamified test version on performance in the speech-in-noise task. These results indicate the potential utility of gamification and "garbled" speech maskers in assessing speech-in-noise ability. Future directions include collecting a larger sample and comparing with self-report measures to explore whether game or noise type performance are more strongly correlated with reported mTBI symptoms.

1aPP7. Effect of musicianship on degraded speech perception during adolescence. Audrey L. Williams (Kent State Univ., 123 Cherry St., Apt. 1A, Kent, OH 44240, awill400@kent.edu) and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., Kent, OH)

Musical experience is associated with speech-in-noise perception among adults, potentially due to better stream segregation, frequency encoding, and temporal encoding. The effect of musicianship on degraded speech perception is less established. To investigate the effects of musical training on speech perception performance in adolescents and young adults, we tested 10- to 23-year-olds on two noise vocoded (NV) speech perception conditions (six-band NV speech followed by silence or babble) and on the QuickSIN. A linear mixed effect model was run on noise vocoded speech perception scores with age, musical experience (in years), NV condition order, and NV condition, and all interactions as fixed effects and participant number as a random effect to account for individual variability. Because all variables interacted and contributed to performance, post hoc regression models were run. As expected, people with more musical experience showed better scores on both NV speech conditions and the QuickSIN. Word report scores on all conditions improved with development. For the hardest NV condition (babble), scores were better when participants heard the easier (silence) condition first, likely due to perceptual learning. Thus, musical experience, age, and learning all contribute to performance on degraded speech perception tasks during adolescence and young adulthood. [Work supported by NIDCD.]

1aPP8. Development of an interactive educational platform for understanding interaural time difference in audiometric disorders. Srinidhi Narayanan (Biomedical Eng., Carnegie Mellon Univ., Pittsburgh, PA, srinidhn@andrew.cmu.edu)

This paper introduces an educational platform designed to enhance the understanding of interaural time difference (ITD) and its role in audiometric disorders. The platform leverages Unity3D for development and is deployed as a WebGL application to ensure accessibility across devices. By integrating interactive learning modules, gamified tasks, and a virtual guide, it offers an engaging and user-friendly experience. The learning modules simplify ITD concepts using animations and visual demonstrations, while gamified tasks challenge users to identify sound directions based on ITD cues. A virtual guide narrates key concepts and provides feedback, creating a dynamic and interactive environment. The platform's design prioritizes accessibility, ensuring usability for diverse audiences, including students, clinicians, and patients. The paper outlines the platform's development process, including modular design, scripting for audio spatialization, and iterative testing for usability. Although formal evaluation is pending, the platform demonstrates the potential of combining gamification and interactive design to make complex auditory concepts more approachable. Future plans include expanding content to related auditory phenomena, incorporating virtual reality, and conducting structured user studies. This platform establishes a foundation for innovative approaches to auditory science education and broader public engagement.

1aPP9. Audiovisual modulation in multi-agent motion dynamics during virtual world co-navigation in immersive rooms. Mincong Huang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, huangm5@rpi.edu), Stefan Radev, and Jonas Braasch (Rensselaer Polytechnic Inst., Troy, NY)

Temporal modulation of auditory and visual motion perception has been extensively studied using personalized virtual reality systems, i.e., head-mounted devices. However, the experiential conditions of headsets limit the ecological validity of such modulation to an egocentric reference frame, and its variability under the physical co-presence of other individuals is often overlooked. In this work, we investigate the influence of moving audiovisual stimuli on human agent movement in immersive rooms during co-navigation of virtual worlds. We situate our work in three game-based environments resembling real-world scenarios, each populated with virtual soundscapes and salient visual and bimodal objects, the so-called spatial beacons. In our approach, we first simulate the agents' head and body movements in the immersive room as a variant of the drift-diffusion model (DDM). Using an integrated digital twin, we then capture motion-tracking data for human participants as they freely explore the spatially augmented environments *in situ*. By comparing simulated and real data, we observe the modality-specific influence of the moving beacons in modulating the emergent motion patterns of the human agents, from which we yield insights about how human interaction in the immersive rooms is shaped by the spatial perception of their augmented surroundings. [Work supported by NSF IIS-1909229 & CNS-1229391.]

1aPP10. Assessing soundscape perceptions of patients and healthcare staff in a University Health Center. Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Bilkent University, Faculty of Art, Design and Architecture, Dept. of Interior Architecture and Environ. Design, Ankara 06800, Turkey, semiha@bilkent.edu.tr) and Oya Yıldız (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

Assessing soundscape perceptions in healthcare environments is significant in improving well-being and functionality. However, understanding how different user groups perceive these environments remains a challenge. This study aims to assess audio-visual perceptions of patients and healthcare staff (doctors and nurses) in a university health center located in Bilkent University Campus, Ankara, Turkey, across three room types: waiting area, pre-examination room, and doctors' office. The research employs an ISO 12913-2-based questionnaire with Turkish-adapted soundscape affective attributes and Equivalent Continuous A-Weighted Sound Level (LAeq) measurements, audio signal processing, and MATLAB-based image analysis. The findings revealed that the soundscape quality was predominantly perceived as "monotonous" and "uneventful." The study seeks to identify differences between patient and staff perceptions and examine how acoustic and visual factors influence perceived pleasantness.

1aPP11. Abstract withdrawn.

1aPP12. Individual factors influencing outcomes with directional hearing aid processing. Varsha H. Rallapalli (Univ. of South Florida, 2240 Campus Dr., Evanston, IL 60208, varsha8904@gmail.com), Sophia Kreismer, Naudy Portalatín-Miranda (Univ. of South Florida, Tampa, FL), and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Individuals with hearing loss struggle with communication in noise, even with specialized hearing aid processing. Underlying perceptual and cognitive abilities play a crucial role in speech-in-noise listening, yet much less is understood about how these abilities support performance with hearing aids. In this study, we investigate how these individual abilities affect speech recognition with directional hearing aid processing, intended to reduce signals from off-axis locations. Participants completed a sentence recognition-in-noise task with wearable hearing aids in omnidirectional processing or binaural beamforming settings. On- (0°) or off-axis (+90°) target sentences were mixed with gender-matched two-talker maskers that were either spatially separated or co-located. Participants also completed a comprehensive test battery to assess individual cognitive (working memory, processing speed) and perceptual (binaural, monaural processing) abilities. Results-to-date from 20 listeners show an interaction between target location, noise location, and spatial hearing aid setting. Specifically, speech recognition with the beamformer is poorer than omnidirectional processing when the target is off-axis in spatially separated noise. Participants were clustered based on the cognitive and perceptual test battery to reveal underlying patterns of individual differences in performance across test conditions. The study has implications for optimizing outcomes with directional hearing aid processing. [Work supported by NIH.]

1aPP13. Do talker effects on auditory segregation vary with selection demands? Sahil Luthra (Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, sahil.bamba.luthra@gmail.com), Hee So Kim, Wusheng Liang, Abigail Noyce, and Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA)

In naturalistic communication contexts, listeners must selectively attend to an auditory source (e.g., a person of interest) while ignoring competing auditory information. Successfully attending requires listeners both to segregate auditory objects in the scene (i.e., determine which portions of the auditory signal correspond with which sources) and to select the target stream. Previous research suggests that auditory segregation can benefit from talker differences between streams; however, it remains unclear whether this benefit might interact with demands on stream selection. In this EEG experiment, we employ a 2x2 design to examine how talker differences might interact with selection demands. Specifically, we measure memory for and electrophysiological responses to a target stream (a male voice producing ba/da/ga syllables) in the context of different kinds of distractors. To examine the influence of talker differences on auditory segregation, we manipulate whether the distractor speech is produced by the same or a different (female) talker. To test how this effect may vary with selection demands, we manipulate whether the distractor stream comprises syllables (high demands on target selection) or digits (low selection demands). Results provide insight into the perceptual and neural mechanisms that support auditory selective attention.

1aPP14. Does the method matter? Quantitative and qualitative perspectives in hearing-aid gain personalization. Janin Benecke (Hearing Sci. - Scottish Section, Glasgow, United Kingdom) and William M. Whitmer (Hearing Sci. - Scottish Section, Level 3, New Lister Bldg., Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, bill.whitmer@nottingham.ac.uk)

In conventional hearing-aid personalization, clinicians cannot hear how their patients hear, and patients often cannot reliably detect or describe what they hear. Letting individuals self-adjust gains could overcome these issues but requires methods to be effective, efficient, and easy. To better understand self-adjustments, the current study evaluated how interfaces and stimuli affect self-adjustment as well as how individuals navigate their way through the personalization process. Current hearing-aid users repeatedly adjusted the gain of music, speech, and speech-in-noise excerpts from an individual prescription to their preference. In a second session, the same participants verbalized their thoughts while adjusting to preference the same stimuli. Both tasks used three interfaces with 1, 2, and 3 user-controls (±18 dB) representing different bass-mid-treble configurations. Participants' self-adjusted gains were generally reliable, invariant across stimuli and could be clustered into three patterns across individuals: increased relative bass, overall reduction, and near initial prescription. Analysis of verbalizations revealed two thought-behavior archetypes: exploratory and anticipatory. While it can be difficult to predict individuals' preferences, each individual may apply similar self-adjustment behaviors across stimuli and interfaces. Identifying and promoting these behaviors can provide more efficient and effective hearing-aid personalization. [Work supported by the Medical Research Council & GN Hearing A/S.]

1aPP15. Evaluation of a deep-learning method for implementing wide dynamic range compression and noise reduction simultaneously. Brian C. Moore (Cambridge Hearing Group, Psych., Univ. of Cambridge, Dept. of Psych., Downing St., Cambridge CB2 3EB, United Kingdom, bcjm@cam.ac.uk), Huiyong Zhang, Xiaodong Li, and Chengshi Zheng (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Most hearing aids incorporate multi-channel wide dynamic range amplitude compression (WDRC) to compensate for the reduced dynamic range associated with sensorineural hearing loss. In theory, fast-acting WDRC should be used, because the reduced dynamic range of hearing is thought to be largely caused by reduced fast-acting compression in the cochlea. However, fast-acting WDRC has undesirable side effects, including "cross-modulation": two sound sources (e.g., speech and noise) that are independently amplitude modulated have envelopes that are partially correlated after WDRC is applied. This hinders the perceptual separation of the sources. A deep-learning method, NN-WDRC, is described in which the speech and noise are estimated separately, fast compression is applied to the speech, and slow compression to the noise. The compressed signals are combined with a controllable amount of noise reduction. The whole system was implemented in a low-complexity network, which was trained using many talkers, audiograms, and types of background noise; the prescribed gains and compression ratios for each audiogram were based on the CAM2 fitting method. Technical measures and evaluations using human listeners indicated that NN-WDRC performed better than conventional fast-acting and slow-acting WDRC, and than "signal-to-noise ratio aware" WDRC, especially for non-stationary noises like clicking sounds and a siren.

1aPP16. Spectral weights for localization and speech-in-speech recognition with talkers separated on the horizontal plane. Emily Buss (Univ. of North Carolina at Chapel Hill, 170 Manning Dr., G190 Physicians Office Bldg, Chapel Hill, NC 27599, ebuss@med.unc.edu) and Richard Freyman (Commun. Disord., Univ. of Massachusetts, Amherst, MA)

Spatial hearing conveys information about source location and supports speech recognition in the context of spatially separated background talkers. However, some data indicate a dissociation between localization and spatial release from masking, suggesting that these abilities could be mediated by different auditory mechanisms. The present experiments evaluated spectral weights for these two tasks by filtering stimuli into 1-octave-wide bands and dispersing those bands on the horizontal plane. Target stimuli were 100-ms bursts of speech-shaped noise or single-syllable words, and maskers (when present) were sequences of words. Young normal-hearing participants were asked to judge target location or report back the target word; for speech-in-speech recognition, participants had to rely on spatial cues to perceptually segregate the target from the masker. Localization data were similar for noise bursts in quiet, words in quiet, and words embedded in a sequence; in all cases, weights peaked around 500–1000 Hz. Weights for correct word recognition also relied predominantly on spatial cues in the 500- and 1000-Hz bands, but trial-by-trial data suggest that correct recognition was not closely related to differences in perceived location of target and masker speech for all participants.

1aPP17. Revisiting binaural release from energetic and informational masking. Emily Buss (Univ. of North Carolina at Chapel Hill, 170 Manning Dr., G190 Physicians Office Bldg, Chapel Hill, NC 27599, ebuss@med.unc.edu), Virginia Best, Christine R. Mason, and Gerald Kidd (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

Binaural release from energetic versus informational masking is typically described as relying on different cues. Whereas speech-in-noise recognition is thought to benefit from low-level binaural difference cues, speech-in-speech recognition is said to improve when differences in perceived location facilitate selective attention to the target. This suggests that binaural masking release might be maximal under different conditions for maskers that are dominated by energetic versus informational masking. Data were collected from normal-hearing adults to test this expectation. Sentence recognition was evaluated in speech-shaped noise and in two-talker speech, with binaural cues created by manipulating interaural phase (0- or π-phase) or interaural time (0 or 660 μs). The results replicate previous studies showing that a robust masking release is achieved for a variety of interaural configurations. The magnitude of the release for speech-in-speech recognition does not appear to be maximal for configurations producing well-localized sound images, however. This result raises questions about the fundamental differences between binaural release from energetic and informational masking.

1aPP18. Facial expression portrait: A non-verbal and interactive approach for subjective evaluation of urban sound perception. Yuxin Yin (Harbin Inst. of Technol., 92 Xidazhi St., Nangang District, Harbin City, Heilongjiang Procince, Harbin University of Technol., Harbin 150006, China, 24B334010@stu.hit.edu.cn) and Qi Meng (Harbin Inst. of Technol., Harbin, China)

Subjective evaluation methods in sound perception research often depend on text or scale-based questionnaires, which language translation issues and the limitations of predefined response options may hinder. Facial expressions, as visual representations of emotions, present an alternative that mitigates comprehension biases stemming from linguistic and cultural differences. This study introduces a non-verbal, interactive approach where participants convey their sound perception through modulation of facial action units, resulting in facial expression portraits. The experiment utilized three typical urban sound recordings, with facial expression portraits developed on the Character Creator 4. Additionally, the participants also submitted subjective questionnaires for further comparative analysis. The study aims to validate this methodology's effectiveness and investigate how sound perception influences facial expression portraits. The results indicate that, first, facial expression portraits serve as an effective tool for assessing urban sound perception. These portraits successfully captured participants' reactions to various sound sources, revealing significant differences in valence among the sounds. Second, notable differences appear in the facial action units AU1, AU2, AU4, AU12, and AU15 across the different sound sources. Furthermore, these five activity units demonstrated significant correlations with subjective loudness, acoustic comfort, and the overall assessment of the acoustic environment.

1aPP19. Investigating contextual effects in sound externalization. Robin Duclermortier (LTDS, ENTPE, 3 rue Maurice Audin, Vaulx-en-Velin 69120, France, robin.duclermortier@entpe.fr), Fabien Perrin (CAP, CRNL, Lyon, France), and Mathieu Lavandier (LTDS, ENTPE, Lyon, France)

This study examines whether the perceived level of externalization of a virtual sound source (internalized versus externalized) can be influenced by the context, and in particular by the perceived level of externalization of a preceding source. Participants listened to pairs of sounds and had to judge whether the virtual source of the second sound (the target) was internalized or externalized. The first sound was either fully externalized (produced by a loudspeaker on the left of the participant) or internalized (reproduced diotically through headphones), while the target was reproduced through headphones as if coming from one of four loudspeakers in the room, at two distances (80 cm or 3 m) on the left or in front of the participant. Perceived externalization was, as expected, stronger for sounds originating from the side compared to those in front, and for distant sources compared to nearby ones, yet no significant influence of the preceding sound's externalization level was found. Further Bayesian analyses are planned to confirm whether an absence of effect can be concluded.

1aPP20. Gated word recognition in cochlear implant listeners: The role of spectral resolution and cognitive abilities. Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., 326 Stevenson Ln., B8, Towson, MD 21204, cpatro@towson.edu), Ariana Bennaim (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD), and Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., Towson, MD)

In a word gating task, listeners are presented with increasing amounts of word-onset information through progressively longer temporal "gates." After each gate, they indicate what they believe the target word is. Listeners with normal hearing (NH) typically recognize a target word before hearing its entirety, while listeners with cochlear implants (CIs) require substantially more word-onset information—often nearly the entire word—to identify it accurately. This study investigated differences in sensory and neurocognitive functioning between high- and low-performing CI users on the gated word recognition task. Sensory perception was assessed using the Spectral-Temporally Modulated Ripple Test (SMRT), a measure of auditory spectral resolution. Neurocognitive functioning was evaluated through the Abbreviated Reading Span Test for working memory and the Trail Making Task for executive function. Initial findings showed that differences in auditory spectral resolution, as measured by the SMRT, were the primary factor distinguishing high- and low-performing CI users. In contrast, non-verbal cognitive measures of working memory and executive function contributed minimally to performance differences. Additional analyses are ongoing, and the presentation will explore the relative contributions of auditory spectro-temporal resolution and cognitive factors to the challenges faced by CI users in gated word recognition tasks.

1aPP21. Effect of sound image spatial splitting caused by interaural differences on speech intelligibility. Ryotaro Hanaki (Toyama Prefectural Univ., 5180, Imizu, Toyama 939-0398, Japan, u454019@st.pu-toyama.ac.jp), Daisuke Morikawa, Parham Mokhtari, and Satoshi Okazaki (Toyama Prefectural Univ., Imizu, Japan)

This study aimed to clarify the effect of sound image splitting perception by Interaural Time Difference (ITD) and Interaural Level Difference (ILD) on speech intelligibility. Thus, different ITDs or ILDs were applied to target (speech) and masker (pink noise) sounds, and mora intelligibility was measured. The sound stimuli were prepared such that either the sum of target and masker ITD values equaled 0, 0.1, 0.2, 0.4, 0.6, 0.8 ms, or the sum of target and masker ILD values equaled 0, 5, 10, 15, 20, 25, 30 dB. The results of the ITDs-added experiment showed that the mean mora intelligibility

improved from around 0.2 ms for ITDs, consistent with the detection limit of spatial splitting of sound images. In the ILDs-added experiments, the mean mora intelligibility improved even when ILDs were smaller than the detection limit of spatial splitting of sound images, owing to reduced unilateral masking. Therefore, the ILDs-added experiment was conducted under conditions where unilateral masking was not reduced by adding ILDs. As a result, the mean mora intelligibility improved at ILDs around 10 dB and worsened above 15 dB. These two values are respectively near the detection limit and saturation point of spatial splitting of sound images.

1aPP22. Effects of head movement on distance localization of sound image. Daisuke Morikawa (Toyama Prefectural Univ., Imizu, Japan, dmorikawa@pu-toyama.ac.jp) and Parham Mokhtari (Toyama Prefectural Univ., Imizu, Japan)

To clarify the effect of head movement on distance perception of sound images, distance perception experiments were conducted under head-still and head-moving conditions by randomly presenting stimuli of different sound pressure levels from several distances. The A-weighted sound pressure levels of the stimuli were 40, 35, and 30 dB at the center of the head, regardless of the distance at which the sounds were presented. The stimuli were presented from 20 to 160 cm from the center of the head, and the presented direction was frontal or diagonally front. In the frontal head-still condition, the perceived distance was approximately constant for the same listening level, with higher levels perceived closer and lower levels perceived further away. In the diagonal head-still condition, differences in distance were more perceptible compared to the frontal condition; however, lower levels of near-distance stimuli tended to be perceived further away. By comparison, in the head moving condition, the distance localization accuracies improved. In particular, the mean response and presentation distances were almost the same in both directions in the range of 20–60 cm. Therefore, it was found that head movement had a positive impact on sound image distance perception.

1aPP23. Comparison among four psychophysical procedures used to assess sound localization. Mark A. Stellmack (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, stell006@umn.edu), Stanley Sheft (Dept. of Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., London, ON, Canada), and Adam J. Sipprell (Psych., Univ. of Minnesota, Minneapolis, MN)

Yost *et al*. (1974; Percept. Psychophys. 15, 483–487) measured the effect of psychophysical procedure to evaluate movement-related cues for auditory lateralization. A similar approach was used in the current work to assess sound-source localization in a 10-foot x 13-foot semi-anechoic sound field. Performance was measured in four listening conditions, a localization task utilizing a pointer response and three location-discrimination tasks (single-interval, same-different, or 2AFC). Four signals were used: a 750-Hz tone, broadband noise, or two versions of broadband noise with narrowband levels roved $+/-20$ dB in 1/3-octave bands between intervals, or frozen across intervals with the same rove. Stimuli were generated as phantom sources using tangent-law panning between $+/-30^{\circ}$ azimuth loudspeaker positions. For each signal type, d' was lower in the pointer condition than in the three discrimination tasks. Across tasks, performance was best for broadband noise and noise frozen across intervals, relative to the pure-tone and fully roved signals. As suggested by Yost *et al*., task effects may relate to source movement cues present in only the two-interval discrimination conditions. Alternatively, slow fluctuations in bias may negatively affect only the pointer localization task, with results from single-interval discrimination offering some support for this interpretation.

1aPP24. Head movement in an auditory stream segregation task: Hearing names and digits in a simulated audio-visual environment. Nathan C. Higgins (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, higgins1@usf.edu), Kayla R. Ardizzone, and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Following conversation in a complex environment requires a listener to maintain continuous attention and use other sensory input to maximize speech reception. In a multi-talker conversation, that means turning the head to match the talker-location and using visual cues (e.g., lip and facial expressions). To test the interactions between these processes, we devised an experiment with two competing speech streams. The first stream used materials adapted from a video-podcast with four talkers, each on separate screens, arrayed horizontally with 30° separation. The second stream was composed of monosyllabic words interspersed with digits (CNIT) [Ozmeral *et al*., Int. J. Audiol. 59(6), 434–442 (2020)]. Head movements were recorded during each 90-sec trial. Both streams were presented co-located, at equal SNR, from the same loudspeaker as the active talker in the video-podcast conversation. Before each trial, participants ($N = 13$, young normal-hearing) were cued to either a single-task (detect names <u>or</u> digits) or dual-task (detect names <u>and</u> digits). Significantly higher accuracy was observed detecting digits versus names in both the single- and dual-task conditions. Correlations between head-orientation and talker-location show that as participants' head-orientation followed the talker-location, name-detection accuracy improved while digit-detection degraded. Together, these results demonstrate the complex interactions of head-movement and visual stimuli on segregation of competing auditory streams.

1aPP25. Effects of subclinical hearing loss, noise exposure, and post-exposure changes in hearing on antiphasic tone-in-noise detection. Gregory M. Ellis (Audiol. and Speech Pathol., Walter Reed National Medical Military Ctr., 4494 Palmer Rd N, Bethesda, MD 20814, gellis@alakaina.com) and Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed NMMC, Bethesda, MD)

Most hearing conservation programs in the United States (US) operate under the assumption that meaningful damage to the auditory system does not begin until auditory thresholds are within the range of hearing impairment. Recent evidence from has challenged this assumption, indicating that individuals with audiometric thresholds in the normal range may report hearing difficulty after repeated exposure to noises that result in temporary changes in hearing. The aim of this study was to quantify the effects of hearing loss, noise exposure, and self-reported post-exposure experience of symptoms on a basic tone-in-noise detection test. Over 10,000 US Service Members (SMs) were recruited to participate in this study. SMs completed an audiogram, a noise-exposure survey, a post-exposure symptom survey, and a tone-in-noise detection test. SMs were asked to detect a 500-Hz tone presented 180 deg out of phase across the ears. The tone was embedded in diotic broadband noise. SNR on the 14 trials was varied between -29 and -1-dB SNR, and proportion correct was calculated as the outcome variable. Hearing loss, self-reported post-exposure symptoms, and noise exposure history were all significant predictors of tone detection; however, post-exposure symptoms were stronger predictors of performance than mere noise exposure.

1aPP26. Comparison of speech perception in virtual and real-life acoustic environments. Hendrik Kayser (Univ. of Oldenburg, Oldenburg, Germany, hendrik.kayser@uni-oldenburg.de), Theresa Jansen (Horzentrum Oldenburg, Oldenburg, Germany), Volker Hohmann (Univ. of Oldenburg, Oldenburg, Germany), and Erik Jorgensen (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

Recent years have seen growing interest in using virtual acoustic scenes in laboratory environments to emulate real-life complex listening environments. Questions remain, however, regarding the degree to which virtual environments reflect typical complex listening environments in daily life. In this study, participants with and without hearing loss completed speech perception testing in eight virtual acoustic scenes. Then, participants were sent home for a week with a hearing aid research platform, the Portable Hearing Lab (PHL), and instructed to seek out situations that represented their typical complex listening environments. The PHL was used to record the environment, and ecological momentary assessment was used to characterize the environment, including the location, number of noise sources, orientation of speech and noise, and access to visual cues. Acoustic, non-acoustic, and speech perception differences between virtual and real-life complex environments were compared. Although most of the complex environments in daily life were encountered at home, the virtual scenes were fairly representative in terms of the sound levels, noise sources, noise locations, and

signal-to-noise ratios. Speech perception scaled similarly between the virtual environments and real life as a function of sound level and acoustic complexity. The results generally support the ecological validity of virtual acoustic scenes.

1aPP27. Comparison of spatial release from masking threshold estimation methods. Tess K. Koerner (VA RR&D NCRAR, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, koern030@gmail.com), Esteban Sebastian Lelo de Larrea-Mancera (Psych., Northeastern Univ., Boston, MA), Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, Columbia, MD), William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Recent efforts have focused on developing auditory assessments that are accessible to a wider range of researchers and clinicians without the need for specialized auditory equipment or expertise. Different threshold estimation methods have been developed in an effort to facilitate more efficient, automated measurements for this purpose. The current work tested whether a commonly used, simple approach to measuring performance on a spatial release from speech-on-speech masking task was equivalent to a procedure that estimates performance from individual psychometric functions. In addition, this work aimed to compare threshold estimates across a computer-based implementation of this task in MATLAB and a tablet-based implementation using the Portable Automated Rapid Testing (PART) app. Spatial release from speech-on-speech masking was assessed using the Coordinate Response Measure (CRM) speech corpus with a progressive tracking procedure that decreased the target-to-masker ratio (TMR) from + 10 to −8 dB in 2-dB steps with each TMR condition repeated twice. Analysis was completed on a dataset consisting of 93 participants who completed testing in MATLAB and a dataset of 44 participants who completed testing in PART. Participants varied in age and hearing sensitivity. Results have implications for the use of portable, automated methods for assessing spatial release from speech-on-speech masking.

MONDAY MORNING, 19 MAY 2025 — GALERIE 1, 7:55 A.M. TO 11:00 A.M.

Session 1aSP

Signal Processing in Acoustics and Acoustical Oceanography: Acoustic Array Processing and Sound Field Reconstruction I

Efren Fernandez-Grande, Cochair
Department of Electrical Engineering, Technical University of Denmark, B.352, Oersteds Plads, Kgs Lyngby 2800, Denmark

Peter Gerstoft, Cochair
Univ. of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093

Chair's Introduction—7:55

Invited Papers

8:00

1aSP1. Sound source identification with sparse algorithm and machine learning method. Zhenming Cui (Institue of Acoust. Chinese Acad. of Sci., Beijing, China), Tongyang Shi (Institue of Acoust. Chinese Acad. of Sci., Beijing, China), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S. Russell St., West Lafayette, IN 47907-2099, bolton@purdue.edu)

Acoustical holography is a useful tool for reconstructing sound fields and identifying sound sources. In essence, holography is an inverse process, based on using microphone array measurements to estimate the weighting coefficients of a chosen basis function. Then the sound field can be reconstructed by using the determined basis functions, which in turn allows the sound source location to be identified. However, in practice, the number of measurements is usually smaller than the number of coefficients that need to be estimated to perform an accurate reconstruction. To solve this problem, in the past few years, sparse algorithms have been applied to estimate the unknowns in an under-determined problem: e.g., l1-norm minimization, wideband acoustical holography, and Bayes estimation method. In previous studies, it was found these algorithms were able to identify sound source locations based on fewer measurements compared with Fourier transform or other holography methods. The latter has been demonstrated using simulations and experiments in the present work. In addition, it will be shown that machine learning can be applied to identify sound sources in low signal-to-noise cases, e.g., in noisy background or reverberant environments, in which cases it is difficult for other methods to identify sound sources.

8:20

1aSP2. Space-time domain point neuron learning for sound field reconstruction. Thushara D. Abhayapala (Australian National Univ., Canberra, Australian Capital Territory, Australia, thushara.abhayapala@anu.edu.au), AMY BASTINE, and Prasanga Samarasinghe (Australian National Univ., Canberra, Australian Capital Territory, Australia)

Physics Informed Neural Networks (PINNs) have been used in array signal processing allocations in recent years, where physical constraints such as governing partial differential equations (PDEs) and/or boundary conditions have been added to the usual data driven loss function. While these methods have helped to solve certain limitations, they also have several drawbacks such as being unable to approximate PDEs that have sharp gradients or strong non-linearities, not being able to move away from local optimums, and convergence to trivial solutions. Recently, we embedded the fundamental solution to the wave equation, the free space Green function, into the network architecture enabling the learned model to strictly satisfy the physical law of sound propagation. In the proposed network, the basic processing unit is called a point neuron whose weight and biases can be learned by back propagation. The physical meaning of point neuron is equivalent to point sources or plane wave sources, and the weight, location (biases) and distribution of equivalent sources can be updated while training. The proposed point neuron network can be implemented to model and estimate an arbitrary sound field purely based on microphone observations without a pre-existing dataset. Building on the concept of point neuron network, which is defined in space-frequency domain, the current work conceptualises the space-time domain sound field representation problem with the wave equation as a constraint.

8:40

1aSP3. Diffusion-model-based approach for inverse problems in optically measured sound field. Hao Di (Waseda Univ., Nishi-Waseda Campus,3 Chome-4-1 Okubo, Shinjuku City, Tokyo 169-8555, Japan, dihao@toki.waseda.jp), Kenji Ishikawa (NTT, Atsugi, Japan), Risako Tanigawa (NTT/Waseda Univ., Tokyo, Japan), and Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan)

Optical sound-field imaging, known for its high spatial resolution, measures sound by detecting small variations in the refractive index of air caused by sound, but often suffers from unavoidable noise contamination. Sound-field reconstruction and extrapolation aim to recover complete sound-field information from limited or patched observations. The tasks of denoising, reconstruction, and extrapolation of sound field imaged by optical measurement can all be viewed as sound-field inverse problems. To address these issues, we propose a diffusion-model-based approach for solving sound-field inverse problems, encompassing denoising, noisy sound-field reconstruction, and extrapolation. During the inference phase of the diffusion model, sound-field degradations are introduced into the reverse denoising process, with range-null space decomposition employed as the solver to iteratively recover information from the observed sound field. The proposed method is trained on sound-field datasets generated from numerical acoustics simulations with randomized parameters, without the need for labeled degradations. Numerical experiments demonstrate that our method outperforms existing deep-learning-based approaches in both sound-field denoising and reconstruction tasks, while also achieving effective performance in sound-field extrapolation. Furthermore, in practical experiments, our method successfully denoised and reconstructed the optically measured sound field, exhibiting excellent performance.

9:00

1aSP4. Solving the inhomogeneous wave equation using physics-informed neural networks. Samuel A. Verburg (Acoust. Technol., DTU Electro, Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, saveri@dtu.dk), Efren Fernandez-Grande (Polytechnic Univ. of Madrid (UPM), Madrid, Madrid, Spain), and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Physics-informed neural networks (PINNs) have emerged as an effective framework for solving forward and inverse problems involving the wave equation. Current PINN formulations primarily address the homogeneous wave equation, where the forcing term is null. Consequently, the absorption or injection of acoustic energy is either unaccounted for or implicitly introduced as boundary conditions. In this work, we explore the use of PINNs to explicitly solve the inhomogeneous wave equation. By incorporating a forcing term in the wave equation residual, our PINN formulation can readily account for time-dependent sources and sinks of acoustic energy. This approach facilitates the inclusion of moving sources or sources with time-dependent output power. As an example, we solve a sound field reconstruction problem involving dynamic sources moving through a medium.

9:20–9:40 Break

9:40

1aSP5. Spatial post-filtering method for first-order spherical signals. Stefan Wirler (D Information and Communications Eng., Aalto Univ., Espoo, Finland), Nils Meyer-Kahlen (D Information and Communications Eng., Aalto Univ., Espoo, Finland), and Ville Pulkki (D Information and Communications Eng., Aalto Univ., Otakaari 5, Espoo 02210, Finland, Ville.Pulkki@aalto.fi)

A method is presented to enhance the spatial selectivity of spatial post-filters estimated with first-order directional signals. The approach involves applying non-linear transformations on two different spatial post-filters and combining them with weights found by convex optimization of the resulting directivity patterns. The estimation of the post-filters is carried out similarly to the Cross Pattern Coherence (CroPaC) algorithm. The performance of the proposed method is evaluated in a two- and three-speaker scenario with different reverberation times and angular distances of the interfering speaker. The signal-to-interference, signal-to-distortion, and signal-to-artifact ratios are used for evaluation. The results show that the proposed method can improve the spatial selectivity of the post-filter estimated with first-order beampatterns. Using first-order patterns only, it even achieves better spatial separation than the original CroPaC post-filter estimated using first- and second-order signals.

10:00

1aSP6. Benefits of Multi-Channel Cross-Talk Cancellation systems in reverberant spaces. Filippo Fazi (ISVR, Univ. of Southampton, University of Southampton, University Rd., Southampton, Hampshire SO171BJ, United Kingdom, Filippo.Fazi@soton.ac.uk) and Jacob Hollebon (Audioscenic, Southampton, United Kingdom)

Cross-Talk Cancellation (CTC) systems are designed to deliver independent signals to the two ears of a listener using two or more loudspeakers. This is achieved by controlling the constructive and destructive interference pattern of the sound fields generated by the loudspeakers at the ears of the listener. Destructive interference causes a reduction of the sound pressure level (SPL) delivered to the listener and reduces the CTC performance when the system is arranged in a reverberant environment. This study demonstrates that the number of loudspeakers controls the balance between constructive and destructive interference required for a given target CTC level. A greater number of loudspeakers reduces the amount of destructive interference, thereby decreasing the overall radiated power (for a given SPL at the listener's ears) and mitigating the CTC degradation due to room reflections. These findings suggest that multi-channel configurations are more effective in maintaining CTC efficacy in reverberant spaces.

10:20

1aSP7. High-resolution imaging of impulse response by high-speed polarization interferometer. Yuzuki Saito (Waseda Univ., RM 407-2 (4F), Bldg 59, Nishiwaseda Campus, 3-4-1 Ookubo, Shinjuku-ku, Tokyo, 169-8555, Japan, Shinjuku, Tokyo 169-8555, Japan, yuzuki@fuji.waseda.jp), Kenji Ishikawa (NTT, Atsugi, Japan), Risako Tanigawa (NTT/Waseda Univ., Tokyo, Japan), and Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan)

Traditionally, microphones have been widely used for sound-field measurements due to their ability to accurately capture sound pressure at specific locations. However, their physical presence interfere with the target sound fields, particularly when precise measurements of subtle variations are required. To address these challenges, optical acoustic measurement techniques, such as the high-speed polarization interferometer utilizing parallel phase-shift interferometry (PPSI), have been proposed. PPSI has been successfully applied to the measurement of aerodynamic sound, sound radiated from fast-moving sources, musical-instrument sound, and determination of acoustic centers. Despite these advancements, PPSI has not yet been utilized for measuring impulse responses of loudspeakers. In this study, we propose high-resolution imaging of impulse response using PPSI. We measured two-dimensional spacial distribution of time stretched pulse (TSP) sounds emitted from a loudspeaker by PPSI and calculated the impulse response by convolving them with an inverse TSP signal in each pixel. We found that this method provides highly accurate, non-contact measurement of the spatial characteristics of impulse responses, offering valuable insights for improving speaker design and tuning accuracy.

10:40

1aSP8. Gridless sparse beamforming with Wirtinger gradients. Yongsung Park (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, yongsungpark@ucsd.edu), Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA), and Christoph Mecklenbräuker (TU Wien, Vienna, Austria)

A gridless sparse beamforming (direction-of-arrival (DOA) estimation) method using gradient-based optimization is presented. The approach minimizes the fit between the sample covariance matrix (SCM) and a reconstructed covariance matrix constrained to contain only a few atoms. This enables analytic derivatives using Wirtinger gradients for efficient optimization. The sensitivity to local minima is mitigated by initializing with optimal DOAs from a user-input-free gridded sparse Bayesian learning. Numerical simulations demonstrate the method provides superior resolution compared to conventional approaches.

Session 1aUW

Underwater Acoustics: General Topics in Underwater Acoustics

Andrew R. McNeese, Chair
ARL:UT, 10,000 Burnet Rd, Austin, TX 78758

Contributed Papers

7:40

1aUW1. Optimization of the matching layer for high frequency hydrophones. Xinyue Man (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University, Harbin, Heilongjiang Province 150001, China, manxinyue@hrbeu.edu.cn) and Chunying Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang Province, China)

This study explores the optimization of matching layers for high-frequency hydrophones operating at resonant frequencies, with the aim of enhancing sensitivity, bandwidth, and signal transmission efficiency. Single, double, and triple matching layers are applied to piezoelectric composite disc, and tone burst signal with varying number of burst cycles. The results indicate that the matching layer acts as a bandpass frequency filter under both two excitation signal types. In the short pulse case, the matching layer can improve bandwidth and sensitivity simultaneously, whereas for the case of long pulse excitation, the increased bandwidth is at the expense of sensitivity. Multilayer designs significantly reduce acoustic reflections and broaden the operational bandwidth, thereby improving reception performance in high-frequency applications. In contrast, the single-layer structures offer better directivity but have a narrower bandwidth at lower frequencies. Through numerical simulations, the study underscores the importance of material properties, layer thickness, and acoustic impedance optimization in enhancing the performance of hydrophones. These findings provide valuable theoretical insights for the design of high-frequency hydrophones, advancing their applications in underwater communication, environmental monitoring, and sonar systems.

8:00

1aUW2. Acoustic scattering from inert underwater munitions with severe corrosion damage. Connor M. Hodges (Chandra Dept. of Elec. and Comput. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 1200 Barton Hills Dr., 167, Austin, TX 78704, cmhodges@utexas.edu), Charles Hubbard (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Vikalp Raj (Texas Mater. Inst., Univ. of Texas at Austin, Austin, TX), David Mitlin (Walker Dept. of Mech. Eng., Texas Mater. Inst., and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

More than 400 underwater sites have been identified by the U.S. Government as potentially contaminated with unexploded ordinance (UXO), posing a potential threat to human safety. While some of these munitions are more recent, some can date as far back as the 18th century, thus having the potential for severe corrosion and biofouling damage. We hypothesize that as UXO age for years or decades in the underwater environment, accumulated biofouling and corrosion will lead to an increasingly larger deviation of the acoustic scattering signature from the pristine state. To explore these relationships, we measured the free-field frequency- and aspect-dependent acoustic scattering response of recently recovered World War II era miniature practice bombs (model AN-Mk 23) that exhibit a range of corrosion and minimal biofouling, due to decades of environmental exposure in a brackish marsh. The acoustic responses of pristine, uncorroded examples of AN-Mk 23 ordnance were also measured. We found significant shifts in the aged samples' resonance frequencies, aspect variation, and overall target strength level. Analytical material characterization and high-resolution imaging techniques were employed to determine how material loss and the formation of specific biofouling–corrosion products correlate to the modified acoustic signatures. [Work sponsored by SERDP and DOD SMART Scholarship.]

8:20

1aUW3. Toward simulation of long-term corrosion effects on acoustic scattering responses of underwater unexploded ordnance. Charles H. Hubbard (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Appl. Res. Labs. The University of Texas at Austin P.O. Box 9767, Austin, TX 78766-9767, charliehubbard@me.com), Connor M. Hodges (Chandra Dept. of Elec. and Comp. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Underwater unexploded ordnance (UXO) remediation faces the challenge of detecting and classifying objects that cannot easily be sensed using optics or electromagnetics. Instead, active sonar can be used to identify these munitions, which may be proud on the seabed, partially or fully buried in sediment. Historically, munitions were fired or discarded into oceans and inland waterways, presenting challenges for identifying ordnance exposed to long-term corrosion and biofouling. A long-term goal is to develop simplified models of corroded UXO with the intention of generalizing the model to expedite remediation efforts for a variety of munitions and environments without the need for physical testing of the many possible outcomes of long-term exposure. Working toward this objective, we are developing finite element (FE) acoustic scattering models to predict UXO acoustic signatures. A preliminary set of models will be presented in this talk with a focus on simplified geometries. Geometries were cylinder and truncated cone shapes with combinations of notches and cylindrical cuts along the axis of rotational symmetry. Free-field acoustic scattering experiments (1–100 kHz) were performed in an underwater test tank facility to examine acoustic effects of individual geometric features and to verify the accuracy of FE models. [Work supported by SERDP.]

8:40

1aUW4. Mechanism study of phase-transformed single crystal-driven transducers. Xuan Yin (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University, Harbin, Heilongjiang Province 150001, China, yinxuan2017@hrbeu.edu.cn) and Chunying Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang Province, China)

The single crystal vibrators in the Tonpilz transducers are inevitably subjected to uniaxial stress, which will result in the changes of properties.

However, few studies have been conducted to characterize the performance of single crystal under uniaxial stress. Besides, it indicates that the uniaxial compressive stress with an appropriate amplitude can induce phase transition from R to O phase in PIN-PMN-PT single crystals, leading to an increase in d_{32}. Therefore, it triggers the idea that the transducers can be driven by the phase-transformed single crystal. This study aims to investigate the novel mechanism of phase-transformed single crystal-driven transducers. A home-built setup was developed to measure the material's performance under uniaxial stress. Meanwhile, the performance of the Tonpliz transducer driven by a d_{32} vibrators was measured. An impedance analyzer was used to monitor the impedance and capacitance curves. The vibration displacement of the acoustic radiation surface was measured by a laser vibrometer. The results illustrate that the amplitude of the conductivity increases 10%, which indicates the occurrence of a phase transition. Finally, a transducer driven by phase-transformed single crystal vibrators is proposed, which provides innovative ideas for the design of transducer with small volume and high sound source level.

9:00

1aUW5. Experimental demonstration of passive acoustic identification tags for AUV localization and wireless backscatter communication. Nizar Somaan (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr NW, J. Erskine Love Bldg., Office 131 & Rm. 130, Atlanta, GA 30332, nsomaan3@gatech.edu), Ananya Bhardwaj, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Accurate positioning is critical for autonomous underwater vehicles (AUVs), particularly during homing and docking. Passive Acoustic Identification (AID) tags offer a scalable, cost-effective alternative to active acoustic and optical methods, excelling in turbid and cluttered environments. These tags guide AUVs by generating distinct acoustic signatures to improve localization and navigation accuracy. This study demonstrates long-range experiments with larger AID tags (up to 24-inch diameter) in a lake environment, tested at frequencies between 100 and 600 kHz. Results show detectability at distances exceeding tens of meters, with optimized signal processing techniques enhancing robustness in cluttered conditions. Furthermore, an active tag design for wireless backscatter communication was implemented. Powered by ultrasonic energy transfer from an interrogating AUV, the tag harvests energy and communicates data via single-frequency modulation, achieving data rates of up to 200 kb/s even in low signal-to-noise environments. These advancements establish AID tags as a dual-purpose solution for improving AUV localization and enabling robust communication in complex underwater missions.

9:20–9:40 Break

9:40

1aUW6. Comparisons of amplification and frequency output of multiple manufactured gas bubbles and gas bubble clusters of different designs, configuration, and material. Luke H. Prentice (CUE, Univ. of Strathclyde, 99 George St., Glasgow, Scotland G1 1RD, United Kingdom, luke.h.prentice@strath.ac.uk) and James Windmill (Univ. of Strathclyde, Glasgow, United Kingdom)

Previous work has shown underwater gas bubbles to be an amplification mechanism for weak-sources outputting at the bubble's Minnaert resonance. Here further work to capture bubbles in manufactured shells and cages is reported. 3-D printed resin cages with direct air–water interfaces, and rubberized shells of specified volume and resting tension and a downward facing pipe-based system, were created. The effect these have on the resonance of the amplification system and amplitude of the output is analyzed. These devices are miniscule compared to other low-frequency devices of a similar center frequency, for example, a 500-Hz cage has a 7-mm radius sphere. Another key feature explored is utilizing inter-bubble coupling at different distances, configurations, and bubble volume deltas as a means of frequency output control and the effect that the different bubble-holding systems have on this coupling. Controlling the exact frequency response of this system is important as one of the major applications for these devices will be as a finely tuned underwater sensor for weak signals in noisy environments.

10:00

1aUW7. Adaptive Robust Capon Beamforming with enhanced beam continuity. Hyung-In Ra (Korea Martime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Busan, 49112, Rep. of KOREA, Busan KS012, Korea, babavivi@g.kmou.ac.kr), Ji-Hyun Lee (Korea Maritime & Ocean Univ., Busan, Korea), and Ki-Man Kim (Korea Martime and Ocean Univ., PUSAN, Korea)

MVDR (Minimum Variance Distortionless Response) beamforming is widely used for its high-resolution capability in suppressing interference and noise. However, it is highly sensitive to steering vector errors, which can occur due to environmental uncertainties or calibration issues. Even small deviations in the steering vector can result in significant performance degradation, which can reduce the reliability of MVDR in situations where deviations occur frequently or channel variability is high. Robust Capon Beamforming (RCB) was previously developed to address this issue. RCB provides greater robustness by increasing the tolerance to steering vector errors. While it results in a more robust beamform compared to MVDR, it nevertheless has its limitations. In particular, even if the robustness is increased by moderately increasing the beamwidth by tolerating errors, the possibility of beam discontinuities appears in the beamforming. This discontinuity can be a serious problem for situations where the reliability of signal tracking is important. To solve this problem, we propose an adaptive RCB method. The proposed method can secure both robustness and beam continuity by adjusting the parameter based on the beamforming results. This method improves the practicality of RCB even in relatively uncertain environments by maintaining tracking in a specific desired direction.

10:20

1aUW8. Localized Maxwell-stress excitation of low-frequency modes of electrically conducting objects in water in reverberant environments. Christopher T. Powers (Phys. & Astronomy, Washington State Univ., Washington State University, Phys. & Astronomy Dept., Pullman, WA 99164-2814, christopher.t.power@wsu.edu), Sterling M. Smith, Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA), Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Ahmad T. Abawi (HLS Res., San Diego, CA)

When testing some specialized sonar systems, it can be helpful to have alternative methods of exciting low-frequency (LF) modes of objects to be studied. This is especially the case in reverberant environments. One non-contact approach for driving modes of electrically conducting solid objects is to excite eddy currents which cause oscillatory Maxwell stresses on objects [B. T. Hefner and P. L. Marston, J. Acoust. Soc. Am. 106, 3340–3347 (1999)]. In the present application, the stress is localized using an appropriate coil-magnetic-core system close to the object. Electric-current tone bursts in the coil cause stresses at twice the frequency of the current. Frequencies and locations are selected to drive modes of interest. This method has been used to excite LF flexural and compressional modes of objects of interest in freshwater-lake as well as laboratory-tank experiments. Some of the objects investigated include spherical and cylindrical shells, solid cylinders, and an aluminum UXO model. In the lake-based experiments, the radiated sound is detected using an array of hydrophones, though in the tank-based experiments it can be convenient to scan a hydrophone. Complications in the responses associated with environmental reverberation can be investigated. [Work supported by the U.S. Office of Naval Research.]

10:40

1aUW9. Modeling signal coherence for a large number of samples. Laura Brownstead (Graduate Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, lgb5113@psu.edu), Jason Philtron (Appl. Res. Lab., Penn State, University Park, PA), Chad M. Smith, and Daniel C. Brown (Graduate Program in Acoust., Penn State, State College, PA)

Seabed-reflected sound measurements allow researchers to better understand the ocean environment based on the received signal's structure. This signal structure is affected by seabed characteristics, like seafloor slope, sediment impedance, and sediment layering. The seabed composition influences both the coherent (signal phase is preserved) and incoherent (signal

phase is disrupted) components of the scattered signal. This environmental influence can be studied with the spatial coherence of the scattered field, which is estimated by the magnitude of the complex correlation coefficient ($|\rho|$) between channels in a normally directed receive array. This work's primary focus is applying the PDF of the coherence of the scattered field to understand the ocean environment. However, measurements of interest capture transient seafloor properties, like sediment layering, that are non-uniformly powerful; these variations occur over large scales, which increases the number of measurement points per transmission without maintaining joint independence. To model transient returns with the magnitude coherence PDF (which assumes joint independence), an equivalent number of independent samples is derived and validated for the PDF. Results are presented for multi-channel synthetic data (white Gaussian noise) and for seabed scattered signals; measurements were made with a Kongsberg SBP-29 aboard R/V Sally Ride in 2024.

MONDAY AFTERNOON, 19 MAY 2025 GALERIE 2, 1:00 P.M. TO 5:00 P.M.

Session 1pAAa

Architectural Acoustics: Acoustics of Sustainable Building Assemblies and More II

Jonathan M. Broyles, Cochair
Civil, Environmental and Architectural Engineering, The University of Colorado Boulder, Boulder, CO 80020

Arthur W. van der Harten, Cochair
Acoustics, Acoustic Distinctions / Open Research in Acoustical Science and Education, 400 Main St. Ste. 600, Stamford, CT 06901

Invited Paper

1:00

1pAAa1. How should acousticians tackle building decarbonization? An academic perspective on solutions to mitigate global warming while achieving high acoustic performance. Jonathan M. Broyles (Civil, Environ. and Architectural Eng., The Univ. of Colorado Boulder, 13582 Via Varra, Broomfield, CO 80020, Jonathan.Broyles@colorado.edu) and Kristen Murphy (Acentech, Cambridge, MA)

Acousticians are becoming increasingly aware of the need to reduce the carbon footprint of buildings. Practical solutions, such as incorporating life cycle assessment in the acoustic design process and the implementation of low-carbon and acoustically viable building products, are becoming more common in acoustic consulting. Despite these decarbonization pathways, many acousticians are still grappling with knowing how to best tackle building decarbonization efforts. To this end, this presentation summarizes the acoustic-decarbonization strategies published in academic papers over the last 25 years. Specifically, this talk will highlight the most common and practical solutions published in academic articles to reduce the embodied carbon and operational carbon emissions of a building while achieving high acoustic performance. This talk will also highlight the specific design contexts that these solutions are most appropriate for. Lastly, opportunities for future research at the intersection of architectural acoustics and decarbonization will be discussed. Overall, this presentation continues the conversation of how acousticians can curb global climate change through the implementation of low-carbon and carbon negative acoustic design solutions.

Contributed Papers

1:20

1pAAa2. 25 years (transparent) micro-perforated sound absorbers. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg, Nds. 26121, Germany, nocke@akustikbuero-oldenburg.de)

The theory of microperforated sound absorbers (MPA) has been introduced by D.-Y. Maa in 1975. Micro-perforated sound absorbers have been used in various applications. Materials that have been used to be micro-perforated have been metal, wood, plastics and many others. A short review of different applications of micro-perforation will be presented. This presentation will concentrate on applications in architectural acoustics. In 2000, the micro-perforation was introduced to transparent sheets making these highly sound absorptive. Over the past 25 years, different set-ups made of micro-perforated layers, porous materials as well as plate resonators have been investigated. In this paper, different applications of various set-ups with micro-perforated stretched foils will be presented. Day-light ceilings, mirror ceilings as well as absorbers in front of glass will be shown as well as newly developed day-light, sound absorptive and heating/cooling systems. The

latest materials used for micro-perforation is a material made from wood and plants.(up to 92% natural ingredients). This replaces petroleum and opens new possibility toward sustainable sound absorption.

1:40

1pAAa3. Assessment of organic and inorganic panels as acoustical treatment in the meeting room. Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Bilkent University, Faculty of Art, Design and Architecture, Dept. of Interior Architecture and Environ. Design, Ankara 06800, Turkey, semiha@bilkent.edu.tr) and Cem Kaan Şen (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

Conventional acoustic materials heavily rely on traditional manufacturing techniques, which can pose environmental threats. As the field of acoustic materials is further explored, it becomes increasingly important to balance sustainability with acoustic performance. Architectural spaces like meeting rooms demand more effective acoustic solutions to ensure optimal reverberation time (T60). This study aims to investigate the acoustic performance of organic (PLA-based) and inorganic (recycled PET-based) micro-perforated panels (MPPs) as sustainable alternatives to conventional acoustic materials. MPPs were produced using 3-D printing technology while considering parameters such as perforation ratio, perforation diameter, panel thickness, and cavity depth. Following the standards of ISO 3382-2 (2008) and ISO 10534-2 (1998), their sound absorption coefficients (α) and effects on T60 were evaluated through *in-situ* measurements and simulation modeling methods, including Odeon Acoustics Software. The initial study indicated that the current acoustic quality in the meeting room was unsatisfactory. Based on existing literature, PLA- and PET-based composite MPPs provided better absorption at mid-to-high frequencies than conventional panels. The study's results are anticipated to show improved performance compared to conventional acoustic panels while encouraging environmentally sustainable material choices.

2:00

1pAAa4. Acoustic absorption of mycelium-based composites cultivated on spent mushroom substrates. Chiara Dognini (Mech. and Industrial Eng. Dept., Univ. of Brescia, Brescia, Italy), Jee Woo Kim (Graduate Program in Acoust., Penn State Univ., 118 Res. West, University Park, PA 16802, jvk6427@psu.edu), Aisa Shams (Dept. of Architecture, Penn State Univ., University Park, PA), John Pecchia (Dept. of Plant Pathol., Penn State Univ., University Park, PA), Yun Jing (Acoust., Penn State Univ., State College, PA), and Benay Gürsoy (Dept. of Architecture, Penn State Univ., University Park, PA)

Cultivating edible mushrooms produces substantial waste—primarily in the form of spent mushroom substrates (SMS)—which poses major environmental and waste management challenges for the mushroom industry. We transform the SMS into mycelium-based composites (MBC) as an effective acoustic biomaterial, where acoustic absorption tests are done for samples in various production cycles utilized in actual mushroom cultivation practices. Our results demonstrate that the SMS acoustic biomaterial reaches absorption coefficients close to 1 at frequency ranges around 1000 Hz, achieving almost complete acoustic absorption. Most samples showed strong absorption at higher frequencies between 2000 and 6000 Hz, with most samples exceeding absorption coefficients of 0.5, confirming the suitability of MBCs cultivated on SMS as an acoustic biomaterial. Repurposing SMS into mycelium-based composites as effective acoustic biomaterials provides a sustainable building material, addresses waste management challenges, and supports the mushroom industry—delivering environmental, economic, and practical benefits in a single approach.

Invited Papers

2:20

1pAAa5. Broadband sound absorption via Algal biofoams and Helmholtz resonator arrays. Mohammad Tabatabaei Manesh (Univ. of Washington, 3950 University Wy NE, Seattle, WA 98105, mhtaba@uw.edu), Hareesh Lyer, Ryan Kim, David Kim, Meghan Dillon, Tomás I. Méndez Echenagucia, and Eleftheria Roumeli (Univ. of Washington, Seattle, WA)

The construction industry is responsible for a large percentage of the global greenhouse gas emissions, material consumption, and waste. The industry is undergoing a process of material replacement to counteract this trend, replacing high carbon and resource intensive materials such as plastics, metals, and concrete, with renewable and biodegradable materials coming from plants. Sound absorptive foams, traditionally derived from plastics, are no exception. This work presents the design, fabrication, and analysis of resonant absorbers with algal foam layers and linings, with the objective of efficient broadband absorption and a reduction of our environmental impact. Different species of algae are lyophilized to create biofoams of different densities and structural properties. The sound absorption coefficient and acoustic impedance of these samples are measured with the impedance tube method, and these properties are then used to create analytical and FEM models of the proposed absorptive panels. These models, in combination with stochastic optimization algorithms, are used to generate the geometry of the Helmholtz cavities that best make use of the biofoam to enhance the viscothermal losses and impedance matching. The study aims to make use of the large design freedom present in Helmholtz resonator geometry to generate designs tailored to the algae.

2:40

1pAAa6. Acoustic properties of microslit resonators fabricated with spent coffee grounds and polylactic acid. Matthew Ripley (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 1129 Sage Ave. Troy, NY 02180, riplem@rpi.edu), Joshua Draper, Dawson Chak, Cooper Myers (CASE Ctr. for Architecture, Sci. and Ecology, Rensselaer Polytechnic Inst., New York, NY), and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

The use of waste material for fabrication of architectural elements is a promising alternative to landfilling. Upcycling organic waste reduces its environmental impact and prevents excess strain on waste management systems. Spent coffee grounds (SCG) can be upcycled as filler in polylactic acid (PLA) based biocomposites, which can be shaped into complex forms such as microslit resonators. By testing the absorptive properties of an SCG-PLA biocomposite sample in an impedance tube, its quality as an acoustic element is determined. This talk details the physical and mechanical properties of SCG-PLA biocomposites, the results of the impedance tube measurements, and possible use case scenarios for architectural acousticians.

3:00–3:20 Break

Contributed Papers

3:20

1pAAa7. Assessment of sound insulation performance in double-leaf walls using Wood-Wool Cement Boards. Rodrigo Scoczynski Ribeiro (COECI-GP and PPGEC-CT, Federal Univ. of Technol. (UTFPR), Avenida Guarapuava 800, Guarapuava, Paraná 85053-525, Brazil, rodrigosribeiro@professores.utfpr.edu.br), Dyorgge Alves Silva (PPGSAU, Federal Univ. of Technol. (UTFPR), Guarapuava, Brazil), and Rodrigo Eduardo Catai (PPGSAU, Federal Univ. of Technol. (UTFPR), Curitiba, Brazil)

Brazilian plantation wood is increasingly being used in the construction industry, expanding beyond its traditional application in furniture. With the global push for sustainable building practices, wood's ability to sequester carbon has made it a valuable material. However, despite its sustainability benefits, wood is an anisotropic material, and its sound insulation properties are challenging to predict. This often leads acousticians, architects, and engineers to specify alternative materials. This study examines the sound insulation performance of double-leaf walls constructed with Wood-Wool Cement Boards (WWCB) and plasterboard, considering both outdoor and indoor applications within a steel framing system. The dynamic and static stiffness moduli, along with density, were determined to predict wall behavior using the INSUL software. Measurements were conducted in a reverberation room following the ISO 10140 series standards. The results suggest that WWCB are effective options for indoor and outdoor walls in terms of sound insulation performance.

3:40

1pAAa8. Sustainability and sound insulation—Impact sound insulation of lightweight floor constructions on base slabs in timber and lightweight solid construction. Normen Langner (Architecture and Civil Eng., Tech. Univ. pf Appl. Sci. Würzburg-Schweinfurt, Röntgenring 8, Faculty of Architecture and Civil Eng., Würzburg 97070, Germany, normen.langner@thws.de) and Alexander Müller (Architecture and Civil Eng., Tech. Univ. pf Appl. Sci. Würzburg-Schweinfurt, Würzburg, Germany)

Timber construction and lightweight solid elements are sustainable options for modern building design. Lightweight floor constructions are essential for both, with slim, weight-reduced cross-sections enabling material savings and adhering to sustainability principles. In Germany, impact sound insulation requirements are defined in DIN 4109-1 (minimum sound insulation) and DIN 4109-5 (enhanced sound insulation). Floating screeds are necessary for compliance, as soft floor coverings (e.g., carpets) are not allowed in assessments according to the standard. Additional filling materials are required to increase the mass of base slabs to meet insulation standards. Reliable calculation methods for timber ceilings and lightweight solid slabs are lacking, leading to dependence on standardized reference constructions or previous designs. However, real-world building conditions often diverge from these references. This study investigates lightweight floor constructions on timber and lightweight solid slabs through literature reviews and practical measurements. Test setups on 1–2 m^2 ceiling sections evaluated various constructions, measuring standardized impact sound levels. Results indicate that sustainable timber and lightweight solid base slab constructions, paired with lightweight floor systems, can achieve high-quality impact sound insulation. While initial designs may follow reference models, ongoing research aims to predict performance deviations accurately, reducing reliance on additional testing.

4:00

1pAAa9. An overview of new testing results on CLT assemblies. Michael Raley (PAC Int., 2000 4th Ave. Canby, OR 97013, mraley@pac-intl.com)

This presentation is an overview of new testing results for 5-ply CLT assemblies using high-performance dry floating floor buildups. The floating floor buildups include low-profile (less than 3-inch) options and options with mass timber pedestals used to provide room for building services in a modular mass timber project. Some of the CLT test results will be compard to test results for a similar floating floor tested on a typical six inch concrete slab. Finally, the presentation will look at using measured vibration deltas for the floating floor to predict its performance when used on a 5-ply CLT.

Invited Papers

4:20

1pAAa10. Working toward net zero light wood frame structures. Aedan Callaghan (Pliteq Inc., 4211 Yonge St., Ste. 400, Ste. 404, Toronto, ON M2P 2A9, Canada, acallaghan@pliteq.com)

In light wood frame construction, balancing acoustic performance while reducing the environmental impact of building materials is a challenge of increasing importance in the effort to decarbonize new construction. Floor-ceiling assemblies often rely on cementitious materials to increase mass, which contribute to higher global warming potential (GWP). Concrete is responsible for 6% of greenhouse gas emissions worldwide. This study focuses on common light wood frame construction types, including open web trusses, and wood I joists. For decades in the USA and Canada, concrete or gypsum concrete toppings ranging from 0.75to 1.5 inches over wood joists and sheathing have been the most widely used approach. This study aims to evaluate alternatives to cementitious toppings that offer reduced embodied carbon. Additionally, incorporating recycled or bio-based materials into acoustic assemblies can increase biogenic carbon sequestration getting closer to a net zero carbon impact. This work utilizes ASTM E90 and E492 testing to quantify airborne and structure borne acoustic performance and environmental product declarations (EPDs) and industry averages to determine the GWP of an acoustic assembly. Several light wood frame floor assemblies are presented and compared, highlighting those that standout as high acoustic performers with below average embodied carbon.

4:40

1pAAa11. Providing acousticians with data on sustainability within their modeling tools. Arthur W. van der Harten (Acoust., Acoust. Distinctions / Open Res. in Acoust. Sci. and Education, 400 Main St. Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com) and Jonathan M. Broyles (Civil, Environ. and Architectural Eng., The Univ. of Colorado Boulder, Broomfield, CO)

Acousticians are so busy advocating for hearing comfort and noise control (noble goals in their own right), in that they often do not realize their own role in mitigating global warming and climate change through minimizing carbon emissions attributed to different building acoustic materials. This paper will demonstrate an updated software package that acousticians can leverage to be mindful of the environmental and health impacts of the materials that they specify everyday, by including data provided in Environmental and Health Product Declarations in the materials library of a popular acoustic simulation tool. The open source tool Pachyderm Acoustic is used as

the simulation tool, and the open-source dataset by Broyles that conveys the environmental impacts and health effects of North American acoustic products is referenced by the software. A prototype interface is presented for discussion. Overall, this paper showcases how acousticians can provide high quality acoustic solutions in building design while being cognizant of the environmental impacts and health effects of the materials specified.

MONDAY AFTERNOON, 19 MAY 2025 GALERIE 3, 12:55 P.M. TO 5:40 P.M.

Session 1pAAb

Architectural Acoustics, Noise and ASA Committee on Standards: Day of ASHRAE Part I—Sound Standards and Codes

Karl Peterman, Cochair
Johnson Controls, 3 Keensford Court, Unit 1, Ajax L1Z 0K4, Canada

David Manley, Cochair
DLR Group, 6457 Frances St., Omaha, NE 68106

Paul F. Bauch, Cochair
100 JCI Way, York, PA 17406

Steve Wortman, Cochair
Engineering, Victaulic Company, 4901 Kesslersville Rd, Easton, PA 18040

Chair's Introduction—12:55

Invited Paper

1:00

1pAAb1. An overview of test standards for HVAC components in North America. Karl Peterman (3 Keensford Court, Unit 1, Ajax, ON L1Z 0K4, Canada, kpeterman@vibro-acoustics.com)

With few exceptions, the acoustical properties of HVAC components in North America are determined by standards authored by ASHRAE and AHRI. There has been a long-standing arrangement between the American Society of Heating, Refrigerating, and Air-conditioning Engineers and the Air-conditioning Heating and Refrigeration Institute that helps maintain boundaries of standards scopes and ensure harmonization of their related documents. Both ASHRAE and AHRI are ANSI accredited standards development organizations that comply with ANSI's Essential Requirements. ASHRAE has primary authority to determine methods of tests for HVAC devices and components and AHRI sets the conditions under which those devices are rated and regulates the associated published information. Where no ASHRAE method of test exists, AHRI may choose to create their own. This presentation will give a summary overview of the various test methods used in the industry to report acoustical performance properties.

Contributed Paper

1:20

1pAAb2. A comparison of commonly used water-cooled chiller sound metrics. R. Troy Taylor (Sound and Vib., Johnson Controls, 5000 Renaissance Dr., New Freedom, PA 17349, r.troy.taylor@jci.com), Patrick C. Marks (Sound and Vib., Johnson Controls, New Freedom, PA), and Paul F. Bauch (Sound and Vib., Johnson Controls, York, PA)

Noise from centrifugal chillers can often be a key driver in the selection process for large equipment. This paper presents the history of sound standards applicable for water-cooled centrifugal compressor chiller sound measurements, specifically AHRI Standard 575 and AHRI Standard 1280. Presented is an overview of the beginnings of AHRI 575 and how it became the de facto method for rating water-cooled centrifugal chillers, though it was never intended for that purpose. Discussion on the creation of AHRI 1280, how it improves on AHRI 575, and, finally, the benefits of using it as the measurement standard for rating water-cooled centrifugal chillers are provided.

Invited Papers

1:40

1pAAb3. Optimized discrete frequency qualification of reverberation rooms for AHRI Standard sound power testing of HVAC equipment. Sudharsan Subramanian (Sound and Vib., Johnson Controls, 5000 Renaissance Dr., York, PA 17349, sudharsan.subramanian@jci.com), Paul F. Bauch, and Roger L. Howard (Sound and Vib., Johnson Controls, York, PA)

Evaluation of sound power levels of HVAC equipment using AHRI Standards may require the use of a qualified reverberation room to maintain a diffuse field, which is crucial for reproducibility and accuracy. AHRI Standard 220 prescribes the process to qualify a reverberation room, which involves qualifying the room at both broadband and discrete frequencies. Discrete frequency qualification is particularly tedious and time-consuming, requiring sequential sound measurements from a speaker emitting discrete frequency tones within each 1/3 octave band. This paper investigates the use of a random noise source (white/pink noise) as an alternative, analyzing the broadband FFT to extract the discrete frequency data required for qualification as per AHRI Standard 220. Additionally, results obtained from this alternative approach are compared with results obtained from traditional discrete frequency methodology used in AHRI Standard 220. This comparison is used to discuss faster alternatives when qualifying a reverberation room for tones.

2:00

1pAAb4. The impact of tones on NC ratings. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Historically, the NC rating of steady-state background noise is determined by comparing the measured octave band sound pressure levels with the NC curves, where the rating is determined by the lowest curve such that no measured octave band level exceeds the NC curve. The 2023 version of the ASHRAE Applications Handbook presents the NC curves in Figure 5 of Chapter 49, and the supporting text states: "In HVAC systems that do not produce excessive low-frequency noise and strong discernable pure tones, the NC rating correlates relatively well with occupant satisfaction if sound quality is not a significant concern." At present, NC curves are defined only in 5-dB increments and only in octave bands, but an ANSI S12 working group is currently working on a revision of ANSI S12.2 that may also define the NC curves in 1-dB increments and in one-third octave bands. ASHRAE Research Project 1707 studied the annoyance of steady-state tones in background noise dominated by HVAC systems applicable to an office environment. One of the results of this research project is a software program that predicts the annoyance of HVAC background noise that may or may not contain tones. This presentation examines several steady-state background noise signals that contain steady-state tones of varying frequency and tone to noise ratio to illustrate how tones in the spectrum can impact the NC rating and the overall annoyance of the background noise containing tones.

2:20

1pAAb5. Progress update on ANSI/ASA S12.2, criteria for evaluating room noise. Brandon Cudequest (Threshold Acoust., Chicago, IL) and Derrick P. Knight (Trane Technologies, 2313 20th St. South, La Crosse, WI 54601, Derrick.Knight@TraneTechnologies.com)

A working group is evaluating the American National Standard, "Criteria for Evaluating Room Noise" (ANSI/ASA S12.2-2019). This presentation will give an overview of our progress and areas of significant proposed change. The goal is to simplify the standard body to match the industry practice of using Noise Criterion curves and A-weighted sound levels as the primary noise level metrics. Less common criteria will be relegated to annexes, with explanations for when it may be appropriate to use these alternate rating systems. Since the beginning of this standard, there has been interest in qualifying the effects of spectral imbalance, temporal variation, and tone prominence. This talk will briefly mention proposals to address these aspects. A procedure for using Noise Criterion curves in one-third octave bands will also be explained. The goal is to solicit feedback from the industry to minimize the challenges associated with implementing this standard.

Contributed Paper

2:40

1pAAb6. A case study on the ANSI/ASA S12.2 progress update. Logan Pippitt (Talaske, 32560 W 171st St., Gardner, KS 66030, lpippitt@talaske.com)

A working group is evaluating the American National Standard, "Criteria for Evaluating Room Noise" (ANSI/ASA S12.2-2019). This presentation will compare the existing standard to the current draft through the lens of a recently constructed performing arts project. The comparison will highlight elements of the standard to remain and areas of significant proposed change. The talk will walk through the use of the simplified standard body, criteria relegated in the annexes, and proposed tools to address the interest in qualifying the effects of spectral imbalance, temporal variation, and tone prominence. The proposed procedure for using Noise Criterion curves in one-third octave bands will also be demonstrated. The goal is to demonstrate a working interpretation of the updated standard and solicit feedback from the industry to minimize the challenges associated with the future implementation of this standard.

Invited Papers

3:00

1pAAb7. A historical review of recommended noise levels according to room type. Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St. Omaha, NE, Omaha, NE 68182-0816, samuelunderwood@unomaha.edu)

Many authors have developed tables of recommended noise levels for different room types, which can be used to develop performance specifications in buildings. As part of an ongoing effort to update ASA/ANSI S12.2-2019, over 60 references have been surveyed to develop a compiled table of recommended noise levels according to room type. The compilation includes a range of books, design guides, standards, performance certification programs, building codes, conference presentations, and research papers that have been published over the past 80 years. While work is still ongoing to curate a refined selection of criteria for the annex of ASA/ANSI S12.2, the full historical survey of recommended noise levels has been made available for general use. Results from this review illustrate the prevalence of empirical design criteria which can be traced back to earlier historical references. Discrepancies in recommended criteria arising from inconsistent conversion methods between metrics are identified, and implications on design practice are discussed.

3:20–3:40 Break

3:40

1pAAb8. AHRI-260 sound power data for ducted VRF fan coil units. Jeffrey Watt (Daikin Appl., 13600 Industrial Park Blvd, Plymouth, MN 55441, jeffrey.watt@daikinapplied.com)

This presentation will highlight the steps that have been taken to qualify a reverberation room to accurately measure inlet, discharge, and casing radiated sound power in accordance with AHRI-260. The setups for measuring inlet, discharge, and casing radiated sound power will be presented along with sample test results for each. Many VRF manufacturers are currently publishing sound pressure level data for ducted fan coil units measured at only one location in an anechoic chamber. However, data collected at one location in an anechoic chamber cannot be used to accurately predict sound levels in a real room. This presentation will show a comparison of the two methods and explain why AHRI-260 measured data are preferred.

Contributed Paper

4:00

1pAAb9. NEW DEGA-regulation 103-1 Sound protection in residential buildings. Christian Nocke (Akustikbuero Oldenburg, Sophienstr. 7, Oldenburg, Nds. 26121, Germany, nocke@akustikbuero-oldenburg.de)

In September 2024, the final version of DEGA Regulation 103-1 “Sound protection in residential buildings—part 1: sound protection classes and higher sound protection” (DEGA-Richtlinie 103-1 “Schallschutz im Wohnungsbai—Teil 1: Schallschutzklassen und erhöhter Schallschutz”) has finally been published. This was a milestone after a lengthy discussion within the Building and Room Acoustics Technical Committee of DEGA and beyond. The concept of sound protection classes was firstly introduced in the DEGA recommendation 103 (“DEGA-Empfehlung 103”) of the year 2009. One revision keeping this idea was published in 2018. At an early stage in the present revison, the combination of guideline VDI 4100 with three sound protection levels (“Schallschutzstufen”) and DEGA 103 was extensivy discussion but finally abandoned. The following changes have been introduced in the new DEGA 103-1:—Introduction and implementation of a parallel (“two-track”) concept with “space-related” and “component-related” parameters for the classification of sound protection—introduction of optional requirements—extension and adaptation of the recommendations to the own living area from the DEGA-Memorandum BR 0104 from February 2011.

Invited Paper

4:20

1pAAb10. Standardization of building assessments: Benchmarking acoustic performance. Viken Koukounian (Parklane Mech. Acoust., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@parklanemechanical.com)

The evolution of acoustic standards and codes in the built environment underscores the growing need for objective, repeatable benchmarking practices. By prioritizing occupant needs, it becomes possible to develop a framework that identifies key categories influencing acoustical satisfaction—such as acoustic privacy, acoustical comfort, and communication—while defining methodologies to evaluate the acoustic parameters affecting these experiences. This session highlights the ongoing work of the ASHRAE Guideline Project Committee (GPC) 45P, “Measurement of Whole Building Performance for Occupied Buildings except Low-Rise Residential Buildings,” emphasizing the importance of “Acoustical Quality” among Indoor Environmental Quality (IEQ) parameters and presenting a holistic approach to benchmarking building performance. Attendees will leave with practical tools and strategies to tackle unique acoustical challenges, enabling them to improve building performance while supporting occupant health and well-being.

Contributed Paper

4:40

1pAAb11. National standards on classroom acoustics: Key descriptors and global perspectives. Virginia Tardini (Dept. of Industrial Eng., Univ. of Bologna, Viale Risorgimento 2, Bologna 40136, Italy, virginia.tardini2@unibo.it), Giulia Fratoni, and Dario D'Orazio (Dept. of Industrial Eng., Univ. of Bologna, Bologna, Italy)

Adequate classroom acoustics enhance communication, support student concentration, and reduce teachers' vocal effort. Many countries follow standards aligned with international guidelines to meet these needs. The study is the second (and last) part of a review of National Standards, incorporating insights from over 100 experts. Despite challenges such as limited data and complex interpretations, the research synthesizes and standardizes an extensive number of datasets coming from 52 countries. Results are summarized in terms of (1) taxonomy of compliance, (2) key components for learning environments, (3) requirements for ancillary spaces, and (4) interactions between acoustic requirements, occupancy, and geometry of the rooms. The findings try to find common challenges and provide actionable recommendations for improving classroom acoustics globally and locally by two key components, namely, the sound absorption of the room and the control of speech intelligibility. The key descriptors for learning spaces must then be placed in a context of harmonization with ancillary spaces. The work aims to investigate how these requirements are effective in the operational conditions (active and occupied) of the environment. By addressing these issues, the research contributes to advancing acoustic quality in schools, fostering better learning and teaching experiences across diverse educational environments.

Invited Papers

5:00

1pAAb12. Revision of standard S12.60 Part 1. Stephen J. Lind (LindAcoustics LLC, 1108 Valley Vue Dr., Onalaska, WI 54650, stephen.j.lind.ut88@gmail.com)

The classroom acoustics standard ASA/ANSI S12.60 Part 1 is currently being revised. The document originally published in 2001 was updated in 2010 in an attempt to make the document easier to reference in building codes. It moved explanatory sections to informative annexes and revised some technical requirements to make it align with expectations of code officials. This was only partially successful and imposed some challenges. The current revision intends to bring the standard in closer alignment with current practice and updates some of the technical requirements included in the standard to be more technically accurate and implementable. For example, the previous reverberation time requirements limits are proposed to be revised to allow calculation of average absorption coefficients. The use of either calculations of average absorption or measurements of reverberation time would be allowed to show conformance. This change allows the limits based on room size to avoid discontinuities as room size changes as occurred in the previous method. The revision also proposes changes to the prescriptive sound isolation requirements used in the previous version. Both outdoor to indoor transmission and indoor sound transmission classes are considered, and methods of conformance by test are expanded and clarified.

5:20

1pAAb13. Case studies of recent experiences with the International Building Code Requirements for Background Sound Levels in Educational Spaces. Jennifer R. Miller (Siebein Assoc., Inc., 625 NW 60TH St., Gainesville, FL 32607, jmiller@siebeinacoustic.com), Matthew Vetterick, Keely M. Siebein, Gary Siebein Jr., Abigail Gulley, Nicolas Ospina, and Gary W. Siebein (Siebein Assoc., Inc., Gainesville, FL)

For K-12 Education spaces, requirements for maximum background sound levels of 35 dBA and 55 dBC for sound sources inside and outside the classroom, to be evaluated separately, and maximum reverberation times were included in Section 808 A117.1 of the International Building Code in 2021. These requirements were adopted in Section 1211 of the Florida Building Code in 2023. Major school districts in Florida are having to adapt their design standards for new and renovated schools to meet these requirements. Acoustical measurements of background sound levels were made of HVAC system noise levels in selected rooms in three existing schools to determine if the classrooms as currently designed meet the sound level limits in the code. The measured sound levels are compared to the required sound levels. Some acoustical mitigation elements will likely be required in many classrooms to achieve the 35 dBA and 55 dBC sound level limits. The code calls for sound levels to be measured during the greatest 1 h period of time. While this may be typical for exterior sources of sound, it is not generally typical for HVAC noise measurements to be taken over multiple hours, presenting practical issues when surveying a large number of rooms.

Session 1pAB

Animal Bioacoustics: Progress on Bioacoustics of Fish II

Kelly S. Boyle, Cochair
Biological Sciences, University of New Orleans, Department of Biological Sciences, 2000 Lakeshore Drive, New Orleans, LA 70148

John S. Allen, Cochair
Mechanical Engineering, University of Hawaii Manoa, Holmes 302, 2540 Dole Street, Honolulu, HI 96822

Chair's Introduction—1:35

Invited Papers

1:40

1pAB1. Fish Acoustic Detection Algorithm Research (FADAR): A powerful tool to explore the role of call types in fish spawning behavior and trend in spawning population structure. Laurent M. Cherubin (Harbor Branch Oceanographic Inst., Florida Atlantic Univ., 5600 N US Hwy. 1, Fort Pierce, FL 34946, lcherubin@fau.edu), Ali Ibrahim (Elec. Eng. and Comput. Sci., Florida Atlantic Univ., Boca Raton, FL), Michelle Schärer-Umpierre (HJR Reefscaping, Boquerón,), and Caroline Woodward (Harbor Branch Oceanographic Inst., Florida Atlantic Univ., Fort Pierce, FL)

FADAR is a user-friendly deep learning algorithm specifically designed to identify and classify the call types of four Caribbean grouper species. FADAR is capable of processing ten thousand 20-s audio files in less than 2.5 h with an average accuracy of 90% for all four species. Therefore, long term recordings can be analyzed in just a few weeks and provide insight into the ecology of grouper spawning aggregations. FADAR was used to study the reproductive cycle of red hind (*Epinephelus guttatus*) over a 12-year period at a spawning aggregation site located off the west coast of Puerto Rico, in the Greater Antilles. Red hind produce four distinct call types that are used in combination during spawning aggregations, but mostly two are used in spawning associated courtship. Yearly times series confirmed in captivity observations of call type relative timing in courtship leading to spawning. The interannual variability observed in the relative variation of call type numbers suggests however, a shift in the male to female ratio toward a male dominated population, which could affect reproductive success.

2:00

1pAB2. Using power spectral band sums to identify significant segments in long-term recordings. Mark W. Sprague (Phys., East Carolina Univ., M.S. 563, Greenville, NC 27858, spraguem@ecu.edu) and Joseph J. Luczkovich (Biology, East Carolina Univ., Greenville, NC)

Long-term passive acoustic recorders (LTPARs) are useful for studying changes in underwater soundscapes by capturing variation in sounds produced by fish mating choruses, marine mammals, and invertebrates (biophony); human activity (anthrophony); and the physical environment (geophony). With advances in storage, LTPARs can record for weeks, months, or longer, capturing information about changes in soundscapes on multiple scales. However, researchers cannot listen to and analyze these recordings in reasonable timeframes. Therefore, we use detection algorithms to obtain reduced sets of recordings for detailed analysis. Here we describe the Power Spectral Band Sum (PSBS) used to identify segments of interest in a recording where target sounds, such as fish choruses, are present or where target sounds change in loudness. The PSBS adds the power spectrum components for all frequencies in a band characteristic to a sound of interest. These quantities are computed without difficulty for long recordings. We give examples of PSBSs of long-term recordings and present the "unknown buzz" recorded off NC, leading to understanding the presence of Atlantic midshipmen (*Porichthys plectrodon*) populations off NC and FL, and perhaps a related species of midshipmen in the Indian Ocean. The PSBS could help identify and locate these and other never-recorded fish choruses.

2:20

1pAB3. High-throughput information processing in fisheries and plankton acoustics. Wu-Jung Lee (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Valentina Staneva, Caesar Tuguinay, Soham Butala, and Brandyn Lucca (Univ. of Washington, Seattle, WA)

Active acoustic data collected by echosounding instruments are widely used to infer the spatiotemporal distributions and behaviors of fish and zooplankton in the ocean. The broad deployment of these instruments on ocean observing platforms has led to a rapid increase in water column sonar data over the past decade. These data hold great potential to link oceanographic and climate drivers to ecosystem responses at the mid- to high-trophic levels that directly support global food systems. In this presentation, we highlight our group′s efforts to accelerate information processing of active acoustic data by tackling the challenges of analysis scalability, consistency,

reproducibility, and interoperability. Taking a multifaceted approach, we develop data-driven methods to extract echo information on commercially targeted species and ecologically relevant spatiotemporal patterns, build open-source software and workflows to leverage the ever-growing local and cloud computing resources, collaborate closely with survey missions to support operational priorities, and create and host open educational resources and workshops to support the broader community. We believe these efforts are essential for achieving high-throughput active acoustic information processing and enabling community-wide collaboration, which are key to distill both scientific knowledge and actionable insights to help us better understand and adapt to rapidly changing marine ecosystems.

2:40

1pAB4. Automatic detection of fish sounds: A comparison of traditional machine learning with deep learning. Xavier Mouy (School of Earth and Ocean Sci., Univ. of Victoria, 266 Woods Hole Rd., Woods Hole, MA 02543, xavier.mouy@whoi.edu), Stephanie Archer (Louisiana Universities Marine Consortium, Chauvin, LA), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Sarah Dudas, Philina English (Pacific Biological Station, Fisheries and Oceans Canada, Nanaimo, BC, Canada), Colin Foord (Coral Morphologic, Miami, FL), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), Francis Juanes (Uvic, Victoria, BC, Canada), Darienne Lancaster (Biology Dept., Univ. of Victoria, Victoria, BC, Canada), Sofie Van Parijs (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA), and Dana Haggarty (Pacific Biological Station, Fisheries and Oceans Canada, Nanaimo, BC, Canada)

Many species of fish produce sounds that can be used to monitor them non-intrusively and could complement traditional monitoring techniques. However, the manual annotation of fish sounds in acoustic recordings remains time-intensive, limiting the use of passive acoustics as a viable monitoring tool. This study compares two automated approaches for detecting fish sounds: Random Forest (RF) and Convolutional Neural Networks (CNN). Both algorithms were trained on 21,950 manually labeled fish and non-fish sounds recorded between 2014 and 2019 in the Strait of Georgia, British Columbia, Canada. Performance calculated on data from the Strait of Georgia, Barkley Sound, and the Port of Miami showed that the CNN performed up to 1.9 times better than the RF (F-score: 0.82 versus 0.43) and was in some cases able to find more faint fish sounds than the analyst. Noise analysis in the 20–1000 Hz frequency band shows that the CNN is still reliable in noise levels greater than 130 dB re 1 μPa in the Port of Miami but becomes less reliable in Barkley Sound past 100 dB re 1 μPa due to mooring noise. We show that the proposed approach can make passive acoustics viable for monitoring fish in a variety of environments.

3:00–3:20 Break

Contributed Papers

3:20

1pAB5. Impact of anthropogenic sounds on coral reef health and sessile invertebrates. Madelyn S. Rangel, Stephanie Bell (University of Hawai'i at Mānoa, Inst. of Marine Biology, Honolulu, HI), Aude Pacini (University of Hawai'i at Mānoa, Inst. of Marine Biology, Kaneohe, HI), Zac Forsman, Ingrid Knapp (University of Hawai'i at Mānoa, Inst. of Marine Biology, Honolulu, HI), John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Holmes 302, 2540 Dole St., Honolulu, HI 96822, alleniii@hawaii.edu), and Robert Toonen (University of Hawai'i at Mānoa, Inst. of Marine Biology, Honolulu, HI)

The impact of anthropogenic sound on marine organisms is an emerging area of concern given the increasing levels human based oceanic activity. Effects of noise on marine mammals and to a lesser extent fish have been the focus of previous studies; however, the impacts on benthic invertebrates are less understood. Altered fish and other taxa behavior due to anthropogenic noise may disturb benthic coral reef organism health and larval settlement may be disrupted. We investigated the responses of a reef-building, scleractinian coral (*Montipora capitata*), a soft coral (*Palythoa mutuki*), and a tube worm (*Sabellastarte spectabilis*) to specific anthropogenic sounds. Coral fragments groups of *M. Capitate* were exposed individual and also combination sounds (1, 10, and 50 kHz) over 3-to 5-min intervals over a month long period. *P. mutuki* and *S. spectabilis* were exposed to 15, 50, 75, and 100 kHz frequency sounds with respect to their tentacle retraction response. Quantification with respect to growth and mortality has indicated limited impact, and these results may be interpreted in terms of the diurnal ambient noise background levels.

3:40

1pAB6. Unsupervised clustering of biological sounds in a Hawaiian coral reef. Daniel Duane (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, daniel.m.duane.civ@us.navy.mil), Nicholas Kroeger (Univ. of Florida, Gainesville, FL), Simon Freeman (Dept. of Energy, ARPA-E, Washington, DC), and Lauren Freeman (Naval Undersea Warfare Ctr., Newport, RI)

Manually labeling datasets for supervised machine learning are often time-consuming, expensive, and subjective, particularly in the ocean acoustic domain where a typical human labeler may not be familiar with the signals of interest. Unsupervised learning can be used to automatically cluster signals in large acoustic datasets with no manual labels, which can greatly expedite tasks related to detection, classification, and soundscape characterization. Here, a convolutional autoencoder was built for the purposes of automatically clustering low frequency (>700 Hz) biological sounds in a Hawaiian coral reef. More than 500,000 detections were clustered into five classes—including humpback whale calls, damselfish courtship displays, parrotfish feeding sounds, and two unidentified fish vocalizations. Unique diel, lunar, and seasonal trends are observed for all five classes, demonstrating the potential for unsupervised algorithms to illuminate the environmental factors influencing biological soundscapes.

4:00

1pAB7. An open-source workflow for organizing fisheries acoustics data from transect surveys for machine learning applications. Caesar Tuguinay (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ctuguina@uw.edu), Wu-Jung Lee, and Valentina Staneva (Univ. of Washington, Seattle, WA)

As the volume of active water column sonar data expands, automated, machine learning (ML)-based echogram analysis methods are increasingly used to detect biological scatterers at scale. Semantic segmentation, which labels each pixel in an image, is a class of methods that is increasingly adopted to detect and classify acoustic targets. However, survey echograms are not standard images: they have spatiotemporal associations and originate

from data collection strategies targeting potentially patchy biological aggregations. In the context of transect surveys, we consider four key requirements to create echogram datasets for ML applications: (1) how to partition data based on survey information, (2) how to subsample spatially via transect-based groupings to minimize model overfit, (3) how to create region masks of biological scatterers to exclude noise and other contaminating sources (e.g., seafloor), and (4) how to reconcile different spatiotemporal resolutions of echo data and human annotations. We present a generalizable workflow for constructing echogram datasets that addresses these requirements, and discuss our implementation using two open-source tools, Echopype and Echoregions. We highlight how the workflow enables a flow of echo data with ancillary spatiotemporal information propagating downstream, and demonstrate its scalability in organizing a multi-year dataset from a fisheries acoustic-trawl survey.

4:20–4:40 Panel Discussion

MONDAY AFTERNOON, 19 MAY 2025 STUDIOS 7/8, 12:55 P.M. TO 3:00 P.M.

Session 1pAOa

Acoustical Oceanography and Underwater Acoustics: Acoustical Oceanography at Deep Water Abrupt Topography I

John A. Colosi, Cochair
Oceanography, Naval Postgraduate School, Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943

Ying-Tsong Lin, Cochair
Woods Hole Oceanographic Inst., Scripps Institution of Oceanography, La Jolla, CA 92093

Lauren Freeman, Cochair
NUWC Newport, NUWC, Naval Undersea Warfare Ctr, 1176 Howell St, Newport, RI 02841

Chair's Introduction—12:55

Invited Paper

1:00

1pAOa1. Modeled physical oceanography of the New England Sea mounts during Summer 2024. John Osborne (U.S. Naval Res. Lab., 1009 Balch Blvd, Stennis Space Ctr, MS 39529, john.j.osborne11.civ@us.navy.mil)

Physical oceanography processes affecting underwater sound speed and acoustic propagation near the New England Seamounts during July and August 2024 are discussed. These processes impact underwater sound propagation through the generation, modification, and destruction of surface ducts and secondary sound channels, as well as modify the deep sound channel and convergence zone horizontal length scale. An additional factor is the presence of Gulf Stream, which serves as a boundary between cold Scotian Shelf waters to the north and warm Sargasso Sea water to the south. These water masses have different sound speed properties, so the variability of the Gulf Stream, thus these water masses, impacts sound speed. Results from real-time data assimilating models are analyzed, verified against *in situ* temperature and salinity vertical profiles from gliders, ships, and other sensors from a large field program. Model results supported field work in multiple ways: Assisting in daily planning of ocean-acoustic adaptive sampling missions; real-time acoustic modeling; and skillful prediction of cold water at 50–100 m depth originating several hundred kilometers away. These results demonstrate a capability for data-informed real-time ocean models to skillfully predict transient physical oceanographic features and their sound speed properties. [Work supported by Office of Naval Research.]

Contributed Papers

1:20

1pAOa2. Using an autonomous surface vehicle to survey low-frequency acoustic propagation near Atlantis II seamounts. Matthew McKinley, Davis Rider (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Ganesh Gopalakrishnan, Laurent Grare, Luc Lenain (Scripps Inst. of Oceanogr., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Deepwater abrupt topography changes, such as seamounts, can significantly enhance the complexity of underwater sound propagation by notably creating highly variable three-dimensional scattering effects, thus making it more challenging to predict numerically. One avenue to precisely survey the spatial variability of deep-water sound propagation induced by isolated seamounts is to use instrumented autonomous surface vehicles (ASV) that can be accurately geo-located. These precise acoustic observations can, in turn, be used to validate numerical model predictions in these complex environments. Here, an ASV called Wave Glider was equipped with a hydrodynamic towed acoustic module (TAM) to survey the spatial variability of low-frequency acoustic propagation across the Atlantis II seamounts in the Northwest Atlantic. The TAM was deployed along the Gulf Stream boundary and crossed over the Atlantis II seamounts, which significantly influenced the TAM's recordings of chirp transmissions (500–600 Hz band) from a bottom-moored source ~30 km from the seamounts by notably causing blockage of in-plane propagation paths and complex reverberation arrivals displaying three-dimensional effects, as confirmed by synthetic aperture beamforming. 2-D and 3-D ray-tracing simulations are performed with input sound speed fields computed from the outputs of a data-assimilated ocean model to compare with experimental observations. [Work Supported by ONR.]

1:40

1pAOa3. Subsurface acoustic ducts near the Gulf Stream. Alice S. Ren (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #21, Woods Hole, MA 02543, Alice.ren@whoi.edu) and Robert E. Todd (Woods Hole Oceanographic Inst., Woods Hole, MA)

Oceanic fronts are regions of rapid change in seawater properties. We examine a collection of approaching 10,000 oceanic sound speed profiles from the surface to 1000 m measured by autonomous underwater gliders within 200 km of the Gulf Stream front between 35 and 41 °N in the North Atlantic Ocean. The underwater gliders sample with approximately 5 km resolution in the cross-front direction and allow us to study subsurface acoustic ducts across the front. We algorithmically detect ducts in the sound speed profiles and examine statistics of subsurface duct axial sound speed, width, and cutoff frequency depending on depth and distance from the Gulf Stream. We associate the formation of subsurface sound speed ducts with ocean water mass properties and dynamics. In particular, we examine the case of ducts formed by cool, fresh water from the continental shelf that is exported, entrained, and subducted beneath the Gulf Stream.

2:00

1pAOa4. Mapping hydrothermal discharge with multibeam echosounders: Insights from numerical simulations and laboratory experiments. Guangyu Xu (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, guangyux@uw.edu), Elizabeth Weidner (Marine Sci., Univ. of Connecticut, Groton, CT), Aaron Marburg (Appl. Phys., Univ. of Washington, Seattle, WA), Karen Bemis (Rutgers Univ., NB, NJ), and Darrell Jackson (Appl. Phys., Univ. of Washington, Seattle, WA)

Seafloor hydrothermal vents, primarily found atop mid-ocean ridges and hotspot submarine volcanoes, act as key conduits for transferring heat and chemicals from the Earth's interior to the ocean. Active acoustic techniques, such as multibeam echosounder, are increasingly employed to detect buoyant plumes from these vents through backscatter imaging of the water column over the abrupt seafloor topography typical of hydrothermal vent fields. Here, we present an alternative method for mapping seafloor hydrothermal discharge based on the loss of coherence (decorrelation) in seafloor backscatter between transmissions separated by short time intervals. Originally implemented on a stationary platform, this method has been adapted for use on underwater vehicles during seafloor surveys. We evaluate the effectiveness of the method using synthetic data from model simulations and sonar data from a tank experiment. In this presentation, we share the results from these numerical and laboratory experiments and discuss insights for implementing the method in future field operations.

2:20

1pAOa5. Influence of anisotropic smoothing of the input sound speed fields computed from submesoscale permitting ocean models on the resulting stability of simulated eigenrays. Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., 771 Ferst Dr NW, office 131, Atlanta, GA 30332, rtouret@gatech.edu) and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Time-varying range-dependent sound speed profiles (SSPs) computed from high-resolution submesoscale-permitting ocean models can exhibit strong lateral and vertical gradients which are numerically challenging for standard ray-tracing methods, especially in complex environments with abrupt topography. These strong SSPs gradients affect the stability of the predicted eigenrays, e.g., yielding spurious ray arrivals intermittently. Here an anisotropic smoothing kernel, with tailored kernel sizes in the vertical and horizontal dimensions, is applied to the simulated SSPs to enhance the stability of predicted eigenray from standard ray-tracing while retaining the underlying dynamics of the original SSPs. The arrival-times predicted from these smoothed ray tracing simulations—for frequency <1 kHz—match those obtained from parabolic equation (PE) simulations used as benchmark. Furthermore, the predicted arrival-times from PE simulations are found to remain unaffected whether the anisotropic smoothing procedure is applied to the input high-resolution SSPs, as expected since frequency-dependent diffraction effects captured by PE inherently apply a smoothing effect to the SSP gradients. Hence using this smoothing procedure allows to retain the computational efficiency of ray tracing in complex environments while ensuring accurate arrival-time predictions comparable to results from more stable but significantly more computationally intensive PE simulations, [Work supported by ONR.]

2:40

1pAOa6. Measurements of target strength of Gulf Stream mesopelagic fishes using autonomous underwater vehicles. Jennifer J. Johnson (AOPE, Woods Hole Oceanographic Inst., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, jjjohnson@whoi.edu), Andone C. Lavery, Zhaozhong Zhuang, Miad Mursaline, and Robert Petitt (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

Mesopelagic fishes are difficult to study due to their remote locations, patchiness, and net avoidance, but are effectively observed using acoustics. Using shipboard acoustics has limitations when discerning individuals at depth; therefore, critical acoustic information such as Target Strength (TS) of individuals is largely absent. Autonomous platforms allow for sampling in the mesopelagic at resolutions unobtainable via shipboard systems. We present data that were collected in western North Atlantic waters apropos of the New England Seamount chain, specifically in the Gulf Stream (GS) and outside the GS, i.e., Slope and Sargasso Sea, using broadband echosounders on an autonomous underwater vehicle (AUV). Comparisons of TS distributions of individual targets above and in the Deep Scattering Layer (DSL) as well as TS distributions during diel vertical migration (DVM) periods were evaluated. Scattering models indicate resonance peaks of gas bearing fishes at frequencies spanning ~10–40 kHz are depth dependent, which impact in-situ TS measurements and should be considered for the interpretation of observed mesopelagic fish communities using shipboard systems.

Session 1pAOb

Acoustical Oceanography: Topics in Acoustical Oceanography II

Shima Abadi, Cochair
University of Washington, 185 Stevens Way, Paul Allen Center – Room AE100R, Seattle, WA 98195

Kaustubha Raghukumar, Cochair
Integral Consulting Inc., 200 Washington Street, Suite 201, Santa Cruz, CA 95060

Contributed Papers

3:20

1pAOb1. Comparing Kauai Beacon receptions to simulated acoustic propagation. John Ragland (Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu), Shima Abadi (Univ. of Washington, Seattle, WA), Nicholas C. Durofchalk (Naval Post Graduate School, Monterey, CA), David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA), and Kay L. Gemba (Naval Post Graduate School, Monterey, CA)

The Kauai Beacon is an ocean acoustic tomography source off the coast of Kauai that began regularly transmitting in March 2023. In this presentation, positive acoustic receptions of the Kauai Beacon are measured by the Ocean Observatories Initiative (OOI) hydrophones and compared to acoustic arrivals simulated with the parabolic equation method for the first 20 months of regular transmissions. Positive receptions of the Kauai Beacon are reported for eight of the eleven OOI hydrophones. The structure of the observed acoustic arrivals over the first 20 months is compared to simulated acoustic arrivals, and the arrival envelope statistics are compared to simulation. The Kauai Beacon stands to serve as a source of opportunity for any passive-acoustic monitoring infrastructure within its acoustic paths to measure ocean basin acoustic propagation and infer oceanographic variables such as water temperature. This presentation provides the necessary context to develop future methods to acoustically infer ocean temperature with single hydrophone receptions of the Kauai Beacon. [Work supported by ONR.]

3:40

1pAOb2. Detection of the Kauai Beacon signal on Ocean Networks Canada's hydrophones in the NE Pacific. Lanfranco Muzi (Ocean Networks Canada, Univ. of Victoria, 2474 Arbutus Rd., Ste. 100, Victoria, BC V8N 1V8, Canada, muzi@oceannetworks.ca), David R. Barclay, and David Huges (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

We present a study of the Kauai Beacon signal as received by the hydrophones of Ocean Networks Canada's (a University of Victoria Initiative) NorthEast Pacific Time-series Undersea Networked Experiments (NEPTUNE) observatory. Long range acoustic-propagation studies are used to produce basin-scale estimates of the variability of bulk ocean temperature and transmission loss. The Kauai Beacon source, located off of the north shore of the Hawaiian Island of Kauai, was designed to support such studies and has resumed its transmissions on a regular schedule since 2023. At a distance of approximately 4100 km from the Kauai Beacon and a depth close to the deep sound channel axis, the NEPTUNE stations around the Barkley Canyon have an unobstructed path to the source. Though Ocean Networks Canada has had a four-element volumetric array of Ocean Sonics icListen HF hydrophones at the "Barkley Node" site since 2021, a single low-frequency icListen AF hydrophone was deployed in the summer of 2024 at the "Barkley Upper Slope" site, specifically for the purpose of supporting the reception of the Kauai Beacon signal.

4:00

1pAOb3. Comparison and combination of modal-dispersion and matched-field inversion for seabed geoacoustic profiles at the New England Mud Patch. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, School of Earth and Ocean Sci., University of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), David P. Knobles (The Platt Inst., Austin, TX), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

This paper considers the information content for seabed geoacoustic inversion of recorded acoustic waveforms processed as modal-dispersion (MD) data (mode arrival times as a function of frequency) and as matched-field (MF) data (multi-frequency complex acoustic fields across a sensor array). These approaches are applied separately and combined in joint inversion, with the MD and MF datasets derived from the same acoustic recordings collected on the New England Mud Patch. At-sea measurements are simpler for MD data, which can be extracted from recordings at a single uncalibrated sensor while MF data require a synchronized, calibrated multi-sensor array. Further, MF inversion requires estimating the source/receiver depths and complex source spectrum. However, MF inversion is sensitive to seabed attenuation and is more easily extended to higher frequencies where mode filtering for MD data is challenging. Comparison of geoacoustic information content is facilitated here using trans-dimensional Bayesian inversion to sample probabilistically over the number of layers of the seabed model as well as over the order of an autoregressive error model. Results indicate MF inversion resolves more-detailed geoacoustic structure with smaller uncertainties, including good estimates of the attenuation profile for wide-band (20–1500 Hz) inversions. No significant advantage is apparent for joint MF/MD inversion.

4:20

1pAOb4. Sound speed inversion of modal group delays of broadband 35-Hz transmissions during the Coordinated Arctic Acoustic Thermometry Experiment. Franklin H. Akins (Scripps, UCSD, 562 Arenas St., La Jolla, CA 92037, fakins@ucsd.edu), Matthew Dzieciuch (SIO, UCSD, La Jolla, CA), Peter F. Worcester, Bruce D. Cornuelle (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway)

The Arctic Ocean is evolving in response to climate change. In addition to the dramatic reduction in sea ice, the interior of the ocean is changing as the amount of warmer water that advects into the Arctic from the North Pacific and North Atlantic Oceans has increased. This paper applies linear inverse methods to the modal group delays of broadband 35-Hz transmissions across the Canada Basin during the Coordinated Arctic Acoustic Thermometry Experiment (CAATEX) in an effort to measure and understand the changes in the interior of the western Arctic. Modal dispersion along a ~850-km

transmission path leads to modal arrivals that can be separated in both time and space using data from 1200-m long vertical receiving arrays. The range-independent sound speed profile is estimated as a function of time over the year 2019–2020 from the measured modal group slowness. The inversion uses the Ice-Tethered Profiler (ITP) dataset to constrain the sound speed.

4:40

1pAOb5. Ambient sound effects and propagation characteristics seismic survey datasets recorded on OOI hydrophones. Alexander S. Douglass (JASCO Appl. Sci. (USA), Inc., 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu) and Shima Abadi (Univ. of Washington, Seattle, WA)

Marine seismic reflection surveys provide an abundance of acoustic data over survey regions and surrounding areas. The data from these surveys, in addition to passive observations made possible with networks like the Ocean Observatories Initiative (OOI), provide a wealth of acoustic propagation data in a variety of environments. Furthermore, the repeatability of airgun array shots yields a minimally variable source from which to extract more precise trends. In this work, we consider data from two seismic surveys, MGL1905 and MGL2104, during which a 6600 in^3 airgun array is fired every 37.5 m along multiple survey lines extending 10s to 100s of kilometers. The data collected on the OOI hydrophones is analyzed to quantify long-term trends in the ambient soundscape. Additionally, characteristics and artifacts in the data are examined for possible usefulness in extracting details about the propagating environment. [Work supported by ONR.]

MONDAY AFTERNOON, 19 MAY 2025 BALCONY K, 1:40 P.M. TO 4:40 P.M.

Session 1pBA

Biomedical Acoustics and Signal Processing in Acoustics: Super Resolution Ultrasound Imaging II

Libertario Demi, Cochair
Information Engineering and Computer Science, University of Trento, Via sommarive 9, Trento 38123, Italy

Pengfei Song, Cochair
Biomedical Engineering, Duke University, 100 Science Dr, Hudson Hall Annex 276, Durham, NC 27708

Invited Paper

1:40

1pBA1. Dynamic biomarkers in ultrasound localization microscopy. Jean Provost (Eng. Phys., Polytechnique Montreal, 2900 Boul Edouard-Montpetit, Montreal, QC H4B1Z1, Canada, jean.provost@polymtl.ca)

Ultrasound localization microscopy (ULM) provides non-invasive, deep-tissue imaging of the microvasculature by tracking millions of individually injected microbubbles, approved for human use, across hundreds of thousands of ultrasound images acquired within minutes. However, state-of-the-art ULM faces significant limitations: it cannot effectively map capillaries and is strongly hindered by tissue motion in the heart. In this work, we present and utilize a novel simulation framework that models connected vascular networks representing the entire vasculature of the mouse brain and human heart to predict the effects of singular value filtering, the mouse skull, and human cardiac motion on dULM image quality. This framework enables the development of novel acquisition sequences and image reconstruction algorithms. Specifically, we demonstrate the feasibility of performing dULM throughout the entire cardiac cycle in rats and pigs *in vivo* by employing a Lagrangian beamformer that virtually eliminates cardiac motion. Furthermore, we show how tracking microbubbles across thousands of frames allows for the detection of single capillary reporters in the mouse brain through both skin and skull *in vivo*, providing unprecedented insights into capillary function during neuroinflammation and other pathophysiological conditions.

Contributed Paper

2:00

1pBA2. Uncoupling a bi-disperse microbubble population for decreased ULM acquisition times: A proof-of-concept study. Giulia Tuccio (DISI, Univ. of Trento, via Sommarive, 5, Povo, Trento 38123, Italy, giulia.tuccio@unitn.it), Lisa te Winkle, Wim van Hoeve (Solstice, Enschede, Netherlands), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy)

Ultrasound Localization Microscopy (ULM) enables to accurately characterize micro-vascular structures by means of ultrasound imaging. In ULM, the backscattering signals of injected microbubbles (MBs) are utilized to sub-wavelength localize and track MBs flowing in the circulatory system. To ensure precise MB localizations, ULM is constrained to low MB concentrations, leading to prolonged acquisition times. Owing to patient comfort, motion, and computational cost to elaborate data, the constraint on low MB concentrations is a bottleneck for ULM clinical translation. To mitigate the need of low MB concentration, we propose to uncouple a bi-disperse MB population. The bi-disperse population is composed of two monodisperse MB populations having diameter (and resonance frequency) of 2.5 μm (3 MHz) and 3.8 μm (5.5 MHz), respectively. The different diameter determines a different resonance response. Experiments are performed injecting the populations singularly and simultaneously in a 3-D printed phantom. Uncoupling is performed by means of a signal processing pipeline, which exploits the strong nonlinear response of MBs having resonance frequency tuned with the transmission frequency. After uncoupling, super-resolved density and velocity flow maps are generated for each MB population. The results demonstrate the capability of uncoupling the selected pair of MB populations, thus potentially permitting increased MB concentrations.

Invited Paper

2:20

1pBA3. Power Doppler and color flow imaging with null subtraction imaging. Michael L. Oelze (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, oelze@illinois.edu) and Zhengchang Kou (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Null subtraction imaging (NSI) is a nonlinear beamforming technique that uses nulls in a beam pattern instead of the mainlobe to beat the diffraction limit. In this study, we used a 256-element linear array with 55-μm pitch to image the microvasculature of a mouse brain using NSI and delay and sum (DAS) with plane wave compounding. We also explored combining pulse inversion (PI) with NSI, i.e., we transmitted pulses at center frequency of 15.6 MHz to obtain the second harmonic at 31.2 MHz. Color flow imaging (CFI) at the fundamental frequency was also generated using traditional methods and NSI and compared. Higher spatial resolution and contrast in power Doppler images were observed in the NSI images compared to DAS (spatial resolution of 1/4–1/6 of a wavelength). The CFI images constructed using NSI resolved small vessels with flows in opposite directions that were not observed in the CFI using DAS, where vessels of opposite flows appeared averaged together as one vessel due to lower spatial resolution. The computation time was increased by only 40% for NSI compared to DAS. No contrast agents were used in these images.

Contributed Papers

2:40

1pBA4. Gas vesicle expression in stem cells and the potential as a biomarker application. John Kim (Mech. Eng., Univ. of Michigan, 3033 LENOX RD NE, # 23311, Atlanta, GA 30324, kjohnw@umich.edu), Alessandro R. Howells (Biomedical Eng., The Penn State Univ., State College, PA), Xiaojun Lian (Biomedical Eng., The Penn State Univ., University Park, PA), and Chengzhi Shi (Univ. of Michigan, Ann Arbor, MI)

Gas vesicles (GVs) are highly promising contrast agents that have been actively investigated throughout the development of contrast agents. Unlike chemically synthesized materials, GVs offer distinct advantages in bio-related studies due to their biocompatibility, making them a safer alternative to existing agents. GVs are gas-filled structures found in microorganisms, enabling their host to remain buoyant in aqueous environments. Since GVs are composed of gas vesicle proteins, it is possible to genetically encode specific cell lines to express gas vesicles at will. However, whether these cells can be utilized in clinical trials to self-contain contrast agents has not been fully explored. In recent years, we have been investigating efficient methods to express GVs in stem cells, which could function as a repeatable contrast agent. To achieve this, we employed a drug selection technique to isolate stem cells that contain gas vesicle genes. Traditional isolation methods required fluorescence-activated cell sorting (FACS) followed by single-cell cloning, which was time-consuming and costly. These findings may guide researchers in cultivating GV-containing stem cells, potentially differentiating them into desired cell types for implantation into organs that require biomarkers.

3:00–3:20 Break

3:20

1pBA5. Generation of modulators for super-resolution imaging. Jian-yu Lu (Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606, jian-yu.lu@ieee.org)

Medical ultrasound imaging has a trade-off between the lateral image resolution and penetration depth. Using the Point Spread Function (PSF) modulation super-resolution imaging method developed recently (Lu, IEEE TUFFC 2024), it is possible to overcome such a limit. To implement the method, an ultrasound array transducer such as a one-dimensional (1-D), two-dimensional (2-D), or annular array transducer can be used to produce a shear wave deep in the biological soft tissues. The shear wave is then focused to form a modulator of a size of about one half of the shear wave wavelength. Notice that since the speed of sound of the shear wave is much smaller than that of the longitudinal waves, the shear wave wavelength can be small. Finally, the modulator is used to induce a phase modulation to the imaging wave for super-resolution ultrasound imaging. In this work, the mechanism of shear wave focusing is studied both theoretically and through computer simulations. The results show that it is possible to focus a shear wave to achieve a wave feature of about a half wavelength of the shear wave, opening up a possibility for super-resolution ultrasound imaging deep in the biological soft tissues.

3:40

1pBA6. Scatterer localization technique for diverging-wave ultrasound acquisitions. Kashta Dozier-Muhammad (Biomedical Eng., The Univ. of Memphis, 3806 Norriswood Ave. Rm 321F, Memphis, TN 38152, kndzrmhm@memphis.edu) and Carl Herickhoff (Biomedical Eng., The Univ. of Memphis, Memphis, TN)

Ultrasound localization microscopy (ULM) provides super-resolution images by tracking microbubbles (MBs) in small vessels, but current MB localization algorithms can be computational complex. In this work, we investigate an MB localization technique for diverging-wave transmit acquisition schemes using multiple receive apodizations. Field II and the MATLAB Ultrasound Toolbox were used to simulate a phased-array transducer (Verasonics P4-2v) transmitting diverging wavefronts steered at $-5°$ and $5°$, and pulse-echo data were obtained for point scatterers placed laterally at -3, 0, and 3 mm, and axially at 12 and 16 mm deep. Two even-function and two odd-function receive apodization profiles were applied to yield four beamformed IQ data frames for each point scatterer. An expression for a quadratic line, fit as a function of wavefront steering angle and scatterer range (distribution of local amplitude maxima/minima), and an expression for precise inter-pixel scatterer position along the line based on the pixel amplitudes from the four IQ data frames, were empirically derived. The technique was implemented and tested experimentally on a wire-target phantom, which showed good agreement with simulations (within 30 μm). These results show that super-resolution ULM images from diverging-wave acquisitions can be rapidly produced from beamformed IQ data.

4:00–4:40 Panel Discussion

MONDAY AFTERNOON, 19 MAY 2025 STUDIO 6, 1:00 P.M. TO 3:40 P.M.

Session 1pCA

Computational Acoustics: Computational Methods II

Amanda Hanford, Chair
Penn State University, PO Box 30, State College, PA 16802

Contributed Papers

1:00

1pCA1. Impact of operational uncertainty on acoustic time difference of arrival multilateration performance. Matthias Ospel (French-German Res. Inst. of Saint-Louis, 5 Rue du General Cassagnou, Saint-Louis 68300, France, matthias.ospel@isl.eu)

This work addresses the influence of operational uncertainty on acoustic time difference of arrival (TDOA) multilateration systems, commonly used in source localization applications. A key challenge during the planning phase of multilateration applications is predicting system performance and resilience to uncertainty, particularly in topographically complex operational environments. A deterministic numerical approach using matrix factorization methods, including rank-revealing QR decomposition, is presented to evaluate system performance and assess sensor positions under specific conditions. Through path-based wave propagation simulations and experiments, we conduct sensitivity analyses to investigate the impact of timing inaccuracies, terrain features, and meteorological conditions on TDOA-based acoustic multilateration. The results are validated by experiments and calculations on the localization of shooters using sparsely distributed ground sensors at the military Lehnin training area, Germany. [Work supported by BAAINBw/WTD-91.]

1:20

1pCA2. Wideband acoustic simulation of anechoic chambers using a finite-difference method with porous media modeling. Jan W. Smits (Univ. of Edinburgh, Alison House, 12 Nicolson Square, Edinburgh EH8 9DF, United Kingdom, jan.smits@ed.ac.uk)

Accurate simulation of the acoustic field within anechoic chambers could be used to optimize the design of new chambers and for enhancing experiments in existing ones. Such simulations are challenging, however, due to the need to model the extended reaction of the wedge-shaped absorption structures that line the chamber walls. This study performs anechoic chamber simulations using a recently developed finite-difference method, which extends the low-dispersion face-centered cubic scheme with modeling of the acoustic propagation in porous volumes. Examples demonstrate the feasibility of computing wideband impulse responses for medium-sized rooms using a single GPU processor. The results are compared with chamber qualification standards and measurements from an open-source data set. Numerical experiments explore the effects of varying certain parameters, such as the room dimensions or the geometry and material properties of the absorbers.

1:40

1pCA3. Validation of a numerical approach for predicting sound absorption in porous materials. Elissa El Hajj (Polytechnique Montreal, 2900 Boul Edouard-Montpetit·, Montreal, QC H3T1J4, Canada, elissa.el-hajj@polymtl.ca), Niloofar Rastegar, Manuel Flores Salinas, Edith Roland Fotsing, Annie Ross, and David Vidal (Polytechnique Montreal, Montreal, QC, Canada)

Porous materials are widely used in acoustic absorption applications, including building acoustics and noise control, among others. Accurate characterization of the acoustic behavior of actual materials is essential for understanding sound absorption and predicting performance. However, current methods predominanlty depend on experimental techniques, which are resource-intensive and time-consuming. These approaches often fail to facilitate the identification of optimal solutions or explain why certain materials outperform others. This study addresses these limitations by validating a numerical approach for predicting the sound absorption properties of porous materials. High-resolution 3-D geometries are obtained using X-ray micro-computed tomography, and simulations using GeoDICT predict key parameters which are applied to the Johnson–Champoux–Allard model to estimate acoustic absorption. Numerical predictions are validated using two experimental approaches: a direct method measuring normal incidence sound absorption coefficients with an impedance tube, and an indirect method determining the materials' acoustic properties, incorporated into the JCA model for predicting the absorption coefficient. The results show strong agreement between numerical simulations and experimental measurements, confirming the reliability of the numerical approach. This validated methodology holds promise for characterizing virtual porous materials that have yet to be fabricated, thereby enabling numerical optimization of porous structures.

2:00

1pCA4. Long-range sound propagation modeling with the Semi-Analytic Finite-Element method. Rafael Castro Mota (Infrasound Res. group / Inst. for Acoust. and Dynam., Physikalisch-Technische Bundesanstalt / TU-Braunschweig, Braunschweig, Germany), Paul Williams (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Sydney, New South Wales, Australia), Ray Kirby (RMIT Univ., Melbourne, Victoria, Australia), and Stefan Jacob (Infrasound Res. group / Inst. for Acoust. and Dynam., Physikalisch-Technische Bundesanstalt / TU-Braunschweig, Bundesalle, 100, Braunschweig, Germany, stefan.jacob@ptb.de)

The prediction of long-range outdoor sound propagation has gained increasing relevance in recent years due to the low-frequency noise generated by renewable energy converters and the monitoring and localization of violent natural and human-made events. In this study, we present a numerical scheme based on the semi-analytic finite element (SAFE) method for modeling long-range sound propagation. The model computes the sound field using a compact basis of acoustic atmospheric modes, making it particularly efficient for predicting fields in the downwind direction under downward-refracting atmospheric conditions. We compare the SAFE solutions to high-resolution numerical results obtained from the linearized Euler equations (LEE) and demonstrate both qualitative and quantitative agreement. For large, range-independent domains, the SAFE method proves to be accurate and significantly more efficient than direct numerical LEE computations.

2:20–2:40 Break

2:40

1pCA5. Surrogate modeling terminology for acoustics: verification, validation, and uncertainty quantification. Kourtney Libenow (Acoust., Penn State Univ., Graduate Program in Acoust. The Penn State Univ. 201 Appl. Sci. Bldg., University Park, PA 16802, kra5346@psu.edu) and Gregory Banyay (Acoust., Penn State Univ., State College, PA)

Surrogate models are emerging as established parts of commercial problem-solving spaces including aeronautics, financial estimating, marketing, automotive, and weather predictions. Being less computationally expensive, these models generally require less expensive hardware, and less time for engineering design. When surrogate models ultimately inform safety and policy decisions prior to monetary decisions, one needs to demonstrate credibility for decision makers. Engineers should be sure that the uncertainty of models are quantified and that the models are verified and validated (a field called VVUQ). To create clarity when discussing VVUQ terminology especially applied to acoustics, we survey the way these terms are currently used and create a glossary for guidance going forward. We use the modeling process applied to a canonical structural dynamics system to demonstrate the application of this vocabulary.

3:00

1pCA6. Shape optimization of a synthetic jet actuator enclosure for aerodynamic efficiency and noise reduction using COMSOL Multiphysics. Trish A. Maduche (Mech. Eng., McGill Univ., 817 Sherbrooke St W, Montreal, QC H3A OC3, Canada, trish.maduche@mail.mcgill.ca) and Luc G. Mongeau (Mech. Eng., McGill Univ., Montreal, QC, Canada)

Synthetic jet actuators are devices that generate a pulsatile flow through cyclic suction and blowing while maintaining a zero net mass flux. Their distinct ability to operate with no external fluid source, short response time, and compactness makes them suitable for use in flow control in aircraft, thrust vectoring of jet engines, and sensor-cleaning in automobiles. However, synthetic jet actuators' acoustic emissions are an undesirable by-product of their performance. Therefore, this study presents the design and optimization of a streamlined enclosure that enhances the actuator's aerodynamic efficiency while reducing the radiated noise. A preliminary parametric study is conducted to determine the best initial design of the enclosure based on the enclosure efficiency calculated. Geometric and shape optimization are performed with the objective of minimizing the pressure drop (thus increasing the enclosure's aerodynamic efficiency). The determined far-field pressure, which quantifies the radiated noise, is also investigated. This study explores several factors, such as computational mesh construction, the suitable optimization algorithm method, and the maximum displacement setting (dmax) in COMSOL Multiphysics, to converge to the optimum enclosure design. The optimum enclosure is 3-D printed and experimentally tested to validate the computational results. [Work supported by McGill University.]

3:20

1pCA7. Predicting acoustic loading on rocket structure at launch. Joseph Ungerleider (Dept. of Eng., James Madison Univ., 42558 Angel Wing Way, Ashburn, VA 20148, ungerljs@dukes.jmu.edu), Valentina Paz Soldan Viscarra (Dept. of Eng., James Madison Univ., Harrisonburg, VA), and Caroline P. Lubert (Mathematics and Statistics, James Madison Univ., Harrisonburg, VA)

Rockets are powerful vehicles that play a critical role in space exploration, satellite deployment, and scientific research. Their value lies in their ability to overcome the challenges of Earth's gravity and enable human exploration of space. However, a major challenge in rocket launches is the threat of the massive acoustic loads that occur during liftoff. These acoustic loads can be caused by many factors, including engine exhaust, aerodynamic turbulence, and shock waves. Predicting acoustic loads on rockets is crucial because the excessive noise and vibration can cause damage to the vehicle's structure, equipment, and payload. By accurately predicting these loads, engineers can design rocket structures to withstand such vibrations and minimize damage. With the recent surge of interest in reusable rockets, predicting loading is especially important for rockets subject to a high number of cycles. A semi-empirical model, NASA SP-8072, was developed in 1971 using existing rocket data to predict the acoustic power generated by a supersonic rocket exhaust. Despite being over 50 years old, it is still the best model available today for acoustic load prediction on rockets. This paper will discuss and compare possible improvements upon the NASA SP-8072 model.

Session 1pEA

Engineering Acoustics, Biomedical Acoustics and Physical Acoustics: Acoustic Holography: Advances and Applications

Randall P. Williams, Cochair
Univ. of Texas at Austin, Austin, TX

Oleg A. Sapozhnikov, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—1:15

Invited Papers

1:20

1pEA1. History and current research in Nearfield Acoustical Holography. Earl G. Williams (Acoust., Naval Res. Lab., Naval Res. Lab, 4555 Overlook Ave. Code 7106, Washington, DC 20375, earl.williams@nrl.navy.mil)

Since its inception in 1980 at Pennsylvania State University, Nearfield Acoustical Holography (NAH) has seen over 40 years of continuous development finding its way into many avenues of research in air and underwater. Its popularity arises from the ability to yield the reconstruction of three-dimensional spatial pressure and acoustic intensity fields, along with the normal velocity and intensity on the surface of noise sources, uncovering the dispersion physics of vibrations and unraveling of the sources of sound. It has excelled as a research tool in experimental facilities for the study of the vibration of and radiation from underwater naval structures. A research of current interest is the prediction of the complete farfield from cylindrical NAH measurements on internally excited, finite shells. [Work supported by the Office of Naval Research.]

1:40

1pEA2. Measurement-based simulation in transcranial ultrasound therapies. Elly Martin (Medical Phys. and Biomedical Eng., Univ. College London, Medical Phys. and Biomedical Eng., Malet Pl. Eng. Bldg., Gower St., London WC1E 6BT, United Kingdom, elly.martin@ucl.ac.uk)

Accurate characterization of acoustic fields and sources is fundamental to planning and evaluation of transcranial ultrasound therapies. This talk will present methods for characterizing both single-element transducers and complex multielement arrays operating at sub-MHz frequencies, which can give rise to additional measurement challenges. Comprehensive characterization of multielement arrays, particularly element positioning and interelement cross-talk, is particularly important when performing aberration correction and beam steering in transcranial applications. I will present a characterization of the performance of a custom pseudorandom multi-element array, and the simplified source definitions derived from the measured holograms that are used to simplify the practical implementation of transcranial targeting of focused ultrasound. Finally, I will discuss the use of these source definitions in measurement-based simulations for prediction of acoustic field propagation through the skull, with comparisons between experimental measurements and computational models.

2:00

1pEA3. Acoustic Cloning and surface-based Digital Twins. Dirk-Jan van Manen (Dept. Earth and Planetary Sci., ETH Zurich, Sonneggstrasse 5, Zurich, ZH 8092, Switzerland, vdirk@ethz.ch), Jonas Müller, Johannes Aichele, Henrik Thomsen, and Johan Robertsson (Dept. Earth and Planetary Sci., ETH Zurich, Zurich, ZH, Switzerland)

Acoustic Cloning enables cloning the complete acoustic scattering behavior of unknown objects in a two-step process: in the first step, an object is insonified for a range of angles using temporally broadband sources while the response is recorded on a circular aperture enclosing the object. The scattering Green's functions (GFs) of the object are retrieved from the recorded responses by multi-dimensional deconvolution. In the second step, the object is removed and reproduced holographically for arbitrary, previously unseen, broadband incident wavefields. At the heart of each acoustic clone are the scattering GFs of the object for radiation conditions. In the reproduction step, these scattering GFs enable real-time extrapolation of arbitrary incident wavefields by acting as the kernel of a Kirchhoff–Helmholtz (KH) integral. Extrapolated to a set of monopole and dipole sources, the GFs allow reproducing the scattered wavefield without the object being present. These scattering GFs can thus be regarded as a surface-based Digital Twin (SBDT) of the scattering object. We present examples of real, experimentally acquired SBDTs and show that they can be probed both numerically and experimentally. The SBDTs open new avenues for inclusion and manipulation of complex real acoustic scatterers in real or numerical scattering environments.

2:20

1pEA4. Acoustic holography methods for characterizing transducers, fields, and materials in therapeutic ultrasound. Wayne Kreider (CIMU / APL, Univ. of Washington, 1013 NE 40th St., Appl. Phys. Lab., Seattle, WA 98105, wkreider@uw.edu), Randall P. Williams (Appl. Res. Labs., The Univ. of Texas, Austin, TX), Pavel B. Rosnitskiy (Dept. of Medicine, Div. of Gastroenterology, Univ. of Washington, Seattle, WA), Vera A. Khokhlova (Dept. of Medicine, Div. of Gastroenterology, Univ. of Washington, Moscow), Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Moscow, Russia and CIMU / APL, Univ. of Washington, Moscow), and Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russia and CIMU / APL, Univ. of Washington, Seattle, WA)

Medical ultrasound applications typically involve megahertz frequencies and millimeter wavelengths in conjunction with centimeter length scales for sources and propagation distances. In this regime, holography techniques for source characterization based on a planar hydrophone scan and linear projection of the corresponding field are well known. However, topics of interest remain regarding practical implementation, e.g., acceleration of hydrophone scans, alignment of field reconstructions, and estimation of relevant uncertainties for inclusion in international standards. In addition, as modern tools facilitate the routine acquisition and processing of large holograms that fully capture an arbitrary beam and its angular spectrum, applications emerge beyond the characterization of therapeutic and diagnostic transducers. Such applications include (1) calculation of radiation force on a spherical particle that represents a kidney stone; (2) high-precision measurement of absorption and sound speed in layered materials without diffraction artifacts; (3) calculation of radiation force on a wide, absorbing target to enable more accurate calibration of hydrophone sensitivity with a radiation force balance; and (4) calibration of hydrophone directional responses by consideration of reference holograms measured with a small hydrophone. Here we describe implementations of holography across this range of applications with sample results. [Work supported by NIH, Grant R01EB025187.]

2:40–3:00 Break

Contributed Papers

3:00

1pEA5. Reconstruction of the bistatic response of an elastic object with edges using its estimated acoustic impedance matrix from diffuse field acoustic holography. Vincent Roggerone (Lab. of Mech. and Acoust., CNRS, Ave. de la tour royale, Toulon 83050 Cedex, France, roggerone.vincent@live.fr) and Sandrine T. Rakotonarivo (Aix-Marseille Univ. and Lab. of Mech. and Acoust., Marseille, France)

This study aims to compute the bistatic response of an elastic object submerged in water based on preliminary measurements of its mechanical impedance in air. This approach allows to predict the field scattered by the object in any environment without prior knowledge of its internal structure, relying on a single measurement. The object's mechanical impedance, which is its response in a vacuum, is estimated in a diffuse noise field. The pressure is measured on two conformal holographic surfaces near the object. Incoming and outgoing fields are separated using either the method of equivalent sources (for objects without geometric discontinuity) or a new alternative method for objects with edges. For more accurate velocity reconstruction in air, the acoustic response of a rigid object with the same shape is computed numerically and then subtracted. Pressure and velocity fields are retropropagated to extract the mechanical impedance matrix which is then used to calculate the bistatic response in water. The approach is tested numerically with two plastic cylindrical shells: one without geometric discontinuity and one with edges. Results demonstrate the possibility of reconstructing bistatic responses in water for both object types, with higher accuracy for objects without discontinuities.

3:20

1pEA6. Insights on deciphering the network of hidden internal soil pipes in agricultural fields using acoustic techniques. Md Abdus Samad (NCPA, Dept. of Civil Eng., Univ. of Mississippi, 145 Hill Dr., University, MS 38677, msamad@go.olemiss.edu), Craig J. Hickey (NCPA, Univ. of Mississippi, University, MS), and Leti T. Wodajo (NCPA, Univ. of Mississippi, Oxford, MS)

Soil piping significantly accelerates erosion in agricultural fields, contributing to substantial soil loss. However, quantifying its impact is challenging due to limited direct observations. Soil pipe locations are often inferred from surface features, such as flute holes and gully windows, or through methods like dye tracer testing, which involves injecting fluorescein dye into uppermost pipe collapse features and sampling downslope. This study investigates an acoustic technique for mapping soil pipe networks, leveraging the propagation of sound waves through air-filled pipes. These waves generate seismic vibrations in the surrounding soil, detected at the surface by geophones. Field measurements were conducted at the Goodwin Creek experimental site with extensive soil pipe networks supported by six gully windows. A speaker placed in various gully windows generated sound waves and surface vibrations were recorded along two lines. Recorded data were analyzed in the frequency domain, calculating energy content using the Riemann sum approximation. Signal-to-noise ratios (S/N), probability density functions (PDFs), and Z-scores were computed for each geophone. High S/N and Z-scores (>2) indicated proximity to large, shallow pipes, while moderate scores ($1 < Z < 2$) suggested smaller or deeper pipes. Results were validated with *in situ* investigations, enabling detailed mapping of primary and secondary soil pipe networks.

3:40

1pEA7. Active noise cancellation in space containing scattering objects based on kernel interpolation. Kota Yamano (Graduate School of Information Sci. and Technol., The Univ. of Tokyo, Tokyo, Japan), Shoichi Koyama (National Inst. of Informatics, Eng. Bldg. 6-140, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, shoichi_koyama@ipc.i.u-tokyo.ac.jp), and Hiroshi Saruwatari (Graduate School of Information Sci. and Technol., The Univ. of Tokyo, Bunkyo-ku, Japan)

A spatial active noise control (ANC) method for canceling noise in a region containing scattering objects is proposed. Spatial ANC aims to suppress noise over a target control region using multiple microphones and loudspeakers. The driving signals of the loudspeakers are obtained adaptively based on the interpolation of the sound field from the microphone measurements. However, most spatial ANC methods assume that the target region is free space, i.e., contains no scattering objects, because they are based on the sound field interpolation methods that do not take into account the presence of scatterers within the target estimation region. On the contrary, in spatial ANC, ANC users will be present within the target region, which may deteriorate their regional noise reduction performance, especially at high frequencies. We propose a spatial ANC method based on the sound field estimation for the region containing scattering objects. The incident sound field in the target region is estimated based on the kernel ridge regression under the condition that the interpolated function satisfies the homogeneous Helmholtz equation, with the scattering effects eliminated. The numerical experiments indicated that the proposed method outperforms the existing methods at high frequencies.

4:00

1pEA8. Imaging in a cylindrical acoustic waveguide using the linear sampling method. Jie CUI (Dept. of Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong University of Sci. and Technol., Hong Kong, Hong Kong, jcuiah@connect.ust.hk) and Mohamed S. Ghidaoui (Dept. of Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

The widespread occurrence of duct defects necessitates effective obstacle detection within ducts. This study presents an imaging method utilizing acoustic waves in a cylindrical duct. Specifically, the linear sampling method is employed to address the inverse scattering problem in a cylindrical acoustic waveguide. By projecting onto modes, we obtain a modal formulation of the linear sampling method, resulting in a system of linear equations. The matrix of this system depends on the scattering coefficients, while the inhomogeneous term is influenced by the positions of the sampling points. For each sampling point, the norm of the solution to this system serves as an indicator for imaging. The method is validated through several numerical experiments, where its feasibility is thoroughly analyzed and confirmed. Notably, this study demonstrates that the linear sampling method can be applied to imaging in three-dimensional cylindrical waveguides, extending its use beyond previous theoretical analyses and two-dimensional experiments. Also, a prior matrix decomposition is used to improve the algorithm's efficiency.

MONDAY AFTERNOON, 19 MAY 2025 — GALERIE 6, 1:20 P.M. TO 4:00 P.M.

Session 1pMU

Musical Acoustics: General Topics in Musical Acoustics II

May Pik Yu Chan, Cochair
Linguistics, University of Pennsylvania, Department of Linguistics 3401-C Walnut Street, Suite 300, Philadelphia, PA 19104-6228

Vasileios Chatziioannou, Cochair
Department of Music Acoustics, University of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria

Contributed Papers

1:20

1pMU1. Pitch-dependent vowel space adjustments by professional singers. May Pik Yu Chan (Dept. of Linguist, Univ. of Pennsylvania, Dept. of Linguist 3401-C Walnut St., Ste. 300, C Wing University of Pennsylvania, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu), Jonathan Havenhill (Dept. of Linguist, The Univ. of Hong Kong, Hong Kong, Hong Kong), and Jianjing Kuang (Dept. of Linguist, Univ. of Pennsylvania, Philadelphia, PA)

Singers are trained to adjust their resonance space depending on pitch target. While existing studies on classical singers have focused on modeling the resonance profile, the articulatory correspondences have been less explored. The goal of this study is to model change in tongue position—a key factor that shapes the resonance space—across singers' pitch range using articulatory data. Fifteen professional singers sang five sets of English vowels at each semitone across their pitch range. Each set included the target vowels ([i], [ɛ], [æ], [ɑ], [u]) in randomized order, and a filler vowel ([ɔ]) closing the breath group. Midsagittal ultrasound tongue images were collected at 81.5 frames per second. Tongue position was automatically tracked using DeepLabCut. Preliminary results of eight participants modelled using Functional Data Analysis show gradual neutralization of vowel contrast as pitch increases relative to the singers' range. Specifically, high vowels tended to lower and low vowels tended to raise at higher pitches. These results suggest that articulatory adjustments operate in vowel-specific directions. Analysis by singer gender and voice type is underway. We discuss these findings in relation to vowel–pitch interactions in voice production.

1:40

1pMU2. Pitch-dependent adjustments of the vocal tract in the singing of Japanese traditional music. Tokihiko Kaburagi (Kyushu Univ., Minami-ku Shiobaru 4-9-1, Fukuoka city, Fukuoka Pref. 815-8540, Japan, kabu@design.kyushu-u.ac.jp) and Mizuki Somura (Kyushu Univ., Fukuoka, Japan)

The adjustment of vocal-tract resonance can enhance the perceptual effect of singing. For example, resonance tuning involves aligning vocal-tract resonance frequencies near the fundamental or harmonic frequencies, a technique well studied in operatic singing. This study focuses on *satsuma-biwa*, a musical genre of Japanese traditional narrative singing that is accompanied by the *biwa*, a pear-shaped short-necked lute. We measured the vocal tract of a professional female singer in 3-D using magnetic resonance imaging as she sang three vowels (/a/, /i/, and /u/) at pitches of 174, 233, 349, and 466 Hz. Notably, the highest pitch was produced using the falsetto register, whereas the others utilized the modal register. We obtained the cross-sectional area function from volumetric images and estimated the transfer characteristic of the vocal tract using an acoustic tube model. Our results clearly showed adjustments in the vocal tract, as evidenced by morphological and acoustic data, which varied according to both vowel and pitch conditions. Specifically, as the pitch increased, the frequency of the first formant also increased for the /i/ and /u/ vowels, aligning with the resonance tuning strategy.

2:00

1pMU3. From roughness to coincidence—Concepts of dissonance and consonance. Christoph Reuter (Systematic Musicology, Univ. of Vienna, Campus Court 9, Spitalgasse 2, Vienna 1090, Austria, christoph.reuter@univie.ac.at) and Robert Mores (Hamburg Univ. of Appl. Sci., Hamburg, Deutschland, Germany)

150 years after the first English translation of Hermann von Helmholtz's opus "On the Sensations of Tone," we review existing consonance theories that have emerged from this seminal work: According to Helmholtz, the degree of roughness between the partials of two simultaneously sounding tones determines the degree of consonance of an interval (the rougher the more dissonant). For chords, on the other hand, Helmholtz saw the difference tones as a way of determining consonance (the more the difference tones correspond to pitches already present in the chord, the more consonant it is). Based on these two psychoacoustic phenomena (roughness and difference tones), a whole series of consonance theories were formed. However, these psychoacoustic explanations proved to be limited in the course of the 20th century. Coincidence theories offered an alternative here, according to which the more matching partials (frequency domain) or periods (time domain) existing in the sounds of an interval, the greater the sensation of consonance. This perspective has been supported since the 1960s by perception experiments on intervals with compressed and stretched partial series. Within the scope of the article, an Internet application is introduced with which such sounds and intervals can be generated easily and intuitively.

2:20–2:40 Break

2:40

1pMU4. Cross-checking and reflecting existing concepts of dissonance and consonance. Robert Mores (Hamburg Univ. of Appl. Sci., Finkenau 35, Hamburg, Deutschland 22081, Germany, robert.mores@haw-hamburg.de) and Christoph Reuter (Systematic Musicology, Univ. of Vienna, Vienna, Austria)

This study re-investigates existing concepts of dissonance and consonance, then reflects these concepts from a musical perspective, and finally searches for neuroscience evidence. First, as a result of careful study, the individual experiments of Stumpf in Halle and Prag allow for a much more differentiated conclusion than summarized by Stumpf himself. This study re-compiles the experiment's outcome. Second, the prominent concepts of the 20th century, which are based on the critical bandwidth (Plomp and Levelt, 1965), on listening tests (Kameoka and Kuriyagawa, 1969), and on auto-correlation (Ebeling, 2008), are re-evaluated and compared by mutual swapping of parameters between models. The resulting rankings of interval consonance agree well with the earlier findings: octave, fifth, fourth, major third, tritone, minor third, major sixth, in most cases. In terms of reflection, third, the past discussion on consonance and dissonance seems to be narrow. From today's perspective, they are ignoring some Gestalt aspects. This is particularly true when reflecting on early church music and on present Eastern European folk music, for instance. Finally, a brief review on recent findings in neuroscience seeks evidence for auto-correlation in the hearing physiology, the brain stem, and the cortex and subcortex.

3:00

1pMU5. Ideal tempo and pitch for two-source mashups. Anh Dung Dinh (Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., The Hong Kong University of Sci. and Technol., Clear Water Bay, Hong Kong, New Territories, Hong Kong, addinh@connect.ust.hk), Xinyang Wu (Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, New Territories, Hong Kong), and Andrew B. Horner (Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Creating a mashup by combining elements from multiple songs typically requires adjusting the tracks to match harmonically and rhythmically. Unless specifically intended, mashup artists often align the tracks' tempos and keys to ensure a pleasant listening experience. Therefore, mashups usually consist of songs with similar tempos and keys, minimizing modifications and distortions. However, artists may wish to explore more diverse song combinations, invoking the question of how the tracks are adjusted will impact perception of the composition. This study investigates listener preferences for tempo and key in two-source mashups. Through an ongoing survey, participants choose between different renditions of various mashups, combining vocal and instrumental tracks from a selection of 10 pop songs, at different tempos and keys. Preliminary results indicate that listeners generally prefer the average tempo of the original tracks while favoring the vocal tracks' original keys. This suggests that alterations to the vocal component are perceived as more disruptive, highlighting the importance of maintaining vocal integrity in mashup creation. Further data will enrich our understanding of these trends and their implications for mashup artists as well as automatic mashup algorithms.

3:20

1pMU6. Retrieval-based automatic mashup generation with deep learning-guided features. Darin Chau (Dept. of Comput. Sci., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Hong Kong, Hong Kong, Hong Kong, yfdchau@connect.ust.hk) and Andrew B. Horner (Dept. of Comput. Sci., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Pop song mashups have emerged as a prominent research direction in the field of creative music information retrieval (MIR). The goal of a successful mashup system is to create a mashup, a creative synthesis of two or more pop songs that maintains perceptual fidelity to the original tracks. We propose a retrieval-based mashup generation pipeline that identifies compatible source materials based on deep learning-guided features from an extensible collection of contemporary pop songs. The selection heuristic is rooted in the mashability score developed by Davies *et al*. (2014). It is further enhanced with recent advancements in music information retrieval, self-supervised latent features with autoencoders, and precomputation of normalized spectral features to ensure efficient large-scale searches. Mashups are crafted by swapping stems between the chosen input and compatible tracks, followed by mastering with equalization (EQ) adjustments predicted from the original tracks. The effectiveness of our system is evaluated through a subjective listening test, which measures the perceptual similarity between the original tracks and the mashups, the accuracy of the compatibility heuristic in predicting user preferences, and the system's performance in comparison to established mashup services like RaveDJ.

3:40

1pMU7. Music mashup generation and trend prediction using hybrid attention net. Xinyang Wu (Hong Kong Univ. of Sci. and Technol., The Hong Kong University of Sci. and Technol., Clear Water Bay, Hong Kong, New Territories, Hong Kong, xwuch@connect.ust.hk) and Andrew B. Horner (Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Music mashups creatively blend elements from different songs to create new auditory experiences. Many researchers have explored automating the mashup creation process, often relying on "mashability" metric to determine compatible sections between tracks. We introduce a novel approach to replace this metric. In this study, we first present a framework that generates music mashups by exchanging stems between two distinct tracks. With a thorough analysis of the original music before generation, it uses advanced beat synchronization to maintain musical integrity. In our experiment setup, we selected 10 diverse songs from various styles and tempo ranges to generate 360 unique mashup pieces using 4 distinct strategies. To assess these creations, we conducted rigorous listening tests using a purposely designed format. Three separate groups of music enthusiasts were engaged to assess different mashup strategies, ensuring a comprehensive and robust evaluation. Based on the results of these listening tests, we developed a strategy-encoded hybrid attention network. This model was specifically trained to predict the rankings of mashups that share one common song, enabling it to identify the most compatible pairs given a single input song and a selected strategy from the database. The model demonstrates high predictive accuracy, with a Spearman ranking correlation of 0.75 on the test set, closely aligning with the rankings from the listening tests.

Session 1pNS

Noise: Assessment of Low-Frequency Sound in Noise Criteria

Walter A. Montano, Cochair
Acoustics research, ARQUICUST, Luis Clavarino 1227, Gualeguaychu E2820BSG, Argentina

Bhisham Sharma, Cochair
Mechanical Engineering - Engineering Mechanics, Michigan Tech University, 1400 Townsend Drive, Houghton, MI 49931

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 County Road 102, Oxford, MS 38655-8604

Chair's Introduction—12:55

Invited Paper

1:00

1pNS1. Low-frequency sound: Method to identify tones masked by high background noise. Walter A. Montano (Acoust. Res., ARQUICUST, Luis Clavarino 1227, Gualeguaychu, Entre Rios E2820BSG, Argentina, wmontano@arquicust.com), Alice E. Gonzalez (Acoust. Res., Eng. Faculty, Montevideo, Uruguay), Elena I. Gushiken (Acoust. Res., ARQUICUST, Lima, Peru), Pablo Gianoli, and Julian Ortiz (Acoust. Res., Eng. Faculty, Montevideo, Uruguay)

Often, noise measurements are performed under non-ideal scenarios, such as high background noise levels or high traffic noise levels, which is a problem when the goal is to analyze low-frequency noise from HVAC, industrial plants, wind turbines, etc., not only to study its impact but also to determine its immission level for legal purposes. Since the introduction of the A-weighting curve, acousticians have been aware that noise measured in dBA levels is not the best tool for assessing low-frequency annoyance, and for this reason some standards and some regulations include a correction factor to consider its presence, in order to add some penalties due to it. Following the theoretical concept of "sound designation" presented in the ISO 1996-1 standard, the authors have developed a method to find low-frequency tones from steady noise sources masked by background noise, not by filtering but by removing the unwanted sound levels from the value file as if they were outliers. Since standards and legislation require the evaluation of the specific sound of sources with emissions containing low frequencies, and not the total sound recorded by SLMs, this paper explains an understandable heuristic mathematical method for doing this, along with a statistical demonstration to validate these algorithms.

Contributed Paper

1:20

1pNS2. Drone-based acoustic measurement for noise level mapping of emitted low-frequency sound from high rise building. Dhany Arifianto (Medical Technol., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), Ahmad A. Najib (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, East Java, Indonesia), Muhammad A. Asyraf (Eng. Phys., Institut Teknologi Sepuluh Nopember, Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, maasyraf.id@gmail.com), Ghunawan Mandala Wibisana, Savitri Rizquna Azzahra Dwihar, and Daffa 'Alauddin (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Noise emitted from the high floor of a building can transmit into the outdoor environment, disturbing the surrounding multiple plane level environment. Defining the sound leakage point from the building is challenging since the measurement needs to be taken on a certain plane level from the ground. Drone-based acoustic measurement then proposed to obtain the noise profile at a certain level from the ground. The sound profile of the drone on standby and hovering state were firstly measured by attaching a calibrated recorder on the drone. This step is necessary to separate and filter the measured sound from the drone sound. The measurement was conducted based on the early measurement that suggested the leakage point was located on the top of the building. The recorded sounds were processed to obtain sound level data. The noise mapping on each frequency band is created, especially for low frequency range. Specific frequency of noise source confirmed by measuring the sound and vibration occurred in the affected area. The results indicate that the annoyed frequency was around 75 Hz, and the noise map on this frequency marks the leakage point on top of the building.

Invited Papers

1:40

1pNS3. Alternative method to determine the residual sound in quiet zones, quiet residential areas, or describing soundscapes. Walter A. Montano (Acoust. Res., ARQUICUST, Luis Clavarino 1227, Gualeguaychu, Entre Rios E2820BSG, Argentina, wmontano@arquicust.com), Alice E. Gonzalez (Acoust. Res., Eng. Faculty, Montevideo, Uruguay), Elena I. Gushiken (Acoust. Res., ARQUICUST, Lima, Peru), Pablo Gianoli, and Julian Ortiz (Acoust. Res., Eng. Faculty, Montevideo, Uruguay)

It is important to know the residual sound in a location to estimate the impact of noise sources prior to their installation, but not only the dBA sound level should be known, it should also be important to know its one-third octave band levels. For describing or measuring residual sound levels in quiet areas, an ANSI/ASA standard recommends procedures adapted to get a single unit in dBA, but not intended to characterize the sound in specific quiet spots or natural places. Wind turbines, pumping and compression stations for natural gas facilities, and other industrial facilities with high levels of low-frequency noise are usually installed near wilderness areas, so it is important to have a description of the entire soundscape in terms of sound spectrum rather than just a single dBA value. An empirical method was developed using measurements recorded in 125-ms samples, to estimate the residual sound levels based on a 90th (or other value) percentile sound level like a threshold, an algorithm rejects those samples above it and keeps those below, then a new spreadsheet is generated with the lowest sound levels contained in the original file, in that way not only the lower $L_{\mathrm{Aeq,T}}$ value is obtained but also $L_{\mathrm{Ceq,T}}$, $L_{\mathrm{Zeq,T}}$, full spectrum, and other noise descriptors.

2:00

1pNS4. Falcon-9 ascent sonic boom measurements in Ventura County, California. Makayle S. Kellison (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, makayle@byu.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (Dept. of Biology, California State Univ., Bakersfield, Bakersfield, CA), Mark C. Anderson, Levi T. Moats, Marcus T. Perkins, Noah L. Pulsipher, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Motivated by resident complaints and potential environmental concerns, measurements of Falcon-9 sonic booms created during vehicle ascent were collected in Ventura County, California. These booms are often low-frequency events that rattle structures, similar to earthquakes. During Summer 2024, a total of 132 measurements across six launches (16–25 per launch) were collected, spanning an area of ~200 square miles (~500 square km). The launches, from Vandenberg Space Force Base around 80–90 miles (130–145 km) to the north, varied by time of day, weather conditions, and trajectory. Described in this paper is the measurement campaign as well as initial results that include maps of peak overpressure values. Perceived level is also calculated to compare Falcon-9 ascent booms to aircraft sonic booms and other transient sounds. The roles of launch azimuth and weather in explaining ascent boom variability are discussed. This measurement campaign represents an important baseline for a broader effort spanning multiple seasons and a greater number of launches. [Work supported by USSF through USACE.]

2:20–2:40 Break

Contributed Papers

2:40

1pNS5. Long-term acoustic monitoring and modeling for improved noise management in electrical substations. Bruno Masiero (Universidade de Campinas, Av. Albert Einstein, 400, Campinas, SP 13083-852, Brazil, masiero@unicamp.br), Rodolfo Thomazelli, Maria Eugênia Fernandes, and Paulo S. Barbosa (Universidade de Campinas, Campinas, SP, Brazil)

In recent decades, Brazil has expanded its power grid, enabling energy from northern hydroelectric projects to reach the south. This requires extensive transmission lines and large substations. A recently built substation operating at 500/440/13.8 kV spans an area of 240,000 m^2 in what was a predominantly rural area. Despite compliance with noise regulations verified via measurement campaigns conducted by environmental agencies, numerous complaints related to noise disturbance highlight potential inadequacies in current standards, such as insufficient sampling points or short recording durations. This study presents the framework for long-term noise monitoring, where comprehensive acoustic measurements will be conducted under various load and weather conditions across numerous points within a 1-km radius from the substation. Integrating these extensive datasets into computational simulations will enable the creation of an accurate and detailed noise map of the substation and its surroundings. By combining experimental field data with advanced computational modeling, this project provides a robust foundation for environmental noise management around electrical substations, ensuring compliance with regulatory standards and minimizing community impact.

3:00

1pNS6. The uncertainty calculation of reference sound source calibration based on ISO 3741:2010. Nurvita Aji (National Standardization Agency of Indonesia, Serpong Lagoon Residence, Area A7 No. 42, Kranggan, Setu, Tangerang Selatan, Banten 15312, Indonesia, ajinurvita@bsn.go.id), Cafer KIRBAŞ, and Enver SADIKOĞLU (TÜBİTAK UME, Gebze, Turkey)

As of October 2024, there is no Calibration and Measurement Capability (CMC) for a reference sound source registered in the Key Comparison Database (KCDB) of the International Bureau of Weights and Measures (BIPM). A reference sound source provides a standard for determining sound power and quantifying the noise emission produced by the source. This report investigates the calibration of reference sound sources for determining sound power based on the ISO 6926:2016 and ISO 3741:2010 standards. The calibration process involves reverberation rooms, designed to control sound reflections and minimize environmental influences, ensuring precise measurements. The report outlines key concepts in sound power determination, including sound pressure level (SPL), reverberation time, background noise, and absorption area. It emphasizes the importance of correcting for environmental factors, particularly background noise, which can distort measurement accuracy. Background noise correction is applied when the SPL difference between the device under test and the background noise is insufficiently large, ensuring reliable results. The report also discusses the uncertainty budget in sound power determination, highlighting the need for

precision in acoustics measurements. The expanded measurement uncertainty is calculated using a coverage factor, ensuring high confidence in the final sound power result.

3:20

1pNS7. Measurement and evaluation of vertical transmission of low frequency noise from music room to clinic assessment room on multipurpose building. Dhany Arifianto (Medical Technol., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), Muhammad A. Asyraf (Eng. Phys., Institut Teknologi Sepuluh Nopember, Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, maasyraf.id@gmail.com), Ghunawan Mandala Wibisana, Hana Qonita Ainani Tajrian, Farah Zamir Salsabila, and Narphati Nawak Pandu Trawacha (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Assessment room in the clinic needed to be quiet so the background noise did not disturbed the patient examination by the doctor. Thus, the position of the clinic is usually located far from high level sound sources or has an adequate insulation system. However in this case, the music room has a noise source located on one level below the clinic assessment room. The measurement and evaluation of noise transmission conducted by measuring the sound level on each room during music played. The results indicate that the existing insulation is only effective to reduce the high frequency noise, when the low frequency part has a high level on the received room. The transmission path of this low frequency noise is then defined by measuring the vibration of connector structures that connect the music room and clinic assessment room. The results indicate that the low frequency noise transmitted through the connected window frame. Additional low frequency insulation structure recommended to reduce the transmitted noise.

3:40

1pNS8. Comparison of sound power levels of vehicles traveling on expressways determined by the Square integration method and the Pass-by method. Kimikazu Ikeya (NEXCO Res. Inst. Japan, 1-4-1, Tadao, Machida, Tokyo 194-8508, Japan, k.ikeya.ab@ri-nexco.co.jp), Daichi Yokomise (NEXCO Res. Inst. Japan, Machida, Tokyo, Japan), Tomoyuki Itiki, Akinori FUKUSHIMA (NEWS Environ. Design Inc, Kobe, Japan), and Yasuaki Okada (Meijo Univ., Nagoya, Japan)

This paper reports on unit patterns (time-histories) of traveling automobile noise measured on expressways. In general, symmetrical patterns were the most common, but many patterns with high noise levels were also recorded for approaching vehicles and after the vehicles had passed. The A-weighted sound power level $L_{WA,\mathrm{Fmax}}$ determined by the Pass-by method were compared with the $L_{WA,\mathrm{RMS}}$ obtained by the Square integration method. The noise unit patterns were classified by pavement and vehicle type. In all groups, $L_{WA,\ \mathrm{Fmax}}$ corresponded well with $L_{WA,\mathrm{RMS}}$, and the average level difference between them were 0.1 and 0.4 dB. It was found that the difference in the levels of $L_{WA,\ \mathrm{Fmax}}$ and $L_{WA,\mathrm{RMS}}$ was related to driving speed, which was thought to be influenced by the time weighting circuit (characteristic F) of the sound level meter. However, the effect is considered to be less than 0.5 dB if the vehicle had been traveling at a speed of about 100 km/h.

4:00–4:20 Panel Discussion

MONDAY AFTERNOON, 19 MAY 2025 BALCONY I, 1:00 P.M. TO 5:00 P.M.

Session 1pPAa

Physical Acoustics and Biomedical Acoustics: Acoustofluidics

Max Denis, Cochair
University of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008

James Friend, Cochair
Mechanical and Aerospace Engineering, University of California San Diego, 9500 Gilman Dr MC0411, MADLab SME344K, La Jolla, CA 92093

Charles Thompson, Cochair
Electrical and Computer Eng., UMASS Lowell, 1 Univ. Ave, Lowell, MA 01854

Invited Papers

1:00

1pPAa1. Acoustic bubble in acoustofluidics: A versatile tool for biomedical applications. Yuan Gao (Mech. Eng., Univ. of Memphis, 3815 Central Ave., 314 Eng. Sci. Bldg., Memphis, TN 38152, ygao6@memphis.edu)

With the development of microfluidics, micro- and nanoscale bubbles have been considered an emerging tool and have become a field of growing interest for addressing challenges in biomedical applications. When acoustic fields interact with bubbles in microfluidics, they induce unique physical phenomena, which offer promising capabilities in the manipulation of cells, particles, and fluids. This presentation will highlight the versatility and innovative use of acoustic bubbles in biomedical applications. The versatile functions of acoustic bubbles will be discussed through three key areas in the biomedical field: fluid transportation, cell manipulation, and biomaterials fabrication. Specifically, we will begin by introducing how acoustic bubbles enable fluid transportation through controllable pumping, mixing, and filtering in microfluidic systems. Next, we will demonstrate how acoustic bubbles facilitate cell manipulation, including enhancing ultrasound thrombolysis and engineering tumor spheroids. Lastly, we will discuss how acoustic bubbles can be applied to create tunable porous biomaterials for tissue engineering.

1:20

1pPAa2. Intelligent acoustofluidic patches for personal healthcare. Feng Guo (Intelligent Systems Eng., Indiana Univ. Bloomington, 043G Simon Hall, Indiana University, Bloomington, IN 47408, fengguo@iu.edu)

Acoustics have tremendously improved medicine by aiding in diagnosing, monitoring, treating, and managing various medical conditions. However, traditional acoustic medical approaches like acoustofluidics still rely on manual operation and post-test data analysis, leading to inaccuracies, low efficiency, and limited robustness in real-world settings. The integration of artificial intelligence (AI) could address these challenges, thanks to its ability to learn from data, make accurate predictions, and automate dynamic system control. As such, our group is dedicated to developing and integrating "Intelligent Acoustofluidics" by leveraging innovative acoustofluidic device design, sensor fusion, and AI-guided system integration to interface with individual patients. Here, we will discuss our recent development of intelligent acoustofluidic patches to showcase their translational applications in personalized therapy. We hope to create innovative intelligent systems that are highly efficient, accurate, and convenient, ultimately revolutionizing precision and translational medicine.

1:40

1pPAa3. Trapping forces by multidimensional fields on particles beyond Rayleigh regimes: From Gorkov potential to Born approximation. Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38655, zhang@olemiss.edu) and Xudong Fan (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS)

The analysis of trapping forces on particles much smaller than the sound wavelength (i.e., in the Rayleigh regime) has been derived from the Gorkov potential, resulting in a gradient force. This leads to the widely used concept of the acoustic contrast factor in trapping with one-dimensional standing waves. The Gorkov potential is also applicable for trapping in the Rayleigh regime using two- or three-dimensional acoustic fields, including both standing and progressive wave fields where compound gradients dominate the gradient force. The contrast factor formula was extended for trapping by two-dimensional standing waves, three-dimensional Bessel beams, and vortex beams [Fan and Zhang, Phys. Rev. Appl. 11(1), 014055 (2019)]. This contrast factor incorporates beam parameters alongside the mass density and compressibility contrast between particles and the surrounding medium. For particles larger than the Rayleigh regime, trapping forces can still be analyzed using the Gorkov potential under the condition of weak scattering, as described by the Born approximation, when the mass density and compressibility contrasts between the particles and the medium are small. This approach offers a simplified framework for examining trapping forces on larger particles [Fan and Zhang, J. Acoust. Soc. Am. 154(5), 3354–3363 (2023)].

2:00

1pPAa4. Scalable device architectures for acoustofluidic separation, isolation, and enrichment in biomedical research. J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm 203K, Saint Louis, MO 63130, meachamjm@wustl.edu), Mohammad Kamali, Yiyi Liu (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, St. Louis, MO), and Mingyang Cui (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA)

We report a highly configurable acoustofluidic technology for separation, isolation, and enrichment of micro-/nanoscale objects, operations that are critical to advancing biological and biomedical research. Decoupling motion of target species from non-target mixture constituents opens opportunities for novel studies involving *in situ* analyses or downstream processing. Acoustic microfluidics provide such precise control by exploiting differences in size and/or particle density and compressibility. Resultant non-contact cell separation minimizes stress on sensitive biological samples. Traditional free-flow acoustophoresis utilizes an ultrasonic standing wave oriented perpendicular to the flow direction so that particle motion is determined by the balance of the acoustophoretic force and viscous drag from the quiescent (in the transverse direction) fluid; the required separation time dictates sample flow rate and channel length, constraining throughput and device size. In contrast, we extend the longitudinal trapping approach to reorient the flow and acoustophoretic separation, eliminating such constraints. Longitudinal standing bulk acoustic wave (LSBAW) devices comprise pillar arrays forming perforated "pseudo walls" that locally augment the pressure field for particle retention at predefined locations. Such structures enable targeted enrichment in an inherently scalable and parallelizable format. Configurations for different applications are discussed, including the possibility for processing throughput exceeding 10 ml/min.

Contributed Papers

2:20

1pPAa5. Measuring acoustofluidic phenomena at the nanoscale using holographic microscopy. Siyang Yu (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA) and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, 9500 Gilman Dr MC0411, MADLab SME344K, La Jolla, CA 92093, jfriend@ucsd.edu)

To understand the mechanisms driving fluid flow behavior in nanofluidics so that they may be used for on-chip biomedical and chemical applications, the fluid's motion itself needs to be observable and measurable, a difficult challenge at the nanoscale, and especially so at the speeds made possible with acoustically driven propulsion. We present a new method for measuring both slow and fast flows in nanofluidics using high-speed digital holographic microscopy. We measure the evaporation-driven flow in 25- and 7-nm tall nanoslit channels, showing that the consequent flow speed is about 15 times slower than open atmospheric evaporation due to the confinement of the nanoslit channel. We also measured the surface acoustic wave-driven flow in the 25-nm channel, showing flow at a speed of 0.12 m/s from acoustic wave propagation at 39.7 MHz interacting with the fluid in the channel. A process to eliminate the many sources of noise to produce these results are provided, showing that—in particular—spatial averaging is

useful to determine the fluid flow and the dewetting of the fluid in the nanoslit channel over time.

2:40

1pPAa6. Dynamics of high frequency bone conducted hearing. Charles Thompson (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 Univ. Ave. Lowell, MA 01854, charles_thompson@uml.edu), Emi Aoki (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), Flore Norceide (Elec. and Comput. eng, UMASS Lowell, Lowell, MA), Vinh T. Tran, Gayathri Boopathy, and Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA)

Explore the potential of bone conduction as a high-frequency alternative to the typical Air conduction (AC) hearing process. High-intensity bone conduction sound in the ultrasonic frequencies above 20 kHz can produce an auditory sensation in humans. Listeners can perceive speech from voice-modulated ultrasonic carrier signals using a head-placed Bone conduction (BC) transducer. In the AC process, airborne acoustic waves enter the outer ear, follow the path through the middle ear, and finally to the sensory organs of hearing. Bone conduction (BC) offers a promising alternative, utilizing direct stimulation of the skull, vibrating the cochlea and its structures. In this presentation, we examine the potential of BC hearing and the dynamic properties that may be used to demodulate the ultrasonic signals and yield a human audible signal.

3:00–3:20 Break

3:20

1pPAa7. Visualizing cavitation bubble transport induced by μs timescale acoustic streaming. Joni Mäkinen (Dept. of Phys., Univ. of Helsinki, Gustaf Hällströmin katu 2, Helsinki 00560, Finland, joni.mk.makinen@helsinki.fi), Jere Hyvönen, Topi Pudas, Tom Sillanpää, Axi Holmström (Dept. of Phys., Univ. of Helsinki, Helsinki, Finland), Nobuki Kudo (Div. of Bioengineering and Bioinformatics, Hokkaido Univ., Sapporo, Japan), Edward Hæggström (Dept. of Phys., Univ. of Helsinki, Helsinki, Finland), Mamoru Hashimoto (Div. of Bioengineering and Bioinformatics, Hokkaido Univ., Sapporo, Japan), and Ari Salmi (Dept. of Phys., Univ. of Helsinki, Helsinki, Finland)

Focused, short-duration and high-amplitude ultrasound pulses are used in applications ranging from medical histotripsy to recent advances in controlled surface erosion for localized material sampling and for e-waste recycling of gold. These applications are enabled by cavitation (and heating to a lesser degree) generated in the focal zone. When cavitation occurs in free water, acoustic streaming also becomes a significant nonlinear effect to consider. Usually, acoustic streaming is treated as a longer timescale, averaged movement of the liquid. However, with intense and short ultrasound pulses, streaming occurs on the same timescale as the pulses. We show high-speed (1 million fps) footage of a focused 4.24-MHz ultrasound field that causes heating, cavitation, and a turbulent jet flow, in water, during a single ultrasonic $\sim$200 cycles ($\sim$50 μs) pulse. A shadowgram produced by the heating enables direct visualization of the turbulent flow. Additionally, cavitation bubbles are generated, and these are transported a significant distance away from their origin by this flow. Understanding and accounting for this single ultrasound pulse timescale flow field is important in applications where cavitation is required to be confined within a well-defined zone, such as localized material removal with focused ultrasound. [Work supported by the Research Council of Finland, Grant Numbers 347459 and 349200.]

3:40

1pPAa8. On the development of an acoustic permeameter to determining the mean Darcy permeability for porous tissue scaffolds. Alessandro Schiavi (Appl. Metrology and Eng., INRIM - National Inst. of Metrological Res., Str. delle cacce 91, Torino, Torino 10135, Italy, a.schiavi@inrim.it), Andrea Prato, Alessio Facello (Appl. Metrology and Eng., INRIM - National Inst. of Metrological Res., Torino, TO, Italy), Giovanni Durando (Sound in Air & Ultrasound Labs., INRiM, Torino, Torino, Italy), and Fabio Saba (INRiM, National Inst. of Metrological Res., Torino, Italy)

The measurement procedure introduced in the ISO Standard 9053-2:2020 allows to determine the airflow resistance in a wide range of porous and permeable materials, by applying a pressure wave-drop method. Flow resistivity is an intrinsic property of porous materials, related to the permeability, as defined by Darcy, and to the hydraulic conductivity. Consequently, once the airflow resistivity of any porous material is known (independently of the fluid flow properties, such as viscosity and density), it is possible to determine other transport properties and characteristics widely used in various application fields. In this work, the development and the application of an accurate sub-infrasonic "acoustic permeameter" (developed and realized at INRIM) are described, and some practical examples, related to the determination of the Darcian permeability for porous scaffold, used in tissue engineering, are illustrated.

4:00

1pPAa9. Direct numerical smulation of Gigahertz acoustic streaming. Virginie Daru (DynFLuid Lab, ENSAM, 151, boulevard de l'hôpital, Paris 75013, France, Virginie.Daru@ensam.eu), Michael Baudoin (IEMN, Université de Lille, Villeneuve d'Ascq, France), and bjarne vincent (INSA Lyon, Lyon, France)

Gigahertz acoustic streaming allows the generation of high-speed micrometric jets that offer myriad possibilities for fluid and particle manipulation. We investigate high-frequency bulk (of Eckart type) streaming using high-order finite difference direct numerical simulations. Such simulations are rare because of their high computing cost, due to the wide spectrum of scales involved in these flows. We solve the Navier–Stokes compressible equations and compute transient and steady acoustic streaming. The simulations cover a wide range of parameters, and by coupling them with a scale analysis, we notably elucidate the scaling law dependency of acoustic streaming on frequency. It is shown that high-speed micrometric jets of several meters per second can only be obtained at high frequencies, due to diffraction limits. The possibilty of in-depth study of several aspects of the flow and limits of the models are also allowed by these simulations: temporal evolution of the streaming flow, characteristics of the source terms (Reynolds stresses) that generate streaming, limits of classical asymptotic developments. Finally, we quantify the maximum time required to reach the maximum jet speed as the frequency increases. This reveals that accelerations within the Mega-g range occur, opening new possibilities for generating ultra-short, high-speed microjets.

4:20

1pPAa10. Particle deflection in a macroscale ultrasonic angled wave device. nicholas rivet (Mech. Eng., Western New England Univ., 1215 wilbraham Rd., Springfield, MA 01119, nicholas.rivet@wne.edu), Bart Lipkens (Pharmaceutical and Administrative Sci., Western New England Univ., Springfield, MA), and Walter Presz (Mech. Eng., Western New England Univ., Springfield, MA)

Macroscale bulk ultrasonic standing waves oriented at an angle relative to a fluid velocity field offer a novel method for sensitive particle separation. We have previously reported on the theoretical development of such a system [Proceedings of Meetings on Acoustics Vol. 30, 045004 (2017), https://doi.org/10.1121/2.0000652]. Particle deflection is uniquely defined by two parameters, i.e., the non-dimensional ratio of the acoustic radiation force to the fluid drag force exerted on the particle and the angle of the standing wave with respect to the fluid velocity. Here, we highlight the development and testing of a novel30 deg acoustofluidic angled wave device with a 0.5″ × 0.5″ main fluid flow channel. A 0.5″ × 0.5″ PZT-8 3 MHz air-backed transducer was fabricated. Acoustic resonance quality factors of 3000 were measured using electrical impedance measurements. The particle suspension enters the fluid channel through a 4 × 0.4 mm^2 inlet centrally located in the square channel. The particle inlet is surrounded by a buffer flow. Testing was performed with 3, 6, and 10 μm polystyrene particles at a total flow rate of 31 ml/min, corresponding to a peak velocity of 5 mm/s. Successful deflection of all particles were achieved at electrical powers in the range of 1–5 W.

4:40

1pPAa11. Modular platform for reconfigurable bulk acoustic wave-based acoustofluidics. Mohammad Kamali (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., St. Louis, MO 63130, a.kamalidolatabadi@wustl.edu) and J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, Saint Louis, MO)

Acoustofluidics integrate acoustic waves and microfluidics to enable precise, label-free manipulation of fluids and particles. Despite vast potential in biological and biomedical applications, commercialization of these technologies is hindered by system complexity, limited reusability, and associated high costs. Here, we introduce a modular bulk acoustic wave-based platform featuring a detachable/reusable piezoelectric transducer and reversible fluidic interconnects. The platform accommodates rigid microfluidic chips (e.g., silicon and/or glass) for highly efficient object manipulation. Side actuation enhances acoustic energy transfer compared to more typical top actuation. Using ultrasound gel as a temporary couplant, the system achieves an acoustic energy density comparable to or exceeding that of the same chip with a permanently bonded, top-epoxied transducer. Practical benefits of the modular design include the capability for rapid transducer and coupling layer exchange/reuse to support diverse applications in biomedical research. Additionally, the fluidic interconnects maintain leak-free operation under high pressure (>10 bars) and flow rates exceeding 2 ml/min. Platform implementations with various microfluidic chips for separation, enrichment, and particle trapping/agglomeration are discussed. By addressing limitations of conventional systems, our platform offers an efficient and reconfigurable approach to facilitate wider adoption of acoustofluidic technologies.

MONDAY AFTERNOON, 19 MAY 2025 BALCONY J, 1:00 P.M. TO 4:50 P.M.

Session 1pPAb

Physical Acoustics and Education in Acoustics: It's Not Physics

Steven L. Garrett, Cochair
1736 Lowell Street, Seaside, CA 93955

Roger M. Waxler, Cochair
Univ. of Mississippi, P.O. Box 1848, University, MS 38677

Chair's Introduction—1:00

Invited Papers

1:05

1pPAb1. Acoustics was physics in the United States—Until it wasn't. Andrew Zangwill (Phys., Georgia Inst. of Technol., 3585 Sunderland Circle, Atlanta, GA 30319, az2@gatech.edu)

This talk surveys the historical fortunes of acoustics as a research topic in American physics departments over the course of the 20th century. Before World War II, nearly a quarter of all PhD granting physics departments employed at least one or two professors who conducted research in musical, architectural, physical, ultrasonic, or electroacoustics. Nevertheless, as early as 1942, Bruce Lindsay felt aggrieved enough to complain that "acoustics has been the stepchild of physics for many years in our universities." After World War II, a few physics departments took advantage of Navy largesse to build acoustics groups large enough to sustain themselves for some time. Most other physics departments did the opposite. Retiring acousticians were replaced by nuclear or solid state physicists and acoustics instruction was moved out of departments of physics and into departments of applied physics or electrical or mechanical engineering. Today, less than a handful of physics departments maintain a critical mass of professors engaged in acoustics research. A comparison with optics research provides one way to understand the decllining fortunes of acoustics in US physics departments.

1:50

1pPAb2. "That IS physics." Acoustics and its existence within Brigham Young University's Department of Physics and Astronomy. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu)

In the April 2024 American Physical Society newsletter, "APS News," Prof. A. Zangwill published an opinion piece describing the boundaries of "physics" and historical debates over where they should be drawn. The title? "That's not physics." The prime example of a discipline that used to be considered physics but is no longer? Acoustics. As an acoustician who somehow became Chair of the Department of Physics and Astronomy at Brigham Young University (BYU)—a department that ranked seventh in the nation for undergraduate degrees awarded in 2022–2023—reading this article naturally resulted in an existential crisis of sorts for me and ruined a particularly pleasant afternoon. As a mentor, professor, acoustician, and now Chair, I have thought hard about the role of acoustics in a physics and astronomy department. In this presentation, I discuss the history of acoustics within BYU Physics, its growth that has opposed national trends, and its present trajectory within the ever-changing definition of "physics." While acknowledging the dangers of both overfitting and extrapolating, I offer a perspective on acoustics within an undergraduate physics curriculum.

2:10

1pPAb3. Acoustics can be physics. Roger M. Waxler (Univ. of Mississippi, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu)

A personal discussion will be presented addressing the question of how being trained, and then working as a physicist influences the way one approaches and studies problems in acoustics. With absolutely no claims to objectivity, it will be suggested that what distinguishes a physicist's approach is that what motivates the physicist to research is a desire to understand and explain observed physical phenomena in a way that is consistent with our understanding of the larger environment in which the phenomena occur. These motivations, and the standards that result for acceptance of an explanation, will be discussed. Examples from the author's experience and research in atmospheric acoustics will be used to illustrate. Comparisons to and critiques of the state of academic physics over the past several decades will be presented.

2:30–2:50 Break

2:50

1pPAb4. Physics Dept. at U.S. Naval Academy still shines a hopeful light into the darkness in acoustics undergraduate research and education. Murray S. Korman (Phys., U. S. Naval Acad., Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21403, mskorman167@gmail.com)

The art of teaching physics concepts, in Underwater Acoustics and Sonar to physics majors, oceanographers, engineers, and general science students, has been the corner stone that has allowed midshipman to pursue acoustics research projects, present papers, and perform demos at ASA or other conferences. The Physics Dept. teaches all sophomores (STEM or non-STEM) fundamentals of physics, while a dwindling few acousticians also teach Acoustics to senior physics majors, along with Sonar. Acoustics emphasizes mathematical methods of physics with connections to quantum mechanics. Words like "now I understand what an eigen-function is and what an eigen-values is," are very encouraging. Sonar emphasizes "physics of sound in the sea" and includes interference, refraction, method of images, and connections to normal modes. Demos by the famous UCLA group, among others, have motivated demo presentations by USNA students at ASA meetings—for over 40 years. Physics building blocks and demos evolved into experimental research: sound speed to sound scattering in bubble clouds, nonlinear scattering of sound by turbulence, PIV flow visualization to Lighthill turbulent streaming jet flow to crossed jet flow, light absorption to photo-acoustics, and resonance and hysteresis to the nonlinear mesoscopic elastic soil—elastic plate oscillator to acoustic landmine detection.

3:10

1pPAb5. Acoustics in an undergraduate physics department. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Dept. of Phys., Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

Physics departments that provide only undergraduate degrees can be a good fit for faculty with acoustics expertise, given that such departments value research experiences for undergraduate students and Acoustics is a branch of applied physics that is amenable to undergraduate level research projects with modest budgets. The caveat is that faculty at such institutions need be at least as passionate about teaching as doing research. Rather than leading a team of PhD students and postdocs, they reap significant rewards in training young students be good physicists and to help some find their passion in acoustics. Motivated faculty can also find ways to incorporate acoustics into the curriculum, as well as into outreach and student club activities. This presentation will highlight experiences from a nearly 30-year career at Central Washington University that demonstrate how acoustics can become an integral part of the undergraduate physics experience.

3:30

1pPAb6. Acoustics provides the primary route for the study of emergent phenomena. Seth Putterman (Phys., ucla, 475 Portola Plaza, Phys. Dept, Los Angeles, CA 90095, puherman@ritva.physics.ucla.edu)

Sound easily propagates through a gas where the thermal noise is nine orders of magnitude larger than the velocity amplitude. Acoustics then answers the key question of emergent theory: how many macroscopic variables are a complete set? It has never been proven theoretically, but everyone who studies turbulence starts with five-variable hydrodynamics as compared to first principles kinetic theory. Uhlenbeck said "the frontiers of physics are all around us" and "acoustics is the royal road." Acoustics is a leading route to nonlinear, multi-scale phenomena. The passage of sound through a fluid can stimulate the expansion and collapse of a bubble which concentrates the acoustic energy density by 12 orders of magnitude to create a light-emitting strongly coupled plasma. Our recent study of the dispersion law for this plasma has led to a unifying theory of SCP that also has five variables as a complete set but a nonlocal equation of state. Ferroelectric crystals can have a spontaneous polarization of 100 kV; strong enough to cause nuclear fusion in deuterium gas. Rudnick used hydrodynamics to discover a quantum phase transition in angstrom-thick films. We have used acoustic radiation pressure to create the only ground-based model for the study of spherical convection.

3:50–4:50 Panel Discussion

Session 1pPP

Psychological and Physiological Acoustics: The Hartmann Effect: Bill Hartmann's Influence on Monaural and Binaural Hearing Research

Matthew J. Goupell, Cochair
Hearing and Speech Sciences, University of Maryland-College Park, 7251 Preinkert Drive, 0141 Lefrak Hall, College Park, MD 20742

Virginia Best, Cochair
Speech, Language and Hearing Sciences, Boston University, 635 Commonwealth Ave, Boston, MA 02215

Chair's Introduction—12:55

Invited Papers

1:00

1pPP1. Bill Hartmann's influence on computational psychoacoustics: Extending and correcting the Woodworth model. Neil Aaronson (Phys., Stockton Univ., Stockton University - NAMS, 101 Vera King Farris Dr., Galloway, NJ 08205-9441, neil.aaronson@stockton.edu)

Bill Hartmann's approach to acoustics through the lens of physics led to novel advancements in acoustical models relevant especially to psychoacoustical applications. While the method of computing interaural time and level differences using an exact diffraction model of a rigid, spherical head is well established, utilizing it in cases where high precision is required, especially at high frequencies, is computationally taxing. The Woodworth formula computes the interaural time difference for a rigid, spherical head, assuming plane-wave incidence (i.e. large source distance) and antipodal ears, based on a computationally efficient ray-tracing acoustical model with a creeping-wave solution. It is only applicable as a high-frequency limit (wavelength much smaller than the head radius), viable for adult human heads above approximately 4 kHz. It further assumes that there is only one creeping-wave path around the head—that of shortest distance—even when paths around the front and back of the head are almost the same length. This approximation is compared to an exact rigid spherical head model, extended to allow for arbitrary ear locations and non-distant sources, and corrected to include both creeping waves while maintaining low computational complexity [Aaronson and Hartmann, J. Acoust. Soc. Am. 135, 817–823 (2013)].

1:15

1pPP2. Non-monotonic binaural cues: Recent insights from the "Acoustical Bright Spot". Paul G. Mayo (Hearing and Speech Sci., Univ. of Maryland-College Park, 0100 LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, paulmayo@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Horizontal-plane sound localization relies on binaural cues, or interaural time and level differences (ITDs and ILDs), created by acoustic interactions with the head. ILDs are known to be non-monotonic functions of azimuth from diffraction of sound around the head (the "Acoustical Bright Spot"), resulting in the predictable mis-localization of tones in the free field by acoustic-hearing listeners [Macaulay *et al.*, J. Acoust. Soc. Am. **127**, 1440–1449 (2010)]. This talk reviews Fresnel diffraction and the Acoustical Bright Spot described by Hartmann and colleagues, and summarizes recent findings and experiments inspired by their work. Recent measurements and modeling studies revealed these non-monotonicities are not limited to just ILDs and have complex interactions with properties of hearing devices such as microphone placement and directionality.

1:30

1pPP3. Transaural synthesis technique and studies in spectral cues on the sagittal plane. Peter Xinya Zhang (Audio Arts and Acoust., Columbia College Chicago, 600 S. Michigan Ave., Chicago, IL 60605, peter@peterzhang.net)

Based on the cross-talk cancellation method, a transaural synthesis technique (VRX) was developed, simulating real sources at an arbitrary location using two synthesis loudspeakers and two probe microphones placed in the listeners' ear canals. The simulation is accurate up to 16 kHz. This method allows the experimenters to modify the presented spectra in the listeners' ears in great detail while enabling the listeners to use their pinna cues during an experiment. The VRX experiments with broadband complex tones show that listeners used a variety of strategies in localizing sources in front and back, comparing cues in multiple frequency bands; the role of spectral dips appeared to be more important than peaks; for interaural time difference less than 200 μs, listeners could correctly localize on front/back, independent of the azimuth cues [Zhang and Hartmann, Hear. Res. **260**, 30–46 (2010)]. The current talk discusses the findings by Hartmann and colleagues using the VRX technique and their impact on our current understanding of sound localization on the sagittal plane.

1:45

1pPP4. Cross-frequency coherence of ITDs ("straightness") minimizes the effects of binaural cue ambiguity in stereo perception. Yi Zhou (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave. Coor 3470, Tempe, AZ 85287, yizhou@asu.edu) and Colton Clayton (GrandValley State Univ., Tempe, AZ)

Rayleigh's original Duplex Theory posits that sound-source localization relies on different cues from low- and high-frequency sounds: low frequencies provide interaural time differences (ITDs), which often dominate localization, while high frequencies offer spatial information through interaural level differences (ILDs). An updated perspective on the Duplex Theory emerged from perceptual studies investigating binaural hearing by pitting ITDs against ILDs [Hartmann *et al*., J. Acoust. Soc. Am. **139**, 968–985 (2016)]. Hartmann and colleagues examined free-field localization of sine tones at 250, 500, and 750 Hz, revealing two key findings: (1) slipped-cycle ITDs at higher frequencies (e.g., 750 Hz) can lead to significant left-right confusions, and (2) naturally occurring ILDs with the same sign as ITDs can resolve these confusions. Building on this work, we applied stereophonic techniques to investigate free-field localization of noise stimuli with conflicting ITDs and ILDs across different frequency ranges. Our findings indicate that ILDs can modulate the effective ITDs in both low-pass and high-pass noise stimuli with a cutoff frequency of 1500 Hz. However, a coherent ITD across frequencies ("straightness") is a stronger grouping cue than same-sign ITD and ILDs in reducing left-right confusions.

2:00

1pPP5. Physiologically plausible and implausible interaural time difference distributions and the dominance of the 700-Hz frequency region. Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., 0141 Lefrak Hall, College Park, MD 20742, goupell@umd.edu), Anhelina Bilokon (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), Mathieu Lavandier (LTDS, ENTPE, Lyon, France), and H. Steven Colburn (Biomedical Eng., Boston Univ., Boston, MA)

Low-frequency (<1500 Hz) interaural time differences (ITDs) are the primary horizontal-plane sound localization cue for humans. Because of the frequency dispersion occurring when sound travels around the head, the physical ITDs experienced by humans increase with decreasing frequency in this region for a given azimuth. The frequency dependency from the head dispersion appears to have little functional consequence to sound localization; it does not appear to lead to changes in source width and produces modest effects on lateralization that can be explained by the ITD of the sound in the auditory filter near 700 Hz [Constan and Hartmann, J. Acoust. Soc. Am. **114**., 998–1008 (2003)]. This talk discusses experiments by Hartmann and colleagues, and recent experiments inspired by them, that use physiologically plausible and implausible ITD distributions across frequency to probe across-frequency processing in human sound-localization. The results of these studies suggest that robust broadband sound localization is primarily dominated by ITD information in the frequency region around 700 Hz.

2:15

1pPP6. Effects of interaural decorrelation on psychophysical sensitivity to interaural level difference cues—Reflections on the "level-meter model" of Hartmann and Constan (2002). Daniel J. Tollin (Dept. of Physiol. & Biophys., Univ. of Colorado School of Medicine, 12800 E 19th Ave., Aurora, CO 80045, Daniel.Tollin@CUAnschutz.edu) and Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

One of the cues to sound location is the interaural level difference (ILD). Use of ILDs implies an energy-like, thus time-integrated, measure of level at each ear independent of stimulus characteristics, like interaural correlation. Hartmann and Constan (2002) dubbed this the "level-meter model." They tested model assumptions by measuring human ILD thresholds for broadband noise stimuli that were interaurally correlated, anticorrelated or uncorrelated. An additional test measured monaural level discrimination thresholds. According to the model, all four thresholds should be the same. While thresholds varied by <0.5 dB across tasks, slight but reproducible effects of decorrelation were evident. Intrigued by these results, Brown and Tollin (2016, 2021) subsequently measured behavioral ILD sensitivity (in humans) and neural ILD sensitivity (in chinchillas) using a variety of narrowband stimuli. In aggregate, results were consistent with an amended version of the level-meter model in which ILDs are calculated within several-millisecond long windows of excitatory–inhibitory integration, sufficient to avoid effects of decorrelation for most, but not all, sounds. Results have implications for localization of sounds in everyday environments and with hearing devices, where interaural decoherence can compromise the use of ongoing interaural time difference cues but has little impact on the use of ILDs.

2:30

1pPP7. Psychophysical measures using low-noise noise. Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, eozmeral@usf.edu)

The term low-noise noise describes a non-Gaussian, random-phase signal with minimal fluctuations in the envelope. Also referred to as low-fluctuation noise and originally adopted for psychophysical investigation by Hartmann and Plumplin [J. Acoust. Soc. Am. **83**, 2277–2289 (1988)], this unique signal has subsequently been used to uncover several basic mechanisms of auditory perception, including in studies on gap detection, the binaural masking level difference, loudness judgment, forward masking, and speech masking. Though originally computationally expensive for real-time generation, narrow-band low-noise noise is now feasible for psychophysical experiments using any of several time-efficient methods. We discuss the historical impact of this useful signal for understanding envelope cues in natural waveforms, and we provide a potential vision for future studies.

2:45

1pPP8. The effect of the Negative Level Effect. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., EC 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Hartmann and Rakerd [JASA 94, 2083–2092 (1993)] measured listeners' ability to identify the source location (front, overhead, or behind) of short (25-μs clicks) and long (880-ms noise bursts) broadband stimuli as a function of intensity. Error rates increased markedly with intensity for clicks—the *Negative Level Effect* (NLE)—but not for noise bursts. In addition to direct practical applications (e.g., stimulus choice in auditory displays), the result of this simple experiment has inspired a sub-field of its own including psychophysical, physiological, and computational modeling studies. The differing effect of level on vertical-plane localization of short- and long-duration stimuli has been replicated behaviorally in humans and cats and for additional stimulus types. Although degradation of spectral cue representations via saturation of auditory-nerve rate profiles provides the most straightforward explanation of the NLE, attempts to account for NLE-release for long-duration stimuli (via temporal integration, high-threshold fibers, lateral inhibition, or efferent effects) have so far proven inconclusive. Those studies have, however, revealed further complexities of spectral cue processing also requiring explanation. This presentation will survey the research inspired by Hartmann and Rakerd's experiment and describe why the mechanism of NLE-release observed at long durations is still an open question.

3:00–3:20 Break

3:20

1pPP9. Complex pitch perception with auditory model-based stimuli in humans. Jackson E. Graves (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, 29 rue d'Ulm, Paris 75005, jgra@med.umich.edu), Daniel R. Guest (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY), Tess M. Starr, and Anahita H. Mehta (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI)

Pitch can be evoked by a single sinusoidal component, a harmonic complex tone, or even from a mixture of multiple harmonic complex tones. An accurate model of pitch perception should account for human perception of all these stimuli. This line of inquiry includes research by Hartmann and colleagues testing pitch models for mistuned harmonics [Lin and Hartmann, J. Acoust. Soc. Am. **103**, 2608–2617 (1998)]. Building on this body of research, our study tests pitch models for multiple simultaneous pitches using "rate-place metamers," synthetic stimuli with rate-place representations matching those of pitch-evoking stimuli but altered temporal representations. We measured F0 difference limens (F0DLs) in 20 normal-hearing human listeners for original stimuli (ORIG) and their rate-place metamers (META), across a range of F0s and spectral regions, for single harmonic complexes and triads. We also measured perception of all three pitches in a triad, with an objective major-vs.-minor task and a subjective expectation rating task. We observed similar patterns of behavioral results for ORIG and META across all tasks, broadly supporting a rate-place view of multiple pitch perception. However, some observed differences between ORIG and META cannot be well explained by the rate-place model. [Work supported by NIH, Grant R00DC017472 (AHM)]

3:35

1pPP10. Studying sound externalization in the context of hearing aids. Virginia Best (Dept. Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave. Boston, MA 02215, ginbest@bu.edu), Pinar Ertürk, Tobias Greif, and Elin Roverud (Dept. Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

Sound externalization is a complex perceptual phenomenon that is difficult to study experimentally. In their fascinating paper on this topic, Hartmann and Wittenberg [J. Acoust. Soc. Am. 99, 3678–3688 (1996)] used a headphone synthesis technique to create virtual sound images that were indistinguishable from real sound images. This allowed them to systematically vary the signals delivered to the ear canals and examine the impact of these variations on the "convincingness" of the virtual sound images. Their results suggested that externalization depends on interaural time and level differences, on a plausible relationship between the two, and on monaural spectral cues. Hearing aids disrupt the natural sound path to the ear canals, and it is easy to appreciate how they may distort the cues that support externalization. This talk will describe an ongoing series of experiments aimed at characterizing in detail the effects of hearing aids on externalization. Results suggest that breakdowns of externalization with hearing aids are (1) related to microphone position and openness of the ear canal; (2) highly dependent on the source stimulus; and (3) partly resolved by head movements. The talk will also consider the consequences of poor externalization for speech understanding in multitalker mixtures.

3:50

1pPP11. The Hartmann Effect in rooms. Brad Rakerd (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, rakerd@msu.edu)

In 1983, Bill Hartmann published an article in *JASA* reporting on a study of azimuthal sound localization that he had conducted in a variable acoustics concert hall [JASA, **74**(5), 1380–1391]. The findings of that study enhanced our understanding of the precedence effect, and of a number of other factors that affect our ability to localize sounds indoors. Those findings also motivated a series of subsequent studies of sound localization in rooms that we conducted at Michigan State University over the years, and that bear the unmistakable hallmarks of Bill's creativity and boundless scholarly energy. This presentation will take a retrospective look at the MSU projects, with particular emphasis on instances of The Hartmann Effect as it operates in rooms.

4:05

1pPP12. The plausibility hypothesis. Pavel Zahorik (Univ. of Central Florida, School of Commun. Sci. and Disord., Orlando, FL 32816, pavel.zahorik@ucf.edu) and Gregory M. Ellis (Audiol. and Speech Pathol., Walter Reed National Medical Military Ctr., Bethesda, MD)

The plausibility hypothesis was originally introduced to explain human sound localization behavior in situations with a single sound-reflecting surface capable of producing abnormally large interaural time cues. The hypothesis states that "interaural time cues are weighted by listeners according to their plausibility" and that "listeners assess plausibility as though all sounds are direct" [Rakerd and Hartmann, J. Acoust. Soc. Am. **78**, 524–533 (1985)]. Thus, the hypothesis explained why interaural time cues contributed to perceived source direction in certain situations but not in other situations with implausible cue values. A key aspect of the plausibility hypothesis is its acknowledgment of the role of expectation in governing perception. Against a backdrop of contemporary auditory research that emphasized the role of stimulus-driven bottom-up processing, appeal to the top-down aspects of expectation was groundbreaking. Hartmann and Rakerd went on to successfully apply the plausibility hypothesis to other aspects of binaural and spatial hearing, such as understanding the Franssen Illusion. Here we discuss additional high-level auditory phenomena, including distance perception and perceptual de-reverberation, through the lens of the plausibility hypothesis.

4:20

1pPP13. Bill Hartmann's contribution to understanding and modeling the perception of reflections in real rooms. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

In my early career, Bill Hartmann's exceptional four-paper JASA series *"Sound Localization in Rooms"* caught my eye when I became interested in modeling the precedence effect. The latter three papers and a subsequent one in 2010 were co-authored by Brad Rakerd. The series investigated the precedence effect in various rooms as a function of signal characteristics, bandwidth, and onset duration and provided a holistic view of a complex problem. As a student, I started to appreciate Bill's willingness to walk beside the beaten path of research, and I remember the lively discussions with the audience that helped me to learn to think more critically. Hartmann and Rakerd's research led to the formation of their *"plausibility hypothesis,"* stating that listeners can discount implausible cues. This hypothesis became a central point in my efforts to model the precedence effect, resulting in different attempts to model the effect based on onsets and ongoing signal components. These models include a modified Lindemann algorithm and two models based on multilayer auto/cross-correlation structures. The presentation focuses on how these and other models can be applied to the studies by Hartmann and Rakerd.

4:35

1pPP14. Interaural time difference sensitivity as a function of interaural coherence in the aging auditory system. Monica L. Folkerts (Commun. Technologies Res. Ctr., Univ. of Central Florida, 3280 Progress Dr., Ste 100, Orlando, FL 32826, monica.folkerts@ucf.edu), David A. Eddins (Commun. Technologies Res. Ctr., Univ. of Central Florida, Orlando, FL), and John H. Grose (Dept. of Otolaryngology–Head & Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Interaural time difference (ITD) cues are the dominant cue for sound localization in the horizontal plane. Encoding of ITD is postulated to occur via an internal binaural cross-correlation function, where the peak correlation serves as the interaural coherence (IAC) and the delay in which the peak occurs serves as the ITD. Sound sources in anechoic chambers have IAC values at or near 1. Sound sources in reverberant rooms have a wide range of frequency-dependent IAC values between 1 and 0, where ITD sensitivity decreases as IAC decreases toward 0 [Rakerd and Hartmann, J. Acoust. Soc. Am. **28**, 3052–3063 (2010)]. The current work builds upon the relationships described by Rakerd and Hartmann to examine the degraded binaural temporal processing of the aging auditory system in complex (reverberant and noise-filled) rooms. Behavioral ITD thresholds were measured as a function of IAC for younger and older listeners with clinically normal hearing. Results indicate that at higher IAC values, older listeners have elevated ITD thresholds. However, at lower IAC values, ITD thresholds between listener groups converge. Following Rakerd and Hartmann, binaural analyses were made in a reverberant room, with the addition of competing noise to simulate listening scenarios older listeners often find difficult.

4:50

1pPP15. Maturation of the Precedence Effect: Honoring Bill Hartmann's wisdom. Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin, Waisman Ctr. 521, 1500 Highland Ave., Madison, WI 53705, litovsky@waisman.wisc.edu), Roya Abdi, Kumar Anshu, and Shelly Godar (Waisman Ctr., Univ. of Wisconsin, Madison, WI)

The precedence effect (PE) is a perceptual phenomenon whereby the first-arriving sound is assigned greater perceptual weight than delayed reflections. The PE matures during childhood, when establishment of accurate spatial hearing skills is paramount to a child's ability to navigate in everyday, reverberant environments. To simplify natural reverberant situations, studies on the PE use lead-lag stimuli with delays that occur naturally in rooms. We investigated spatial hearing in nearly 80 participants aged 5–25 years, using brief trains of noise bursts. On PE conditions, the lead was left or right of midline at varying angles with the lag fixed at midline. Single source stimuli served as control. Results showed significant age effects in accuracy for PE and single conditions; a more protracted maturation on PE conditions indicates that perceptual dominance of the lead (strength of the PE) matures into adolescence. In other studies, we harnessed paradigms that involved sound localization in a multi-loudspeaker space to investigate children's ability to ignore information from the lag enough to be able to localize the lead. In older children, strong effects of lead-lag delay impact localization. Importantly, the task is critical to engaging young children, and cognitive factors are likely to also play a role. [Work supported by NIH-NIDCD Grants R01DC020355 and R01DC019511 to R.Y. Litovsky.]

5:05

1pPP16. Psychoacoustics and me. William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu)

This talk is an expression of gratitude to the speakers today and to the entire field of hearing research. I did not start out doing hearing research. I received the D. Phil. degree in 1965 for research in condensed matter theoretical physics, and I worked in that field for a further ten years. In 1976, I gingerly put my foot into the psychoacoustical waters, and I was hooked. Certainly, a primary reason for my conversion to the study of mammalian hearing was the people who work in that area. The fields of hearing research and acoustics, and the Acoustical Society of America in particular have been both exciting and extremely friendly. I cannot imagine a more enjoyable career. Thank you all.

MONDAY AFTERNOON, 19 MAY 2025 BALCONY M, 1:00 P.M. TO 4:40 P.M.

Session 1pSA

Structural Acoustics and Vibration: General Topics in Structural Acoustics

Anthony L. Bonomo, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Allison M. King, Cochair
Mechanical Engineering, University of Michigan, 2370 GG Brown, Ann Arbor, MI 48109

Contributed Papers

1:00

1pSA1. Acoustic source localization on a plate suspended at an air–water interface. Allison M. King (Mech. Eng., Univ. of Michigan, 2370 G.G. Brown Lab., 2350 Hayward St., Ann Arbor, MI 48109, kingalli@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Acoustic waves can be ideal for remote sensing and structural health monitoring because they carry source information and can be measured without contact. However, traditional time-of-flight array methods for source localization are ill-suited for structural engineering applications. Specifically, for a plate suspended at an air–water interface, the coupling between the vibrating structure and the surrounding mediums complicates localization efforts. Thus, source localization experiments were conducted using Matched Field Processing (MFP) for a 0.91-m diameter round aluminum plate suspended at an air–water interface and excited by the impact of a 1.3-cm diameter stainless-steel ball bearing dropped from 11.4-cm. A remote linear seven-microphone array placed 10.5-cm above the plate in the air and a remote linear seven-hydrophone array placed 13.5-cm below the plate in the water measured the sound radiated by the 0.64-cm-thick plate at frequencies up to 20 kHz. MFP array signal processing localization techniques were used along with a physics-based finite element acoustic model to localize the impact excitation on the structure. Source localization results using each set of acoustic pressure measurements independently are compared to results using both sets of acoustic pressure data. Localization success rates in a noisy environment are also presented. [Work sponsored by a SMART Scholarship.]

1:20

1pSA2. Nondestructive acoustic mapping of cold sintered ZnO: Effects of density, porosity, and transient phase. Silas Wieland (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci. Bldg., University Park, PA 16802, srw5795@psu.edu), Haley N. Jones (Eng. Sci. and Mech., Penn State Univ., State College, PA), and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

The cold sintering process (CSP) is a low-temperature densification method for ceramics, metals, and composites. CSP enables densification at 70–500°C, halving processing times compared to standard firing temperatures greater than 1000°C. A transient liquid phase drives densification by facilitating mass transport, influencing final density and microstructure. This study explores how transient phase affects longitudinal wave speed and attenuation of CSP ZnO. A single-element 20-MHz transducer in a water immersion pulse-echo system provides thickness-averaged spatial maps of longitudinal wave speed and attenuation. Sample sets include a range of densities (∼80%–95% of theoretical density) as well as heat-treated conditions to evaluate the impact of residual amorphous phases on elastic properties. Scanning Electron Microscopy characterizes microstructural features, complementing acoustic data. Comparisons with Hashin–Shtrikman bounds assess alignment between measured wave speeds, theoretical elastic moduli, and microstructural effects. Attenuation data further highlight porosity and phase differences across samples. These comparisons aid tuning of CSP parameters and solvent selection. Nondestructive acoustic analysis provides insights into CSP ZnO microstructure and elastic properties. Findings may guide future optimization of transient phase chemistry and processing variables in functional ceramics, leveraging acoustic techniques for rapid feedback on materials prepared through innovative processes.

1:40

1pSA3. An acoustic emission test for the detection of anomalies in hydroelectric generator stator bars and coils. Kevin Venne (Hydro-Québec, 1800 Bd Lionel-Boulet, Varennes, QC J3X 1S1, Canada, venne.kevin2@hydroquebec.com), Mathieu Kirouac, Hélène Provencher, and Mathieu Soares (Hydro-Québec, Varennes, QC, Canada)

To meet the ever-increasing demand for electricity, Hydro-Quebec is seeking to simultaneously increase the power of its generating stations while improving its service quality. Hydro-Quebec's research center has been tasked to investigate innovative methods to meet the aforementioned goals. The current study presents an acoustic emissions test capable of detecting delamination and cavities found between the insulation layers of stator bars and coils. The developed method was corroborated with data obtained using impact time series analysis and frequency analysis of structure borne vibrations. The dataset utilized consisted of 10 stator coils with varying synthetic defects. These variations included cavities of different sizes (1.2, 1.0, and 0.8 mm) and delaminations of varying depth both at different positions on along the straight portion of the stator bar. Furthermore, the dataset consisted of measurements taken before and after electrical aging, and after electrical failure for a total of 120 measurements. Such measurements demonstrated the acoustic method's ability to detect and quantify the degradation induced by high voltage stresses.

2:00

1pSA4. Nonlinear vibration characterization of the synthetic jet actuator to reduce noise levels. Jomar C. Morales (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Rm. 270, Montreal, QC H3A 0C3, Canada, christopher.morales@mcgill.ca)

Synthetic jet actuators (SJAs) generate unwanted noise levels that can be detrimental to human well-being if exposed to these noise levels over long periods of time. SJAs produce noise through the pressure variations from the jet and through the vibration of the piezoelectric actuator. Laser Doppler velocimetry has shown that harmonic excitation causes vibration response at overtone frequencies. The reasoning for excitation at overtone frequencies is not well understood. Understanding of vibrational nonlinearities can lead to developing methods or selecting design parameters to minimize noise levels while maximizing jet performance. The vibration of the SJA is characterized at three separate levels: the piezoelectric disk, the single membrane, and the full SJA structure. The piezoelectric disk is subject to hysteresis in the piezoelectric effect and electrostriction. The single membrane is a sandwich structure consisting of two piezoelectric disks with a laminate material as the sandwich layer, which may contribute to additional nonlinearities due to material properties. Finally, the complete SJA structure has coupling between the structural and acoustical system. Characterization of the sources of nonlinear behavior is to be used in establishing a more accurate numerical simulation, which can be used to optimize design parameters such as dimensions and voltage input to minimize noise levels while maximizing jet performance.

2:20

1pSA5. Experimental analysis and vibration mitigation using anti-vibration pads in rotating machinery: A study on defective shaft dynamics. Gaurav Sharma (Mech. Eng., Defence Inst. of Adv. Technol., C-116 Points Hostel Defence Inst. of Adv. Technol., Pune, Maharashtra 411025, India, sharmag603@gmail.com), Anand Rengaraj, and Adepu Kumaraswamy (Mech. Eng., Defence Inst. of Adv. Technol., Pune, India)

Industrial machinery, vehicles, and equipment are often subjected to noise, shocks, and vibrations, which can reduce efficiency, lifespan, and safety while increasing maintenance costs. This study presents a comprehensive approach to monitoring and mitigating mechanical vibrations using a custom-designed vibration simulator equipped with piezoelectric accelerometers, PLC, and HMI systems. The research investigates the dynamic behavior of various defective shafts—misaligned, imbalanced, eccentric, cocked, and defective bearings—under variable operating speeds (300–500 RPM) with and without the application of anti-vibration pads (AVPs). The experimental setup allows for precise measurement of RMS accelerations across multiple points, capturing vertical, horizontal, and base vibrations. Fourier Transform and Fast Fourier Transform (FFT) methodologies are employed for frequency analysis, and results are validated through MATLAB-based graphical visualizations. Anti-vibration pads, manufactured with 60 Shore A and 65 Shore A rubber, demonstrate significant vibration attenuation, with reductions exceeding 75% in several scenarios. This work highlights the critical role of AVPs in vibration control and provides insights into their effective selection and deployment for industrial applications.

2:40–3:00 Break

3:00

1pSA6. A phase-based approach to speed of sound estimation in DAS. Peyman Moradi (SoundFalls LLC, 1600 Springwoods Plaza Dr., Apt 616, Spring, TX 77389, peyman.moradi@soundfalls.com)

This study presents an approach for estimating the Speed of Sound (SoS) in fluids flowing through pipes equipped with Distributed Acoustic Sensing (DAS) arrays. The method integrates phase-based coherence analysis, robust time windowing, and targeted filtering in both frequency and spatial domains to reduce the impact of noise on solution confidence. Collective phase correlations are calculated across selected DAS signal pairs, with key frequency bands isolated to enhance measurement accuracy. Results are displayed as likelihood plots for both positive and negative SoS over various ranges, providing insights into fluid type, flow velocity, and flow regime within the pipe. Our findings demonstrate a promising reduction in SoS measurement uncertainty compared to both 2-D FFT FK and cross-correlation approaches over an equivalent spatial aperture, highlighting the method's potential for more accurate flow characterization in DAS-equipped systems.

3:20

1pSA7. Performance assessment of ultra-low noise solid-state accelerometers for vibration-based diagnostics in thermal process equipment. Eli Hughes (Wavenumber LLC, 2180 N Oak Ln., State College, PA 16803, ehughes@wavenumber.net) and Stephen Wells (Boeing, Huntsville, AL)

The maturation of MEMS (micro-electromechanical systems) technology for acoustic and vibration sensing has been driven by mass-market applications, particularly in mobile communications and motion-controlled interfaces. Silicon-based MEMS accelerometers have historically been optimized for low-frequency and quasi-static (DC) applications, exhibiting higher noise spectral density characteristics compared to piezoelectric sensors. However, MEMS technology offers significant advantages in terms of miniaturization, system integration, cost-effectiveness, and repeatability, enabling deployment in challenging operational environments. Some applications demand compact packaging and enhanced performance characteristics. An example is condition monitoring of large plate heat exchangers, where fluid-flow-induced structural vibrations in inlet and outlet pipes provide diagnostic indicators. These applications require usable bandwidth up to 2 kHz and the the ability to resolve spectral amplitudes on the order of −120 to −110 dBg per Hz. This paper presents a comparative analysis of three contemporary devices designed for high-performance condition monitoring applications. We demonstrate a field-deployable implementation, including packaging for *in situ* measurements and the associated embedded signal processing pipeline for data acquisition. Experimental results characterize the performance boundaries and technical limitations of commercially available MEMS sensing technology.

3:40

1pSA8. Magnetostrictive-based Jerk Sensor: Experimental characterization and analytical estimation of sensitivity. Ehsan Vatankhah (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 311 E 31st St., Apt 104B, Austin, TX 78705, e.vatankhah@utexas.edu), Yuqi Meng, Xiaoyu Niu, Zihuan Liu, and Neal A. Hall (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

In this study, we explore the utilization of the magnetostrictive transduction principle for the creation of a single-axis inertial sensor. A prototype sensor is constructed using magnetostrictive Terfenol-D. We characterize the accelerometer's sensitivity through two independent methodologies,

yielding results that align with finite-element analysis. Notably, the sensor exhibits an inherent response to jerk—the time derivative of acceleration—resulting in an accelerometer sensitivity that maintains a +6 dB/octave slope relative to frequency, up to the sensor's first resonance. This design features a low output impedance, akin to moving-coil geophones, thus eliminating the necessity for intermediary buffering electronics. Moreover, we present a first-order analytical method for estimating the maximum achievable sensitivity of the sensor and thereby offering an efficient tool applicable to a broader spectrum of magnetostrictive-based sensor designs.

4:00

1pSA9. Radiation resistance models for estimating the sound power of thin unbaffled flat plates using the vibration-based sound power method. Ian C. Bacon (Phys. & Astronomy, Brigham Young Univ., 333 W 100 S, Provo, UT 84601, icbacon@byu.edu), Micah Shepherd, Scott D. Sommerfeldt, and Jonathan D. Blotter (Brigham Young Univ., Provo, UT)

The Vibration-Based Sound Power (VBSP) measurement method offers distinct advantages over traditional sound-pressure and sound-intensity techniques, particularly in environments with low signal-to-noise ratios or uncontrolled acoustic conditions. This method, based on the elementary radiator model, uses the acoustic radiation resistance matrix along with measured surface velocities to estimate sound power. Previous studies have validated the VBSP approach for various structures, including flat plates, cylindrical and spherical shells, and both simply and arbitrarily curved plates. This work extends the VBSP method to thin unbaffled flat plates by incorporating monopole, dipole, and multipole representations of the radiation resistance. Experimental VBSP measurements on aluminum and steel plates are presented, comparing sound power estimates from each representation and evaluating their agreement with the established ISO 3741 standard. The results demonstrate the high accuracy of the VBSP method in estimating sound power. Key findings show that the sound power of unbaffled plates can be modeled effectively using the equation for two out-of-phase point sources, capturing both monopole and dipole behaviors depending on the acoustic wavelength relative to the plate's dimensions. This advancement broadens the applicability of the VBSP method to unbaffled structures, offering more precise sound power predictions for a wide range of practical applications. [Work supported by the National Science Foundation.]

4:20

1pSA10. Analysis of the acoustic radiation of an infinite periodically stiffened cylindrical shell near a free surface. Xavier Plouseau-Guédé (INSA Lyon, Laboratoire Vibrations-Acoustique (LVA), UR677, 25bis Ave. Jean Capelle Ouest, Villeurbanne 69621, France, xavier.plouseau-guede@insa-lyon.fr), Laurent Maxit (INSA Lyon, Laboratoire Vibrations-Acoustique (LVA), UR677, Villeurbanne, France), Fernand Leon, Farid Chati (Laboratoire Ondes et Milieux Complexes, LOMC UMR CNRS 6294, Université Le Havre Normandie, Le Havre, France), Valentin Meyer, and Patrick Dutto (Naval Group, Ollioules, France)

The acoustic radiation from an immersed cylindrical shell, periodically stiffened by internal axisymmetric frames, has been studied in the past due to its interest in underwater applications. Particularly, it has been shown that Bloch–Floquet waves induced by the periodic arrangement of stiffeners can lead to significant radiation of the shell in the far field. However, the fluid domain in these studies was generally unbounded, which is not representative of practical applications when the submerged structure is close to the sea surface. This work, therefore, investigates the influence on radiated pressure of a free surface close to a periodically stiffened cylindrical shell immersed in water. The shell is excited by a harmonic point force. The free surface corresponds to a pressure released boundary condition. A semi-analytical model is developed based on a frequency–wavenumber decomposition. The cylindrical shell and stiffeners are represented, respectively, by Flügge's analytical model and finite element models. The radiation impedance of the fluid domain including the free surface is evaluated using the image source method. Radiated pressure results are presented as a function of angle and frequency to study the effects of the free surface, specifically on Bloch–Floquet waves.

Session 1pSC

Speech Communication: Speech Communication Poster Session

Irina Shport, Chair
Louisiana State University, 260 Allen Hall, Dept. of English, LSU, Baton Rouge, LA 70803

All posters will be on display from 1:20 p.m. to 4:20 p.m. Authors of odd numbered papers will be at their posters from 1:20 p.m. to 2:50 p.m. and authors of even numbered papers will be at their posters from 2:50 p.m. to 4:20 p.m.

Contributed Papers

1pSC1. Evaluating the relationship between speech production variability and psychomotor dysfunction in individuals with major depressive disorder. Erika L. Exton (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, eexton@umd.edu), David Klemballa (Psychiatry and Behavioral Sci., Northwestern Univ., Chicago, IL), Joseph Keshet (Andrew and Erna Viterbi Faculty of Elec. and Comput. Eng., Technion - Israel Inst. of Technol., Haifa, Israel), Vijay Mittal (Psych., Northwestern Univ., Evanston, IL), Stewart Shankman (Psychiatry and Behavioral Sci., Northwestern Univ., Chicago, IL), and Matthew Goldrick (Linguist, Northwestern Univ., Evanston, IL)

Individuals with many disorders, including major depressive disorder (MDD), often experience motor symptoms (psychomotor dysfunction). These symptoms include slowing (psychomotor retardation; PmR) and jerkiness or restlessness (psychomotor agitation; PmA). Previous work has assessed these using non-speech motor tasks, such as drawing circles quickly (indexing PmR) and smoothly (indexing PmA), or holding a steady degree of force when pressing a button (indexing PmA). The goal of this study was to test whether PmR and PmA are instantiated similarly in speech as in these non-speech motor tasks. Participants with current MDD, remitted MDD, and healthy controls completed the alternating motion rate diadochokinetic speech task, quickly repeating single syllables (/pa/, /ta/, /ka/). Syllable length (release of the consonant to vowel offset) and inter-syllable pause duration were annotated within each trial. Preliminary results support similar instantiation of motor symptoms across speech and non-speech tasks. Individuals with higher phonation rate (speaking time, excluding pauses, divided by total trial time) showed greater velocity when drawing circles. Independently, individuals with greater variability in inter-syllable pause durations had greater force variability (in the button press task) and jerk (degree of agitation in hand movement when drawing). The relationship between these motor measures and depression symptomology will be discussed.

1pSC2. Evaluating auditory-perceptual assessment of dysphonia with smoothed cepstral peak prominence measures. Ning Zheng (Dept. of Linguist, Purdue Univ., 3384 Peppermill Dr., Apt 1B, West Lafayette, IN 47906, zheng874@purdue.edu)

Standards voice diagnostics have various ways to assess human voice sound in clinical examinations, which can be both subjective and objective. Subjective assessment methods include patient's self-reported symptoms, clinician's perceptual and visual assessments, while objectives ones involve application of professional instruments to get acoustic or aerodynamic parameters. This intra-database study investigates the reliability and validity of auditory-perceptual ratings in dysphonia by means of smoothed cepstral peak prominence (CPPS) values, in attempt to better aid pre-trained voice disorder diagnosis in clinical practice. The examined clinician-based hearing assessment is the grade, roughness, breathiness, asthenia, strain (GRBAS) scale, and the investigated 296 audio files were taken from Perceptual Voice Qualities Database (PVQD). The results indicate that perceptual ratings can be trusted in most cases to identify the presence of dysphonia, and they are more reliable in sustained vowels than in connected speech according to the CPPS cutoffs. The variations in perceptual assessment across age and voice signal types were observed in detecting the presence of dysphonia.

1pSC3. Mandible movements in Japanese: A comparative study with Mandarin speakers. Kexin Wang (Kobe Univ., 1-2-20-301 Hiehara-cho, Nada-ku, Kobe, Hyogo 6570054, Japan, wkx19970303@hotmail.com), Kaiqiao Chen, Jing Sun (Kobe Univ., Kobe, Japan), Johan Frid (Lund Univ., Lund, Sweden), Ryoko Hayashi (Kobe Univ., Kobe, Japan), Donna Erickson (Haskins Labs., New Haven, CT), and Oliver Niebuhr (Univ. of Southern Denmark, Sønderborg, Denmark)

This study analyzed Japanese mandible movements using a recently developed technique, the MARRYS helmet [Gudmundsson *et al*., Interspeech (2024)]. Data were collected from 9 first language and 10 second language (Mandarin Chinese) speakers of Japanese. The speech materials included three sentences composed entirely of /a/ vowels, one neutral sentence and three focus sentences with emphasis on different target words. The results revealed that Japanese speakers showed greater variations in the jaw movements compared to Mandarin speakers. For native speakers, we observed that the right boundaries of phonological phrases and utterances were often accompanied by an increased mandible lowering, while the tendency was not noticeable for Mandarin speakers. On the other hand, at the right boundary of prosodic words, the mandible lowering was not pronounced for either Japanese speakers or Mandarin speakers. Moreover, the mora carrying the accent pitch nucleus was not often accompanied by the increased mandible lowering for both groups. For the focus sentences, native Japanese speakers showed more increased mandible lowering for emphasized words than Mandarin speakers. The findings indicate that there still remain challenges in acquiring articulatory and prosodic features for Japanese learners.

1pSC4. Audio- and sensor-based classification of essential vocal tremor with and without respiratory involvement. Rosemary A. Lester-Smith (Speech, Lang., and Hearing Sci., The Univ. of Texas at Austin, 2504A Whitis Ave Stop A1100, Austin, TX 78712, rosemary.lester-smith@austin.utexas.edu), Jinuk Kwon, and Jun Wang (Speech, Lang., and Hearing Sci., The Univ. of Texas at Austin, Austin, TX)

Essential vocal tremor (EVT) involves modulation of the fundamental frequency and intensity related to oscillation within the respiratory, laryngeal, or vocal tract subsystems. Current methods for clinical assessment of EVT focus on laryngeal involvement detected using laryngoscopy. In addition, the most common treatment for EVT (i.e., botulinum toxin injections) targets laryngeal involvement. Detection of tremor outside the larynx requires additional procedures that increase clinical training and time demands. Thus, more readily accessible and automated methods are needed for detection of tremor affecting each subsystem. The current study analyzed microphone and neck-surface vibration sensor signals collected from

12 speakers with EVT producing sustained vowels. Based on visual and tactile assessments and respiratory inductive plethysmography, nine of the 12 participants exhibited respiratory tremor. The participants with respiratory tremor had higher extents of intensity modulation in the microphone and vibration sensor signals than participants without respiratory tremor. Machine learning analyses using the modulation patterns in both signals revealed a trial-level classification accuracy of 85% and a person-level classification accuracy of 100%. These findings indicate that patterns of modulation in microphone and vibration sensor signals may differentiate the subsystems affected by tremor and have the potential to advance clinical assessment of EVT. [Work supported by NIDCD R21DC017001.]

1pSC5. LSTM based end-to-end U-Net speech enhancement middle module design method. Kaikun Pei (TongJi Univ., No. 4800, Cao'an Hwy., Shanghai 201804, China, kaikunpei@163.com), Lijun Zhang, Zhuang Zhang, JianFeng Wu, and Dejian Meng (TongJi Univ., Shanghai, China)

As a key component of intelligent voice interaction, speech enhancement based on neural network methods has been widely studied. In particular, the time-domain end-to-end speech enhancement method based on the U-Net network architecture has gradually become a hot research topic. In order to improve the quality and intelligibility of enhanced speech, we have conducted in-depth research on the design of middle layer in the U-Net network architecture, and explored the impact of LSTM layers and residual mechanisms on model performance. In addition, for feature extraction over different time scales, we propose a multi-scale collaborative modeling method to achieve synchronous extraction of short- and long-term features from speech signals. Finally, to address the issue of spectral distortion during reconstruction of enhanced speech, we propose a post-processing method using multi-scale overlap-addition to further improve the quality and intelligibility of enhanced speech. Extensive experiments were conducted on the VoiceBank + Demand dataset, and results show that our proposed method significantly outperforms other baseline models in terms of speech quality and intelligibility metrics.

1pSC6. Comparative Magnetic Resonance Imaging analysis of alveolar sound articulation in Korean, Chinese, English, and German. Jiwon Lee (Linguist, Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, 2-108, Seoul 08826, Korea, chris1790@snu.ac.kr), Ho-Young Lee (Linguist, Seoul National Univ., Seoul, Korea), and Jeong Min Lee (Post-doctoral researcher, Cheon-Ji-In Inst. of Vocalization, Seoul, Korea)

This study examines the articulatory differences of alveolar sounds in Korean, Chinese, English, and German using Magnetic Resonance Imaging (MRI) technology. Data were collected from 32 Seoul Korean speakers and 3 Chinese speakers. English data came from five speakers courtesy of the USC SPAN group, and German data from a Youtube video by the Max Planck Society. Results revealed significant cross-linguistic and intra-linguistic variations. In Korean, /t*/ and /n/ were primarily articulated as apico-denti-alveolar by 21 and 18 speakers, respectively, with some using lamino-alveolar articulation (11 and 14 speakers). The Korean /l/ showed considerable variability: 15 speakers employed apico-denti-alveolar articulation, 9 apico-alveolar, 5 retroflex, and 3 lamino-alveolar. In Chinese, /d/, /l/, and /n/ were predominantly articulated as apico-denti-alveolar, with single exceptions: /d/ as lamino-alveolar, /l/ as apico-alveolar, and /n/ as dorso-palatal. In English, /n/ and /l/ were predominantly articulated as apico-alveolar, except for one instance of apico-denti-alveolar /l/. The sounds /t/ and /d/ were split between apico-alveolar (three speakers) and lamino-alveolar articulations (two speakers). The German speaker consistently produced all alveolar sounds as apico-denti-alveolar.

1pSC7. AI classification of genuine laughter versus polite laughter. Huanxi Xia (Sichuan Univ., Chengdu, Sichuan, China), Jocelynn Cu (College of Comput. Sci., De La Salle Univ., Manila, NCR, Philippines), Kainam T. Wong (College of Comput. Sci., De La Salle Univ., Beijing, China), Merlin Suarez (College of Comput. Sci., De La Salle Univ., Manila, Philippines), and Yue I. Wu (Sichuan Univ., Sichuan University, Chengdu, Sichuan, China, ivan.wuyue@scu.edu.cn)

Emotion is central to interpersonal communication and is often disclosed by non-linguistic vocal bursts of laughter—spontaneous, semi-involuntary, even uncontrollable and insuppressible. Laughter categories (e.g., genuine delight, polite agreeableness) can provide practical insights into persons who are inarticulate or duplicitous. This research employs artificial intelligence to classify audio samples of laughter. This research identifies which "features" (and feature combinations) are efficacious and which are not.

1pSC8. Estimation of physiological vocal features from neck surface acceleration signals using probabilistic Bayesian neural networks. Joaquin Sepulveda (Dept. of Elec. Eng., Pontificia Universidad Catolica de Chile, Santiago, Chile), Jesus Parra, Emiro Ibarra, Mauricio Araya (Dept. of Electron. Eng., Universidad Tecnica Federico Santa Maria, Valparaiso, Chile), Patricio De la Cuadra (Dept. of Elec. Eng., Pontificia Universidad Catolica de Chile, Santiago, Chile), and Matias Zanartu (Dept. of Electron. Eng., Universidad Tecnica Federico Santa Maria, Av. Espana 1680, Valparaiso 2390123, Chile, matias.zanartu@usm.cl)

This study introduces a Probabilistic Bayesian Neural Network (PBNN) for estimating vocal function variables, advancing non-invasive ambulatory voice monitoring by addressing aleatoric and epistemic uncertainties in regression tasks. The PBNN estimates key physiological parameters, including subglottal pressure, vocal fold contact pressure, and thyroarytenoid and cricothyroid muscle activations, from seven aerodynamic and acoustic features. Trained on the Triangular Body-Cover Model (TBCM) of the vocal folds, the PBNN establishes a non-linear inverse mapping between inputs and outputs. These features, obtainable in ambulatory settings, enhance the practical applicability of the method. Transfer Learning integrates real voice data into the synthetic-trained network, refining subglottal pressure estimations. The PBNN generates confidence intervals that correlate prediction errors with estimated uncertainties, effectively identifying potential inaccuracies. Notably, increased uncertainty is observed at operating points with likely TBCM non-linear behaviors, such as higher subglottal pressures, indicating limitations in the selected features for capturing these effects. These findings suggest incorporating new features and additional measurements to better capture non-linear responses, paving the way for future research in the ambulatory assessment of vocal function.

1pSC9. Study of neural correlates between speech production and the basal ganglia with neural network model. Chao-Min Wu (National Central Univ., Rm. E1-338, Dept. Elec. Eng., National Central University, No. 300, Zhongda Rd., Zhongli District, Taoyuan City 320317, Taiwan (R.O.C.), Zhongli, Taoyuan 320317, Taiwan, wucm@ee.ncu.edu.tw)

Speech production is a complex action. In addition to the coordination of articulators, the transmission of neural signals and control of articulators are also important. Apart from control function of cortical areas, the influence of subcortical areas on speech has received more and more attention. However, the research methods on the basal ganglia (BG) are still limited, and there are no effective treatments for many speech disorders so far. Therefore, the purpose of this study is to investigate the correlation between basal ganglia and speech production with GODIVA, a neural network based model, simulating the left inferior frontal sulcus, the pre-supplementary motor area, the frontal operculum, and the caudate nucleus circuits to generate brain signals used in speech production. By simulating the process of the basal ganglia participating in speech production with a computational model, we predict the possible motor speech disorders from simulation results under abnormal conditions. Finally, we combined the GODIVA with the DIVA model previously developed in our laboratory with Chinese tones to build the function that the GODIVA model transmits brain signal instruction to control DIVA model for producing speech sound. In this way, we could find out the correlation between brain and motor speech disorder.

1pSC10. Voice quality distributions in a perceptual space. Vishwas Shetty (Elec. Eng., Univ. of California at Los Angeles, Los Angeles, CA), Abeer Alwan (Elec. Eng., UCLA, Los Angeles, CA), and Jody Kreiman (Head/Neck Surgery and Linguist, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Recent research [Lee and Kreiman, J. Acoust. Soc. Am. 151, 3462 (2022)] suggests that important psychoacoustic space. However, without knowledge of how voices are distributed in this space, it is difficult to

speculate about how listeners exploit its structure perceptually, or to predict which voices sound similar or different. Building on prior studies employing principal component analysis, we defined a voice space using the first two PCs that emerged from analyses of acoustic variability in voice, and examined how voices vary in their distributions within that space. The center of the distributions for individual voices in this space varied minimally across speakers, but other features of the distributions, such as the nature and extent of variability within the space, revealed substantial individual differences. These nuances of vocal variability provide insights into the psychoacoustic structure of voices, highlighting how individual patterns of variability are distributed across a shared voice space. Our analysis also explores how such variability impacts speaker verification, with implications for the accuracy and robustness of automatic systems.

1pSC11. Testing the stability of a phonological category in naturalistic speech: The case of hiatus in European Portuguese. Johanna Cronenberg (UFR Linguistique, Université Paris Cité, 8 Pl. Paul Ricoeur, Paris 75013, France, johanna.cronenberg@u-paris.fr), Lori Lamel (LISN, CNRS, Paris, France), and Ioana Chitoran (UFR Linguistique, Université Paris Cité, Paris, France)

The sequence /ia/ is phonologically classified as hiatus in European Portuguese, i.e., it consists of two full vowels with a syllable boundary in between. This stands in contrast to most Romance languages which either classify the vowel sequence as a diphthong /ja/, or follow rule-based alternations between hiatus and diphthong. While this typology is supported by small sets of controlled data in laboratory settings, we tested the stability of /ia/ in Portuguese using a large corpus of naturalistic, spontaneous speech. After a substantial cleaning and annotation process, 6806 tokens of /ia/ were selected for analysis. For each token, the first two formants and their duration were analysed using functional PCA, a technique that identifies systematic variations in multidimensional time-varying signals. The main finding was that the acoustic realization of /ia/ in Portuguese ranges from hiatus-like formant configurations when the sequence is stressed to almost monophthongal realizations in posttonic position. This study highlights the benefits of testing effects found in laboratory conditions also in naturalistic speech, where reduction processes can expose the phonetic gradience of phonological categories. In addition, the findings show that analyzing large corpora can help uncover unexpected patterns of variation and thus reveal the range of typological diversity.

1pSC12. Confidence intervals for forced alignment with the Mason-Alberta Phonetic Segmenter. Matthew C. Kelley (English, George Mason Univ., George Mason University, 4400 University Dr., Fairfax, VA 22030, mkelle21@gmu.edu)

Forced alignment is a common tool in experimental phonetics to align audio with orthographic and phonetic transcriptions. Phonetic segmentation is not a straightforward process, however, and boundaries between phonetic segments cannot be easily determined. Most forced alignment tools provide a single estimate of a boundary based on conditional probabilities of segment categories given some acoustic data. The present project introduces a method of deriving confidence intervals for these boundaries using a neural network ensemble technique with the Mason-Alberta Phonetic Segmenter. Ten different segment classifier neural networks were previously trained, and the alignment process is repeated with each model. The alignment ensemble is then used to place the boundary at the median of the time points, and 97.85% confidence intervals are constructed using order statistics. On the Buckeye and TIMIT corpora, the ensemble boundaries show a slight improvement over using just a single model. The confidence intervals are incorporated into Praat TextGrids using a point tier, and they are also output as a table for researchers to analyze separately.

1pSC13. All exchanges are created equal: A reconsideration of conversational timing analysis. Timothy Beechey (Hearing Sci. - Scottish Section, Glasgow, United Kingdom) and William M. Whitmer (Hearing Sci. - Scottish Section, Level 3, New Lister Bldg., Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom, bill.whitmer@nottingham.ac.uk)

Conversations can be studied in many different ways (e.g., content, organization); within the hearing sciences, researchers often focus on quantifying the structure of conversations acoustically, specifically in terms of turn-taking. Beginning with Norwine and Murphy, the primary metric for quantifying turn-taking has been the floor-transfer offset (FTO), which assumes that the gaps between talker turns and the overlaps between talkers lie on a single temporal dimension. There are, though, several theoretical and practical concerns with the assumed temporal unidimensionality of FTOs and their analyses. Using turn-taking data from a series of free conversations conducted with varying background sounds and numbers of interlocutors, we demonstrate the multidimensionality of FTOs; that gaps and overlaps represent separate behaviors with differing relation to other conversational metrics. The current practice of solely reporting the central tendencies of FTOs can oversimplify and potentially water down the conversational benefits that different behaviors and/or technologies may provide. Considering that each datum in the distributions of gaps and overlaps represents an event in a conversation, and that each would bear meaning to the interlocutors, we propose more nuanced analyses of conversational dynamics. [Work supported by funding from the Medical Research Council, Grant No. MR/X003620/1.]

1pSC14. A pilot corpus acoustic analysis on the alveolar tap in Turkish. Janalyn A. Miklas (English, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, jmiklas@gmu.edu) and Matthew C. Kelley (English, George Mason Univ., Fairfax, VA)

In Turkish, the “r” is pronounced as an alveolar tap as in the word *araba* [ˈɑ.ɾə.bə] “car.” Impressionistic descriptions of the Turkish tap say that word-initial taps may be fricated as in the word *resim* [ɾ̌e.ˈsim] “painting” and that word-final taps are devoiced and fricated as in the word *bir* [biɾ̥̌] “one.” Little previous research has been done to explore these acoustic characteristics and what may predict this frication. This pilot study uses the Turkish dataset from the Mozilla Common Voice corpus to examine if tap duration and segment position are predictors of frication and if the first three mel frequency cepstral coefficients are discriminative for frication on alveolar taps. The data will be analyzed using generalized additive mixed modeling, the results of which will be discussed with relation to the distribution of frication on the tap in Turkish.

1pSC15. Exploring cross-dataset generalization of Speech Emotion Recognition models. Dimitra Emmanouilidou (Res., Microsoft, One Microsoft Way, Redmond, WA 98052, Dimitra.emmanouilidou@microsoft.com)

Speech Emotion Recognition (SER) is a technology that enables machines to identify, interpret, and respond to emotional nuances of human speech. Its role in enhancing human-computer interactions becomes increasingly apparent as we prioritize the development of more intuitive and empathetic AI systems. Despite the high performance achieved by prior work on individual datasets, a critical challenge remains: cross-dataset generalization. This aspect continues to pose significant challenges and hinders technology adoption. In this study, we investigate the cross-dataset generalization capabilities of various SER models. We build on our prior observation that large pre-trained models can result in fragmented class representations and further explore model capabilities in a multi-corpora learning paradigm, toward constructing corpus-independent class representations. We utilize audio-only and joint language-audio representation learning including Wav2vec, VGGish, WavLM, and CLAP. Additionally, we explore the impact of class-agnostic data augmentation in this multi-corpora training setting. Our experiments reveal significant insights into the robustness of these embeddings for cross-dataset generalization. The findings underscore the importance of evaluating SER models beyond single-dataset performance to ensure applicability in real-world scenarios. Additionally, this comprehensive evaluation helps confirm whether the models are learning the intended features and behaviors, thereby enhancing their reliability and effectiveness.

1pSC16. Acoustic characteristics of speech-language pathologists’ productions based on perceived age and language ability of the listener. Elizabeth Ancel (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, 115 Shevlin Hall, Minneapolis, MN 55455-0279, ancel014@umn.edu), Matthew B. Winn, Lizbeth Finestack, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Speech-language pathologists (SLPs) administer assessments to determine whether children have communication disorders. Although these

assessments specify procedures to ensure comparable administration across examiners, there are still individual factors that are unaccounted for. A commonly used measure is sentence repetition, in which a child repeats a live-voice production of a sentence. Live-administered sentence repetition tasks can be affected by individual differences in prosody used by the talker, particularly their rate of speech and F0 patterns. In this study, we investigated SLPs' rate and F0 deviation as they administered a sentence repetition task to an adult and four different hypothetical children: two 3-year-olds and two 12-year-olds who are described as having developmental language disorder (DLD) or not. We conducted an online study in which SLPs produced 16 sentences, which were either taken from the Redmond Sentence Recall (Redmond, 2005) or newly developed syntactically similar sentences. For each production of a child-directed sentence, we measured the rate of speech and F0 deviation and compared those to their adult-directed productions. The results indicate the degree to which individual SLPs adapt their speech for various hypothetical children. A post-survey question suggests differing philosophies on the appropriateness of adapting speech to individual children during assessment administration.

1pSC17. Extended high frequency information improves phoneme recognition: Evidence from automatic speech recognition. Zhe-chen Guo (Commun. Sci. and Disord., Northwestern Univ., 2531 Jackson Ave APT 3W, Evanston, IL 60201, zcguo@northwestern.edu) and Bharath Chandrasekaran (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

While speech information in the extended high-frequency (EHF) range (>8 kHz) is often overlooked in hearing assessments and automatic speech recognition (ASR), there is accumulated evidence that it contributes to speech perception. It remains unclear whether this benefit arises from EHFs providing direct cues to speech categories or indirectly reflecting general auditory health. We addressed this by testing how high-frequency content affects ASR in simulated spatial listening. English speech from the VCTK corpus was resynthesized with head-related transfer functions to create spatial audio, where target speech was masked by a competing talker separated by 20°, 45°, 80°, or 120° azimuth at target-to-masker ratios (TMRs) from $+3$ to -12 dB. A CNN-BiLSTM phoneme decoder was trained on cochleagram representations of broadband or low-pass filtered (6 or 8 kHz cutoff) speech. In quiet, phoneme recognition accuracy did not differ between broadband and low-pass filtered speech. Yet, in masked conditions, higher-frequency energy improved recognition across all spatial separations, particularly at TMRs ≤ -9 dB. Furthermore, consistent with the fact that consonants contain higher-frequency components, removing EHFs disproportionately increased errors for consonants over vowels. These findings suggest a direct role of EHFs in phoneme recognition, highlighting their importance in audiometric evaluations and ASR development.

1pSC18. Effects of noise and lexical factors on English word recognition in native Japanese listeners. Takeshi Nozawa (Ctr. for Lang. Education and Res., Ritsumeikan Univ., Ritsumeikan University, 1-1-1 Nojihigashi, Kusatsu, Shiga 5258577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL)

Numerous studies on non-native sound perception highlight challenges tied to segmental perception influenced by native language phonological categories. However, the link between sound perception and lexical recognition in non-native languages is often overlooked. Conversely, research on lexical recognition focuses on factors like vocabulary frequency and phonological neighborhood density but rarely considers phoneme confusability. This study bridges these perspectives. Native Japanese listeners identified English words differing in lexical frequency and phonological neighborhood density under two conditions: no noise and speech-shaped noise (SNR + 6 dB). Participants typed the word they heard in each trial. Results showed high-frequency words were recognized more accurately than low-frequency words ($p < .001$) in both conditions. Noise reduced overall recognition accuracy, with /CCVC/ and /CVCC/ words more affected than /CVC/ words. Coda consonant identification in /CVC/ words declined notably in noise, particularly for low-frequency words. Fricatives (/f, ʃ, θ, ð/) had low identification accuracy, and vowels were influenced by lexical frequency, though to a lesser extent. Phonological neighborhood density had no observable effect, likely due to the small vocabulary size. However, lexical frequency had a significant impact. The Japanese syllable structure likely contributed to reduced accuracy in recognizing consonant clusters and coda consonants.

1pSC19. Understanding the challenges of older adults in perceiving young children's speech: The role of vowel space characteristics. Suisui Xu (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, suisuixu@u.northwestern.edu) and Marisha Speights (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

As people age, they increasingly report difficulties understanding adults' speech, suggesting even greater challenges with young children, whose speech often has lower intelligibility. Few studies, however, have explored older adults' perception of young children's speech or the reasons behind these difficulties. We hypothesized that the frequency region and area of the vowel space influence older adults' ability to understand young children's speech. Our pilot study showed that older adults' average perception accuracy for six children was below 50%. Analysis revealed that the frequency region of the vowel space, rather than its area, had a greater impact on older adults' speech perception. This differs from young adults, for whom vowel space area is more significant. Additionally, children whose vowel space was closer to adults' vowel space were more intelligible to older adults ($r = 0.89$). In this study, expert listeners, specifically speech-language pathologists, will rate children's intelligibility and categorize them into low and high intelligibility groups. We will test whether expert ratings correlate with the area and frequency region of children's vowel spaces. Results from hypothesis testing will be reported to better understand the relationship between vowel space characteristics and intelligibility when testing perception in older adults.

1pSC20. Shifting focus: Advancing prosodic assessment with naturalistic tasks. Piper MacLean (Commun. Sci. and Disord., Univ. of New Hampshire, Dover, NH), Rachel S. Burdin (Linguist Program; English Dept., Univ. of New Hampshire, Durham, NH), and Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu)

We evaluate newly developed tasks from the Naturalistic Contrastive Focus (NCF) tasks by comparing performance to that of the Profiling Elements of Prosody in Speech and Communication task (PEPS-C; Peppé, 2015) with the goal of developing a clinically relevant, ecologically valid, and easy-to-use prosodic assessment for speech language pathologists (Diehl and Paul, 2009; Hawthorne and Fischer, 2020). Our aims are to (1) examine construct validity of the NCF and PEPS-C tasks via perceptual judgments and prosodic annotations made using Points, Levels, and Ranges framework (PoLaR; Ahn *et al.*, 2021) and (2) determine the concurrent and discriminant validity between the two assessments with naïve listener judgements. Data from 17 adults and 10 children (with and without speech sound disorders) have been collected. Sentences containing target words are annotated for prominence, intonational phrase boundaries, turning points of the f0 pitch contour, and pitch ranges using PoLaR, with analyses ongoing. Initial perceptual analyses reveal that adults understand and produce contrastive focus more often during the NCF tasks (M=89%–99%, SD = 2.5%–5%) when compared to the PEPS-C tasks (M=84%–89%, SD = 10%–15%). Next steps include examining naïve listener judgments of prominence and naturalness between expressive tasks. These newly developed tasks will aid in the diagnosis and assessment of populations with speech-language disorders.

1pSC21. From wiggles to words: Acoustic and linguistic patterns in caregiver–child museum interactions. Haley McCreight (Commun. Sci. and Disord., Univ. of New Hampshire, Durham, NH) and Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu)

Child-directed speech (CDS) engages infants and young children using simplified language and exaggerated prosody (Jones *et al.*, 2023). Recent work shows that caregivers adjust speech/language features to cater toward older children as well (Hämäläinen *et al.*, 2018; Shi *et al.*, 2020). Children's museums offer a valuable resource in communities to encourage families to engage and learn together and are natural, ecologically valid settings for

data collection. This study examines (1) the relationship between caregiver prosody and caregiver linguistic complexity, hypothesizing that *more* prosodic variation will correlate with *less* complex language, and (2) whether adult prosodic patterns and linguistic complexity are reflected in child speech and language during museum exploration, hypothesizing that children whose caregivers use *high* linguistic complexity and *low* prosodic variation will have stronger language skills. Twenty-minute audio recordings of exhibit exploration were analyzed for 36 caregiver–child dyads (3–6 years old). Files are transcribed following SALT conventions (SALT, 2023) and acoustically analyzed using Praat (Boersma and Weenink, 2024). Linguistic complexity measures including mean length utterance, subordination index, and type-token ratio are reported. Prosodic measures of wiggliness and spaciousness are extracted to provide data on prosodic variation over time (Wehrle, 2022). Ongoing analyses examine measures to address stated hypotheses.

1pSC22. Exploring pixel difference noise floor in tongue ultrasound data. Pertti Palo (Commun. Sci. and Disord., Univ. of Alberta, Edmonton, AB, Canada, pertti.palo@taurlin.org), Steven M. Lulich (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), and Daniel Aalto (Commun. Sci. and Disord., Univ. of Alberta, Edmonton, AB, Canada)

Pixel Difference (PD) is a change metric for image sequences used to detect tongue movement in ultrasound data. It measures overall change by treating each frame as a vector and calculating vector norms between consecutive images. The PD curves have a substantial noise floor but it is unclear to what extent this noise is caused by physiological processes like the pulse and tensing and relaxation of muscle fibres. To evaluate the contribution of physiological processes to the PD noise floor, we measured one human participant at rest and an excised bovine tongue using the same ultrasound imaging parameters. The mean PD $\pm$ standard deviation for human tongue was $93{,}851 \pm 8409$ in rest and $274{,}348 \pm 207{,}791$ in movement. For the bovine tongue we obtained five different locations between the root and the tip. The mean PD range was (109,156, 176,433) and standard deviation range was (986, 1273). In the bovine samples, mean PD correlates with standard deviation ($r = 0.9972$) in line with the multiplicative noise model. The data from human at rest differ from this pattern due to a higher standard deviation. This suggests that the PD standard deviation—not the PD noise floor—reflects physiological processes in resting tongue ultrasound data.

1pSC23. Beyond the binary: Investigating gradient palatalization in Russian with deep learning models. Allen Shamsi (Linguist, Univ. of Florida, 289 Corry Village, Apt 10, Gainesville, FL 32607, allenshamsiev@ufl.edu), Rachel Meyer, and Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL)

Recent research demonstrates that posterior probabilities of phonological features derived from deep learning models, such as Phonet (Vásquez-Correa *et al*., 2019), reliably quantify phonetic variation, with applications in linguistic analyses (e.g., Spanish lenition) and clinical diagnostics (e.g., distinguishing Parkinson's Disease from Atypical Parkinsonism; Wayland *et al*., 2024, 2023). Building on this framework, our study examines gradience in Russian secondary palatalization, traditionally characterized as a binary distinction between palatalized and non-palatalized consonants. However, emerging research challenges this binary view, revealing context-dependent gradient variation (Parker, 2015), with articulatory evidence supporting incomplete neutralization of the plain-palatal contrast in Russian (Oh *et al*., 2024). Using the 1240-h Russian spoken corpus (Karpov *et al*., 2021), we train a deep learning model to derive posterior probabilities for phonological features, such as [+ high, −back], tied to palatalization (Padgett *et al*., 2024). This enables a detailed analysis of gradient patterns across consonant types and vocalic contexts. Our findings are expected to reveal a continuum of palatalization influenced by phonetic factors, supporting a gradient contrast utilization perspective (Parker, 2015). This study underscores the value of probabilistic approaches in capturing phonetic gradience, bridging computational methods and phonological theory, and advancing understanding of phonetic patterns in Russian and beyond.

1pSC24. Speaker and listener gender effects during speeded classification. Emelia P. BensonMeyer (Communicative Sci. and Disord., New York Univ., 665 Broadway, New York, NY 10012, epb8015@nyu.edu), Nichole Houle (New York Univ., New York, NY), and Susannah Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Sibilants, especially /s/, are strongly correlated with cisgender speaker sex (e.g., Houle *et al*., 2023; Brown, 2015). However, the relationship between sibilants and gender expansive (GE) speech is less known. The current study investigates whether GE listeners perceive GE voices differently than cisgender listeners. Listeners were presented with speech produced by transgender and cisgender men and women in two conditions: natural productions (unmodified f0) and modified f0 in which f0 was set to 160 Hz. Listeners were asked to identify the initial phoneme of minimal pairs of words as either /s/ or /ʃ/. The results indicated that listeners processed the speech samples with the modified f0 differently; nonbinary listeners showed no difference in processing speed (reaction time) for unmodified and modified tokens, whereas cisgender listeners were slower when responding to the modified productions. This is notable given that the change in f0 did not alter the fricative portion, as all fricatives were voiceless. These results suggest that GE listeners may be more "accepting" of voices that do not adhere to a strict binary. This work seeks to expand the acoustic measures of focus as it increases options for individuals regarding their voice and communication when seeking gender-affirming care.

1pSC25. Neural network-based measure of voice quality in Parkinson's disease. Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL), Rachel Meyer (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611, rmeyer2@ufl.edu), Kevin Tang (Dept. of English Lang. and Linguist, Inst. of English and American Studies, Heinrich-Heine-Universität Düsseldorf, Düsseldorf, Germany), Sophia Vellozzi, and Rahul Sengupta (Comput. & Information Sci. & Eng., Univ. of Florida, Gainesville, FL)

Parkinson's Disease (PD) significantly affects speech and voice. The Parkinsonian voice is often described as breathy, rough, hoarse, tremulous, abnormally pitched, having reduced pitch range, and unusually quiet. Changes in voice quality result from altered neurological controls of the muscles in the respiratory and phonatory systems, impacting breath support and vocal fold vibration. This study investigates the effects of PD on voice quality (e.g., breathiness) in vowel production among native Spanish speakers. The degree of breathiness is estimated from posterior probabilities calculated by recurrent neural networks trained to recognize spread glottis phonological features in Gujarati, a language contrasting breathy and modal voicing in vowels and between voiced aspirated (breathy voice) and voiceless aspirated versus plain voiced stops in consonants. It is hypothesized that vowels produced by PD patients will exhibit a higher degree of breathiness than those produced by normal controls, with degrees of breathiness potentially varying as a function of disease progression.

1pSC26. Fricative voicing in the Buckeye corpus. Sean A. Fulop (Linguist, California State Univ. Fresno, 5245 N Backer Ave. Linguist PB92, Fresno, CA 93740-0001, sfulop@csufresno.edu) and Hannah J. Scott (Comput. Sci., Oregon State Univ., Corvallis, OR)

Many claims about the prevalence of phonetic voicing in English obstruents have been made in the literature over the decades, and this study is focused on the fricatives [v, ð, z, ʒ]. Textbooks summarize the consensus that these consonants are voiced through no more than a small portion of their duration unless followed by another voiced sound. In this study, the prevalence of voicing in all 51,361 voiced fricatives of the Buckeye Corpus of American English (39 speakers) has been examined in the utterance-initial (i.e., following a pause), intervocalic, utterance-final (preceding a pause), and preconsonantal positions. The fricatives are indeed less fully voiced in general when not followed by a voiced sound, but are usually voiced for at least 50% of their duration, and often are fully voiced in these environments. Devoicing is most prevalent in utterance-final positions, more so than before a voiceless consonant.

1pSC27. Deep learning-driven phonetic profiling of dysarthric speech. Fenqi Wang (Neurology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, wang.fenqi@mayo.edu), Rene Utianski, Joseph R. Duffy, David T. Jones, and Hugo Botha (Neurology, Mayo Clinic, Rochester, MN)

Dysarthric speech poses significant challenges for clinical assessment and diagnosis due to wide variability in the nature and severity of speech impairments associated with it. To address this, this study investigates the phonetic variations in different types of dysarthric speech using the deep learning model Phonet. Recordings of the word "Catastrophe" from 54 patients diagnosed with dysarthria were analyzed to extract posterior probabilities of consonants and vowels. The analysis revealed distinct posterior probability patterns for consonants and vowels across three types of dysarthria, indicating disorder-specific articulatory-based speech characteristics. Moreover, within each dysarthric group, variations in posterior probabilities aligned with the severity of rated articulation difficulty, reflecting nuanced intra-disease phonetic differences. These findings demonstrate Phonet's ability to capture subtle speech variations and link them to both dysarthria type and severity. By quantifying these phonetic alterations, this study offers a novel and robust framework for objectively evaluating articulation difficulties in dysarthric speech. The results emphasize the clinical potential of Phonet as an advanced tool for improving the diagnosis and monitoring of dysarthria, paving the way for innovative applications in speech-based clinical assessments.

1pSC28. Training computational models with accents. Leo Moore (Linguist, Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), Osama Khalid (Comput. Sci., Univ. of Iowa, Iowa City, IA), and Ethan Kutlu (Linguist, Psychol. and Brain Sci., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com)

Accent adaptation research demonstrates that monolingual speakers can quickly adapt to novel foreign accents with brief exposure or training (Baese-Berk *et al*., 2013). However, variability in language exposure among monolinguals complicates the findings (Castro *et al*., 2022). Computational modeling provides an avenue to control linguistic diversity and isolate the effects of exposure on accent adaptation (Paszke *et al*., 2019). Using Large Language Models (LLMs) such as Wav2Vec2 and Whisper that are pretrained for Automated Speech Recognition (ASR), we explore the process of accent adaptation by simulating monolinguals exposed to single or multiple accents. We implement and fine-tune the Wav2Vec2 model using PyTorch and Whisper model with HuggingFace transformer library. These facilitate replicating human experimental designs and building decoders to evaluate model outputs. Findings suggest limited exposure to one or multiple accents fails to significantly enhance adaptation to novel accents (no accent = 69% correct, one accent = 68.6% correct, multiple accents = 69% correct) in Wav2Vec2. However, Whisper model outperforms in any training model (word error $\sim$5%). By comparing simulation data with human participant studies, we aim to identify the effects of exposure and individual variability on accent adaptation. Our results highlight the challenges of determining sufficient exposure for adaptation and underscore the importance of computational approaches to complement human-based accent perception research.

1pSC29. Using computational tools to collect naturalistic data: Phonetic alignment in political debates. Emerson Peters (Linguist, Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), Osama Khalid, Sarmad Chandio (Comput. Sci., Univ. of Iowa, Iowa City, IA), and Ethan Kutlu (Linguist, Psychol. and Brain Sci., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com)

Language science research, particularly in spoken language, has often faced challenges when it comes to collecting naturalistic production data. While in-lab production studies specifically focus on word- or sentence-level productions with extractions of certain phonetic features, sociolinguistic studies focus on understanding characteristics of a speaker and how they manifest in their production, and the interaction of these two domains led to many theoretical and empirical advancements in sociophonetic research. However, due to the laborious nature of procuring naturalistic data along with significant logistical and methodological barriers, these data are scarce. Here, we created a brand-new computational pipeline called the Acoustic Analysis Pipeline, which is a collection of various modules that can be used independently or combined to procure, process, and analyze naturalistic, multi-speaker acoustic-phonetic data automatically. Currently, the pipeline's main components are the detection, transcription, and analysis of vowel pitch and vowel formants in multi-speaker conversations. We present some preliminary data of phonetic alignment from political debates in the USA (2016–2024).

1pSC30. Expressive speech synthesis Indonesian language with extraction F0 using instantaneous frequency amplitude spectrum (IFAS). Dhany Arifianto (Medical Technol., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia) and Aprianto Dwi Prasetyo (Eng. Phys., Institut Teknologi Sepuluh Nopember, Kampus ITS, Sukolilo, Surabaya, Surabaya, Indonesia, arkiven4@gmail.com)

Voice synthesis is one of the technologies being developed at this time, not only natural sound synthesis, because humans have expressions or emotions, so expressive voice synthesis also needs to be developed. In this study, Fundamental frequency is raised as a sound synthesis feature that determines the type of expression, so an estimator is needed to get the fundamental frequency features, one of which is the IFAS Gaussian Window, which has been tested for performance with other estimators, where the IFAS Gaussian Window has a mean error that is smaller than the other estimators. Then, because the performance of the IFAS Gaussian Window is known, the HMM-based Speech synthesis is modified so that it uses the fundamental frequency features of the IFAS Gaussian Window method, then it is compared to the HMM-based Speech synthesis using the STRAIGHT Vocoder. The database used in this study uses 1 female database and 1 male database, each of which has happy, angry, and sad emotions. From this study it was found that with a better fundamental frequency estimator, it can reduce the Mel-Cepstral Distortion (MCD) value, but to improve the quality of synthetic sound, a better Vocoder is needed in accordance with the results of the Mean Opinion Score (MOS) test given to 19 respondents by playing the synthesized sound randomly.

1pSC31. Constraining lexical candidates in a landmark-based model of speech perception. Camila Moran-Hidalgo (Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, cmohi@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

Human speech perception significantly outperforms Automatic Speech Recognition (ASR) systems in noisy environments or across linguistic variations. For possible insights into how to improve these systems, it may be useful to implement models of human speech perception. This study examines processes involved in constraining lexical candidates in a landmark-based model of speech perception, which leverages detected acoustic landmarks—abrupt events signallingsignaling manner contrasts in the speech signal—to efficiently narrow down potential lexical matches. A dynamic programming algorithm compares the landmarks detected in the speech signal with the landmarks generated from the lexical representation of candidate words, updating match probabilities and pruning candidates below a defined threshold (adapted from Kenney *et al*., 2013). This process reduces computational demands while maintaining high accuracy. Preliminary experiments on single-word utterances with clean phoneme labels yielded promising results: the correct lexical candidate was identified as the top match in 86% of cases, increasing to 99.9% when considering the top three candidates. Ongoing work aims to extend this approach to longer utterances and data with imperfect labels, with the goal of benchmarking the system against human speech perception studies.

1pSC32. Investigating functional unit interactions via Granger causal analysis using diffusion and tagged MRI. Hyeonjeong Park (MGH/Harvard, Boston, MA), Fangxu Xing (Radiology, Harvard Med. School, Boston, MA), Hahn Kang (MGH/Harvard, Boston, MA), Jiachen Zhuo (Univ. of Maryland, Baltimore, Baltimore, MD), Tim Reese, Van Wedeen (MGH/Harvard, Boston, MA), Maureen Stone (Univ. of Maryland, Baltimore, Baltimore, MD), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), and Jonghye Woo (MGH/Harvard, 125 Nashua, Boston, MA 02114, jwoo@mgh.harvard.edu)

The human tongue is a muscular hydrostat that performs critical roles in speech, swallowing, and mastication through highly coordinated muscle

activity. Previous research applied Granger causality analysis to time-series strain values from individual tongue muscles, revealing predictive relationships that shed light on their sequential interactions during protrusive and speech-related tasks. Building on these findings, this study shifts the focus from single muscles to functional units of tongue motion—groups of cohesive local muscle regions—to better understand how they collaborate to produce articulate speech. We collected diffusion MRI to capture muscle fiber orientation and tagged MRI to capture motion dynamics from four participants while they articulated the phrase "a kouk." After validating the stationarity of the strains along fiber orientations with statistical tests, we identified optimal time lags and applied Granger causality analysis to evaluate predictive relationships among these functional units. Our pipeline provides new insights into tongue-movement coordination, enabling improved predictions of motion patterns and articulatory behaviors. These findings not only enhance our understanding of the biomechanics of speech but also hold promise for informing rehabilitative strategies for individuals with speech disorders.

1pSC33. Quantify the degree of similarity between two voices using relative entropy. Zhaoyan Zhang (Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave. 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

While each voice is unique, some voices sound more similar to each other than others. The goal of this study is to understand what makes two voices sound similar and how to quantify the degree of similarity between two voices. Despite large within-speaker acoustic variability in voice, voice production and its variability are shaped by the underlying vocal physiology. These physiological constraints play a crucial role in establishing the attractor state that speakers find most comfortable for speaking and in influencing how speakers modulate their voice. It is hypothesized that voices with similar attractor state and modulation patterns may sound similar to each other, and quantifying the difference in the attractor state would provide a means to quantify the degree of similarity between voices. In this study, we propose to quantify this difference by calculating the relative entropy between two voices. Specifically, the probability density function in the psychoacoustic space is calculated for different speakers, which is used to calculate the relative entropy between different voices. Preliminary listening experiments showed that voices with a smaller relative entropy tend to receive a higher similarity score.

1pSC34. A computational study of the effect of formant tuning on vocal intensity. Zhaoyan Zhang (Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave. 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Formant tuning refers to matching one of the lower vocal tract resonances to a harmonic of the voice source to maximize the radiated vocal intensity. While it is often observed in sopranos singing at high notes, there have been conflicting reports for its use in male voices. Formant tuning can have large impact on vocal intensity at high pitches, where the harmonics are far apart and their amplitudes are sensitive to frequency spacing between the harmonic and the nearest vocal tract resonance, whereas at low pitches it may become less effective. In this study, the effect of formant tuning on vocal intensity is investigated in computational simulations with parametric variations in both the vocal fold and vocal tract properties. The effect of formant tuning is quantified by the difference between the A-weighted radiated sound pressure level (SPL) and the source SPL. The results showed that formant tuning significantly increases SPL at high pitches but has only small effect at low pitches. With sufficient formant tuning, the radiated SPL also increases with increasing pitch, indicating that formant tuning is more effective in increasing SPL when vocal tract resonance is tuned to a lower harmonic.

1pSC35. Comparing audio-visually determined and automatically detected creaky voice. Sarah R. Bellavance (Communicative Sci. and Disord., New York Univ., 665 Broadway, 6th Fl., New York, NY 10012, srb664@nyu.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Recent research suggests that creaky voice varies in its acoustics, with some types demonstrating irregular cycles, period doubling, or low fundamental frequency, among other characteristics (Keating *et al*., 2015; Keating *et al*., 2023). Much of the research on creaky voice uses audio-visual criteria for determining instances of creaky voice (Dallaston and Docherty, 2020), making the identification of creaky voice largely subjective. There has been an increase in the use of COVAREP (Degottex *et al*., 2014), an algorithm for detecting creaky voice, particularly in the speech of voice disorder populations (Marks *et al*., 2023; Roy *et al*., 2024). However, validation of the algorithm beyond its initial development remains to be tested. In this study, we will examine more than 1500 sound files produced by vocally healthy speakers that have been hand-coded for the presence of creaky voice and compare these to the COVAREP output. In particular, we will examine overall accuracy (same labels for human coders and COVAREP), as well as the audio-visual characteristics of false alarms (COVAREP indicates creaky voice when there is none) and misses (COVAREP indicates no creaky voice when it is present).

1pSC36. Gender and lesion characteristics modulate the sibilant acoustics in continuous speech after tongue cancer surgery. Gillian de Boer (Univ. of Alberta, 2-164 Clinical Sci. Bldg., 11304 83 Ave NW, AB T6G 2G3, Canada, gdeboer@ualberta.ca) and Daniel Aalto (Commun. Sci. and Disord., Univ. of Alberta, Edmonton, AB, Canada)

Oral and oropharyngeal cancer and its treatment can have a devastating impact on speech. The acoustics of 4385 productions of /s/ from 89 patients (66M,23F) mean age 58.67 (range 22–82 years) were analyzed before and after (1, 6, and 12 months) glossectomy surgery. Center of gravity of the fricative power spectrum (COG) was analyzed with a linear mixed effects model with assessment time, age, gender, and amounts of resections (%) within oral and pharyngeal structures as fixed effects and random intercepts for speaker and phonetic context. Before surgery, greater tumor invasion into the floor of the mouth, male sex, and older age lowered the COG. After surgery, COG was reduced (1 month: 1900 Hz; 6 months: 1520 Hz; 1 year: 1080 Hz) and dropped more for women than men. Lesion site impacts COG: the greater the tumour invasion in the tongue, the more the 1-month COG was lowered compared to the rest of the model pointing to a transient aggravating effect. The results suggest partial recovery of speech function at 1 year. The recovery is gendered with women remaining further away from the pre-treatment values after surgery. Future analysis will consider the effects of chemo-therapy and radiation. [Work supported by the Alberta Cancer Foundation, Grant RES0061908.]

1pSC37. Effects of noise type and delivery method on voice production and task load during simulated teaching. Mark L. Berardi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA, mark-berardi@uiowa.edu) and Eric J. Hunter (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

Vocal responses to elevated background noise (Lombard effect) show promise for clinical voice assessment for occupational voice users, but implementation is limited by impractical loudspeaker setups in clinical environments. Additionally, the relationship between communication-related stress and voice use suggests that different noise types may impact vocal behavior differently. This study investigated how individuals adjust their vocal patterns in response to different types of background noise (pink noise versus babble noise) and noise delivery methods (loudspeakers versus open-air headphones). Participants simulated grade-school lectures under the four noise conditions calibrated at 75 dBA. Vocal responses were measured using sound pressure levels (SPL), self-perceived vocal effort through the Borg CR-100 scale, and subjective task difficulty was assessed using the NASA Task Load Index (NASA-TLX). Participants demonstrated equivalent changes in vocal loudness and perceived task difficulty regardless of noise delivery method, suggesting that open-air headphones can effectively replace traditional loudspeaker setups. Babble noise was consistently rated as more challenging than pink noise across both delivery methods. These findings validate the use of open-air headphones for Lombard effect studies in clinical environments and highlight the differential impact of noise types on communication stress. This research provides valuable insights for clinical voice assessment protocols.

1pSC38. Sliding-block-type vocal-tract model with a nasal cavity. Takayuki Arai (Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

A physical model of the human vocal tract developed by Umeda and Teranishi (1966) was used to test relations between phonemic and vocal features of speech. Several similar models (UT models, hereafter) were further developed by our group in the past, settling on one type with a nasal cavity, as the original UT model had. The newly developed model, or UT30-D9 model, has six blocks in the main tract, and they are inserted from the bottom side. Each block can move up and down, so that the vocal-tract configuration changes depending on their positions. The velopharyngeal port can be controlled by a dial, and its opening changes the nasality of output sounds. Such physical models are mainly being applied to the area of education, but they are also used for engineering purposes, such as designing speaking robots. In addition, UT30-D9 is applied to the area of speech pathology, where cleft palate speech was tested, because it has a nasal cavity as well as an ability to simulate many types of vowels and voice qualities. [Work supported by JSPS KAKENHI, Grant No. 24K06423.]

1pSC39. Exploring the impact of speaking rate on reaction time: Cognitive load or motor synchronization? Keiko Ishikawa (Commun. Sci. and Disord., Univ. of Kentucky, 900 South Limestone, Lexington, KY 40536, ishikawa@uky.edu), Jennifer W. Yeatts (Rehabilitation and Health Sci., Univ. of Kentucky, Lexington, KY), Brooklyn Leslie, and Isabelle Cramer (Commun. Sci. and Disord., Univ. of Kentucky, Lexington, KY)

Background: Understanding cognitive load during speech modification is critical in speech therapy, where optimizing it can enhance outcomes. Reaction time (RT) for a visual secondary task increases for clear speech, which typically involves a slower speaking rate, compared to habitual speech, potentially reflecting cognitive load or motor synchronization. This study aimed to determine which factor—cognitive load or motor synchronization—primarily affected RT changes during speech modification. Methods: Six healthy female speakers of American English (ages 19–21) participated. They counted numbers at three speaking rates—slow, habitual, and fast—while performing a visual RT task, pressing a key when a square appeared on a screen. We hypothesized that if RT reflects motor synchronization, RT would vary systematically with speaking rate, showing differences across all three conditions. Results: Mean RTs were 0.47 s (Slow), 0.44 s (Habitual), and 0.51 s (Fast). Speaking rate significantly affected RT, $F(2, 2665) = 17.50$, $p < .001$, with RTs longer for Fast compared to both Slow ($p = .001$) and Habitual ($p < .001$). Slow and Habitual rates did not differ ($p = .120$). Conclusion: These preliminary findings suggest cognitive load, not motor synchronization, drives RT differences, with Fast speech imposing the greatest demands.

MONDAY AFTERNOON, 19 MAY 2025 — GALERIE 1, 1:00 P.M. TO 5:00 P.M.

Session 1pSPa

Signal Processing in Acoustics: Acoustic Array Processing and Sound Field Reconstruction II

Efren Fernandez-Grande, Cochair
Department of Electrical Engineering, Technical University of Denmark, B.352, Oersteds Plads, Kgs Lyngby 2800, Denmark

Peter Gerstoft, Cochair
Univ. of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093

Invited Papers

1:00

1pSPa1. Room transfer function estimation from background noise processing. Efren Fernandez-Grande (Polytechnic Univ. of Madrid (UPM), Nikola Tesla S/N, Madrid, Madrid 28031, Spain, efren.fernandez@upm.es)

This study investigates how the background noise naturally present in a room can reveal the acoustic properties of the space. We hypothesize that the transfer functions in the room emerge from observing background noise events over time and across different locations. Similar techniques have been widely used in seismology and underwater acoustics to estimate Green's functions, but less so in room acoustics and audio applications. In this work, we develop a model that enables to estimate room transfer functions in the presence of isotropic noise, and validate it experimentally. By analyzing synchronous noise observations from multiple microphones distributed throughout the room, we can approximate the transfer functions between them. The approach provides information about sound propagation between all pairs of sensors subject to the noise field, unlike classical measurement methods that focus on propagation between a given source and all sensors in the field. The principles examined can be of significant value for room acoustics and immersive audio technologies.

1:20

1pSPa2. Sound field synthesis based on plane wave decomposition reconfiguring Lamé function loudspeaker array. Ming Gao (Waseda Univ., 3-4-1 Ookubo, Shinjuku-ku, Oikawa Lab, No. 59, Nishi-Waseda Campus, Waseda University, Tokyo 169-8555, Japan, kou_mei@ruri.waseda.jp), Tomohiro Sakaguchi, and Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan)

Sound field synthesis aims to physically generate acoustic fields. When considering sound field synthesis in real environments, most indoor shapes are rectangular. Despite this, conventional sound field control methods often employ circular loudspeaker array configurations, which may not efficiently utilize space and may impact usable areas. Therefore, a method based on non-circular loudspeaker array configurations is needed. Lamé function is an algebraic curve whose shape can transition circular to rectangular by increasing its order. By using Lamé function to formulate the distance from the center, plane wave generation based on non-circular layout has been verified. In this paper, we propose a method for sound fields synthesis by combining multiple plane waves, applying the principle of plane wave decomposition based on a loudspeaker array geometry using the Lamé function. We start by reformulating the existing approach for loudspeaker array configurations using the Lamé function to establish our theoretical framework. Then, we validate the feasibility of our method through MATLAB simulations of two-dimensional point source field synthesis and evaluate the synthesis performance under various parameters, offering a comparison with ideal sound field situation and conventional methods.

Contributed Papers

1:40

1pSPa3. An analytic study of high-frequency sound field reproduction using active control methods. Justin Tufariello (MITRE, 202 Burlington Rd., Bedford, MA 01730, jtufariello@mitre.org), Andrew Dominijanni, and Matthew Adams (MITRE, Bedford, MA)

Sound field reproduction through active control measures provides a means to mitigate, or "cancel," a scattered acoustic field from an ensonified object. This concept is studied for acoustic cloaking of a rigid sphere to assess the number of control sources and error sensors necessary to achieve significant high-frequency scattered field attenuation. This study expands upon previous work by Cheer [Cheer, J. Acoust. Soc. Am. **140** 1502–1512 (2016)] which explored active control techniques for simplistic scattering geometries relegated to *ka* numbers below four using finite element method. This work extends the prior study to *ka* numbers greater than 16 by instead leveraging canonical analytic solutions to calculate the scattered and radiated pressure fields. This approach overcomes computational limitations in high-frequency finite element methods and allows for the study of multiple control source geometries and variable error sensor densities. At high *ka* numbers, scattered field attenuation metrics are presented, which indicate impracticability of an isotropic acoustic cloaking solution, even for idealized scattering geometries. Further commentary on achieving realistic implementation is provided within the limitations of sensor element count and reproduced sound field frequency.

2:00

1pSPa4. Comparative study of microphone arrays comprising omnidirectional and first-order directional microphones applied to remote virtual sensing. Achilles Kappis (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, A.Kappis@soton.ac.uk) and Jordan Cheer (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom)

Virtual sensing techniques have been variously investigated within the context of active noise control, and it has been demonstrated that accurate estimation is critical to active control performance. The current work aims to compare the performance and robustness, of monitoring microphone arrays used for virtual sensing comprising omnidirectional pressure sensors and microphones with first-order directivity characteristics. Configurations with the standard first-order directivity patterns, dipole, cardioid, hypercardioid and super-cardioid, and their combinations are investigated and compared to conventional arrays with omnidirectional microphones. The estimation is performed through the formulation of observation filters that project the measured responses to the estimate of the sound field at the position of virtual microphones, using the Remote Microphone Technique. The study explores the performance and robustness of the monitoring configurations when used to estimate the pressure in a diffuse sound field. A closed-form formulation of the problem is presented, and simulations are performed to validate the theoretical results.

2:20–2:40 Break

2:40

1pSPa5. Lucky detection and localization in a time-dependent medium. Geoffrey Edelmann (Code 7160, Naval Res. Lab., 521 E Luray Ave. Alexandria, VA 22301, geoffrey.f.edelmann.civ@us.navy.mil), Daniel J. Brooker (Code 7160, Naval Res. Lab., Washington, DC), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Spatio-temporal variations in the ocean can cause distortion to acoustic wavefronts that are detrimental to the detection and localization of signals of interest using a hydrophone array. Motivated by lucky processing, we will describe and demonstrate a method to show that measurements are spatially distorted in underwater acoustic data, those distortions are intermittent, and time windows during which the wavefronts are relatively undistorted and can be exploited using lucky processing. [Work supported by the Office of Naval Research.]

3:00

1pSPa6. Experimental application of active room resonance control methods in medium-sized concert halls. Yves Pene (Loudspeaker Systems, L-Acoust., 13 rue levacher Cintrat, Marcoussis 91460, France, yves.pene@l-acoustics.com), Yoachim Horyn, Thomas Boursaud, and Christophe Combet (Loudspeaker Systems, L-Acoust., Marcoussis, France)

Room resonances can severely degrade the quality of perceived sound at low frequencies during concerts. A new method has recently been proposed to enable loudspeakers to control these resonances themselves in room without secondary control sources. These "self-controlling" loudspeakers function in a feedforward setup, using the audio signal intended for broadcast as the reference signal. Beginning with a calibration phase where impulse responses are measured between each loudspeaker and an array of microphones, control filters are subsequently calculated using field separation techniques to reduce the reflected sound field. In this work, we focus on the experimental application of this method in various medium-sized concert venues of different geometries. For each case, different microphone configurations are studied with the aim of maximizing control performance and minimizing calibration time. The active control performances obtained are analyzed in terms of RT20 reduction mapping, frequency response flattening and spectrogram homogenization.

3:20

1pSPa7. A learning-based model matching framework for tunable global audio telepresence. You-Siang Chen (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan, Hsinchu 30013, Taiwan, s108033851@m108.nthu.edu.tw), Chin-Yen Wang (Elec. Eng., National Tsing Hua Univ., Hsinchu, Taiwan), and MingSian Bai (Power Mech. Eng., National Tsing Hua Univ., Hsinchu, Taiwan)

This study presents a Learning-based Tunable GlObal Audio Telepresence (LT-GOAT) system to bring the near end users to the virtual far end, with a tunable balance between the target source field and the ambient field consisting of interference and late reverberation for acoustic array systems. The system can seamlessly vary between "Enhanced" mode for speech quality and "Immersive" mode for spatial ambience. The far-end audio signals are captured by a microphone array, and then a neural network is pre-trained to separate the target source field and ambient sound. Another network is then implemented to perform multichannel filtering based on the model matching principle. The training process minimizes the matching error between the synthesized sound field and the desired sound field at the control points situated virtually at the far end. Simulations were conducted for the LT-GOAT system involving a four-microphone array at the far end and a six-loudspeaker array in the near end. A target source and an interferer at the far end were studied under various reverberation conditions. Performance was assessed using objective metrics including Perceptual Evaluation of Speech Quality (PESQ), Short-Time Objective Intelligibility (STOI) and matching errors. The results showed that the proposed system outperformed conventional baselines.

3:40–4:00 Break

4:00

1pSPa8. Continuous-wave focusing of audible sound in a reverberation chamber using time reversal. Bryce A. Lundstrom (Phys. & Astronomy, Brigham Young Univ., BYU Dept. of Phys. & Astron., N284 ESC, Provo, UT 84602, bal83@student.byu.edu) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time Reversal (TR) is a technique used to focus of sound waves, using signals that are typically impulsive in nature, at a particular location in space and time. An impulse response is obtained between a source and a receiver, which records a record of the timing of various reflections as they arrive at a receiver. The TR technique involves broadcasting the time reversed impulse response from the source, which results in broadcasting of the various reflections in reversed timing order so that they all arrive at the receiver location at the same time. Previous work using TR of ultrasonic sine waves in elastic media [Anderson *et al.*, Appl. Phys. Lett. **94**(11), (2009)] showed that multiple sources must be used to achieve continuous-wave focusing. Here the number of sources needed to focus audible sine waves in a reverberation chamber is determined. The dependence of the number of sources on the frequency used, in relation to the room's Schroder frequency, is also explored. A modal summation numerical model is also used to compare to experimental results.

4:20

1pSPa9. Determining 3-D acoustic intensity from 2-D probes. Jacob Sampson (Brigham Young Univ., 800 W University Parkway, Orem, UT 84058, jacobbsampson@gmail.com), Carson F. Cunningham, Kent L. Gee, and Micah Shepherd (Brigham Young Univ., Provo, UT)

Acoustic intensity measurements are a useful method for determining the location of a complex acoustic source such as a rocket. However, field probes for measuring acoustic intensity may need to be placed on the ground to reduce the effects of ground reflections. Since it would be useful to obtain three-dimensional intensity from a two-dimensional probe, a new approach for processing acoustic intensity is being investigated. Two components of the acoustic intensity in the plane of the probe are calculated using the PAGE method [Thomas *et al*. J. Acoust. Soc. Am. 137 (2015)]. Using the average pressure recorded by the microphones, the total magnitude of the acoustic intensity can be estimated and used with the two components already determined to solve for the third component. To test this method of obtaining the 3-D intensity from a 2-D probe a controlled experiment was designed. Intensity probes were set up in a field with no large obstructing structures nearby and a manlift was used to place a loudspeaker at various known positions from the probes. At each position, recordings were obtained of white noise and chirp signals. Using the data collected in this controlled environment, the efficacy of this method of obtaining the acoustic intensity is assessed.

4:40

1pSPa10. Wide-band sound reproduction using a 576-channel focusing parametric array loudspeaker. Jiaxin Zhong (Graduate Program in Acoust., Pennsylvania, 201 Appl. Sci. Bldg., Graduate Program in Acoust., College of Eng., The Penn State Univ., University Park, State College, PA 16802, Jiaxin.Zhong@psu.edu), Tao Zhuang, Jing Lu (Inst. of Acoust., Nanjing Univ., Nanjing, China), and Yun Jing (Graduate Program in Acoust., Pennsylvania, State College, PA)

Generating high-contrast, wideband audio over a confined target area remains challenging for electrodynamic loudspeaker (EDL) arrays due to the large source size and extensive multi-channel signal processing requirements. In this study, we address this challenge with a multi-channel focusing parametric array loudspeaker (PAL) array. Unlike traditional multi-channel EDL/PAL arrays that apply time delays directly to wideband audio signals, we apply time delays to the ultrasonic signal to achieve beam focusing. This approach, utilizing a square-wave ultrasound signal, significantly reduces hardware resource demands while effectively controlling nonlinearly generated audio beams. We designed and fabricated a 576-channel PAL array capable of wideband sound reproduction at the focal point. Experimental results demonstrated the system's effectiveness, achieving wideband (250 Hz to 4 kHz) sound at the focal point, with audio outside this region attenuated by more than 14 dB. Our work advances the potential for high-performance wideband sound reproduction systems using massive multi-channel PAL arrays.

Session 1pSPb

Signal Processing in Acoustics: Signal Processing Potpourri I

Daniel J. Brooker, Chair
Underwater Acoustics, Navy Research Lab, 4555 Overlook Ave SW, Code 7167, Washingtron, D.C. 20375

Contributed Papers

1:00

1pSPb1. The application of Lucy Processing to dominant mode rejection. Daniel J. Brooker (Underwater Acoust., Navy Res. Lab, Washington, DC, daniel.brooker@nrl.navy.mil) and Geoffrey Edelmann (NRL, Alexandria, VA)

Lucky Processing is a formalism for underwater signal processing inspired by advances in astronomy that takes advantage of the observation that certain measurements are more valuable than others to enhance coherent signal gain. In this study, we apply Lucky Processing to acoustic source localization using horizontal beamforming on a long array in a multi-source scenario. A new covariance estimator is presented using a modified approach from Ge and Kirsteins with non-linear weighting of a dense snapshot ensemble. Our results show that Lucky Processing offers significant improvements in signal coherence and processing performance, even in snapshot-deficient scenarios. [Work sponsored by the Office of Naval Research.]

1:20

1pSPb2. Hearing-aids system using distributed assistive device and blind speech extraction method under diffuse noise. Yuto Ishikawa (Graduate School of Information Sci. and Technol., The Univ. of Tokyo, Naka-cho, 2-17-14, Odawara-city, Kanagawa 2500005, Japan, yuto_ishikawa.jp@ieee.org), Tomohiko Nakamura (The National Inst. of Adv. Industrial Sci. and Technol. (AIST), Koto-ku, Japan), Norihiro Takamune, and Hiroshi Saruwatari (Graduate School of Information Sci. and Technol., The Univ. of Tokyo, Bunkyo-ku, Japan)

To perform a smooth dialogue using the hearing-aids system in various acoustical environments, it is necessary to develop a real-time speech extraction method for hearing aids under diffuse noise conditions even with a limited computational resource. In recent years, almost everyone has a smartphone; thus, we utilize this as an assistive microphone array. For these reasons, we aim at extracting the target speech under diffuse noise conditions with distributed microphone arrays and a limited computational resource. We previously proposed a real-time speech extraction framework based on spatially regularized independent low-rank matrix analysis and rank-constrained spatial covariance matrix estimation. In this paper, we apply our method to hearing-aids systems with an assistive device and evaluate the performance. In experiments, we used eight-channel distributed microphone arrays, which consist of three microphones equipped with each ear and two microphones equipped with an assistive device simulating a smartphone. First, we confirmed that the real-time method using some spatial prior information improves the performance compared to that performed blindly with a high computational resource. Second, considering the constraints of a limited computational resource, we investigated the number and arrangement of microphones to achieve superior performance and confirmed the effectiveness of the assistive device.

1:40

1pSPb3. Learning-based blind passive weak signal detection using a particle motion vector sensor in unknown time-spreading distortion underwater channels. Rami Rashid (New Jersey Inst. of Technol., 323 Dr Martin Luther King Jr Blvd, Newark, NJ 07102, raa62@njit.edu), Ali Abdi, and Zoi-Heleni Michalopoulou (New Jersey Inst. of Technol., Newark, NJ)

Signal detection in underwater channels is a challenging task, particularly when dealing with time-spreading distortion (TSD). The challenge becomes more complicated when aiming at blind passive signal detection, where both the signal and the TSD channel are unknown. In this research, we utilize a dictionary learning (DL) approach to perform blind passive signal detection, leveraging the sparsity of underwater channels impulse responses. We conducted underwater experiments to evaluate the performance of the DL-based detection method and to compare the results with conventional detection approaches. The data were collected using GeoSpectrum's M20-040 particle motion sensor along with its analog box and extension cable. This vector sensor comprises three orthogonal accelerometer dipole sensors and one acoustic pressure omnidirectional sensor. The high detection probabilities obtained by our learning-based method using one single compact sensor exhibit its usefulness in detection scenarios where the signal and the channel response are not known in advance.

2:00

1pSPb4. Development of a time-domain Active Noise Equalizing (ANE) controller using a finite element model and real-time simulations. Alexander Schulz (Helmut-Schmidt-Univ., Holstenhofweg 85, Hamburg 22043, Germany, schulza@hsu-hh.de), Tim Karl, Sachau Delf (Helmut-Schmidt-Univ., Hamburg, Germany), and Anton Homm (Tech. Ctr. for Ships and Naval Weapons, Maritime Technol. and Res., Eckernförde, Germany)

The acoustic signature of a ship is critical for detection and identification. Therefore, studying and controlling hydroacoustic signatures is increasingly important, especially in military applications where stealth is crucial. Frequency-domain ANE controllers are commonly used, because they can target specific frequencies effectively. Although, time-domain controllers provide fast and robust noise control, they are still underexplored. Here a time-domain ANE controller is being developed, using rapid control prototyping and a finite element (FE) model of a simplified ship hull section. To enable real-time simulation, the FE model is converted into a time-discrete linear-time-invariant-state-space model, using subspace methods. The controller combines Active Vibration Control with the FxLMS algorithm, modified by a signal processing algorithm to manage multi-tonal noise reduction and amplification. Validation occurs in a combined simulation, with noise reduction and amplification, tested under dynamic environmental conditions. Furthermore, the simulation results are validated experimentally.

The controller is implemented and tested on a scaled model of a patrol boat (1:8) under laboratory conditions (airborne sound field). The measurement setup is similar to the FE model of the controller development. Results show that the time-domain controller is capable of reducing emissions of tonal frequencies in multi-tone excitation by up to 15 dB.

2:20–2:40 Break

2:40

1pSPb5. Loudspeaker parameterization using physics-informed transfer learning. Trent Furlong (The Penn State Univ., 111 Osborn Hall, University Park, PA 16802, trentfurlong@gmail.com) and Karl Reichard (The Penn State Univ., State College, PA)

A loudspeaker's electrical impedance can be modeled as an equivalent electrical circuit that is characterized by its Thiele–Small parameters. The Thiele–Small parameters are related to the electrical, mechanical, and acoustical properties of the loudspeaker. Monitoring changes in these parameters throughout a loudspeaker's life is expected to provide health-significant insights to the cause of a given failure. To test this, a series of loudspeakers were run to failure while continually measuring the total electrical impedance across its electric terminals. The Thiele–Small parameters for each loudspeaker were measured prior to each run-to-failure experiment for reference to predicted values. A physics-informed neural network was pretrained as a source model to predict the Thiele–Small parameters from a loudspeaker equivalent circuit model, which was used to generate the synthetic training data. Transfer learning was accomplished by training the source model with a small subset of the data obtained from the run-to-failure experiments to then monitor the Thiele–Small parameters throughout the life of a loudspeaker. While not exact predictions, the trend in the monitored parameters provides physical insight into the cause of each failure, making it easier to predict when failure may occur, which is applicable to other health monitoring applications.

3:00

1pSPb6. Modular convolutional neural networks for adaptable ultrasound sensing and echo analysis. Hyung-Suk Kwon (Dept. of Mech. Eng., Univ. of Michigan, University of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, kwonhs@umich.edu), Ganesh U. Patil (Dept. of Mech. Eng., Univ. of Michigan, Urbana, IL), Bogdan I. Epureanu, and Bogdan-Ioan Popa (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Using convolutional neural networks (CNNs) to process ultrasound signals is a widely studied research field with applications in nondestructive evaluation, medical diagnostics, and remote sensing. While some studies have demonstrated the ability of CNNs to classify target objects by analyzing the ultrasound echoes they produced, past approaches rely on fixed, monolithic network architectures that cannot be easily expanded to learn new objects after the initial training. In this presentation, we introduce a modular CNN approach for adaptable ultrasound sensing. Our approach uses a set of small CNNs, each specialized in distinguishing a particular target object by analyzing the echoes the object produced. The received echoes are inputs to specialized CNNs, where each CNN outputs a classification probability, i.e., the likelihood of the shape of the object that produced the input echoes. The object with the highest probability is used as the final prediction. We also demonstrate the adaptability of this modular architecture, which allows expanding the pool of known objects by simply adding CNNs specialized in identifying additional objects. This adaptability shows remarkable perception accuracy despite its simplicity. Additionally, specialized CNNs are further analyzed to understand distinct echo features that determine the object identification.

3:20

1pSPb7. Improving the signal chain of a Doppler simulation. Andrew Christian (Appl. Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov) and Randall Ali (Inst. of Sound Recording, Univ. of Surrey, Leuven, Belgium)

A recent paper described the N-Oversampled Transmitted Acoustic Pressure (NOTAP) method of auralizing moving sound sources and simulating the Doppler effect. This method eliminates possible aliasing and imaging artifacts via oversampling using the relatively simple zero-order hold interpolation heuristic (also called sample and hold). However, this oversampling approach has two main disadvantages: the generation of broadband noise and a moderate attenuation of high frequency content. Canonical strategies for dealing with these problems are not directly applicable to the NOTAP framework. This presentation explores several methods to address these problems by either extending existing strategies or proposing altogether new strategies. These include asynchronous filtering to compensate for the high-frequency losses, exploring the use of new interpolation kernels, and estimating the induced noise signal so that it can be subtracted from the final auralization. The efficacies of these strategies are discussed in terms of their computational burden and the complexity that they add to the NOTAP method.

3:40

1pSPb8. Applications of coherence to acoustic imaging for munitions response surveys. Thomas E. Blanford (Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu)

Synthetic aperture sonar systems are employed to detect unexploded ordnance in marine environments. These systems use broadband, low frequency (below 50 kHz) waveforms to penetrate the sediment. Three-dimensional imagery is then reconstructed using delay and sum beamforming. Despite the complicated nature of these sonar systems, accurate detection, localization, and classification of objects remains challenging. The reconstruction algorithms, however, do not currently exploit all the available information in the signals. Recent work in biomedical ultrasound imaging has shown how differences in spatial coherence between targets and the background can be used to improve image quality. While the signals from a munitions survey system are expected to have similar properties, differences in array geometries and issues like uncompensated platform motion pose challenges to their adoption. This presentation will describe recent investigations applying coherence-based reconstruction to synthetic aperture sonar systems. By adapting algorithms from biomedical ultrasound such as delay-multiply-and-sum, as well as exploiting the multi-static nature of these systems, it is possible to recover vital clues about an object's shape, pose, and burial depth.

4:00–4:20 Break

4:20

1pSPb9. Matched insonified waveform design using multi-tone sinusoidal frequency modulation. Steven R. Craig (Naval Undersea Warfare Ctr., Div. Newport, 1176 Howell St., Newport, RI 02841, steven.r.craig13.civ@us.navy.mil) and David Hague (Naval Undersea Warfare Ctr., Div. Newport, Newport, RI)

Matched Insonified (MI) active sonar transmit waveforms leverage information theoretic methods to optimize their Energy Spectral Densities (ESD) for optimal object detection in environments with estimatable noise and clutter Power Spectral Densities (PSD). While these methods determine the optimal waveform's ESD, they do not specify a time series that realizes that optimal ESD shape. The Multi-Tone Sinusoidal Frequency Modulated (MTSFM) waveform is an adaptive waveform model that controls its spectral shape with a discrete set of coefficients. Manipulating these coefficients allows MTSFM waveforms to approximate optimal MI waveform ESDs while maintaining a constant modulus, a critical waveform property that facilitates efficient transmission on high-power amplifiers. This research evaluates the detection performance of MTSFM waveforms for several hollow elastic objects in a physics based acoustic simulation with variable noise and clutter PSDs. Receiver Operating Characteristic (ROC) curves compare the detection performance of ideal MI waveforms, MTSFM waveforms approximating the MI waveform's ESD, and flat spectrum waveforms with transmit energy and bandwidth constraints. The MTSFM based MI waveforms closely approximate the performance of ideal MI waveforms and generally outperform flat spectrum waveforms when the object, noise, and clutter spectral content varies greatly across the frequency band of interest.

4:40

1pSPb10. Genetic algorithm-based framework for optimal sensor placement in acoustic leak localization of water distribution networks. Pranav Agrawal (Civil and Environ. Eng., Univ. of California, Los Angeles, 580 Portola Plaza, 5731 Boelter Hall, Los Angeles, CA 90095-1593, pranav0505@g.ucla.edu), Yongjie Zhuang (Stony Brook Univ., Stony Brook, NY), and Sriram Narasimhan (Civil and Environ. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

Accurately localizing leaks in water distribution networks (WDNs) remains a critical and ongoing research challenge. Acoustic-based methods are widely utilized for leak localization due to their higher sensitivity in detecting leaks and their reliance on sensor data rather than physical models. However, the problem of optimal sensor placement (OSP) for acoustic sensors has yet to be fully addressed, as robust analytical or numerical models for acoustic wave propagation in fluid-filled, two-dimensional (2-D) pipe networks are not readily available. Most existing OSP approaches rely on underlying hydraulic models, which are challenging to apply in acoustic sensing contexts. To address this gap, we propose a novel framework for solving the OSP problem specifically for acoustic sensors in WDNs. The framework defines an objective function based on "interior points"—specific locations within the network where leaks can be accurately identified using a given sensor placement configuration. The optimization problem is solved using a Genetic Algorithm, incorporating constraints to account for sound attenuation, ensuring that leak signals remain detectable at the optimally placed sensors. This study provides valuable insights for water utilities, engineers, and other stakeholders, enabling them to make budget-conscious and informed decisions about the number and placement of acoustic sensors.

5:00

1pSPb11. Unlocking the power of sound: Fundamentals and applications. Leke m. Adewunmi (urban and regional planning, federal Univ. of Technol. akure, Akinduro Close Goshen Estate, 7, Akure, Ondo 340222, Nigeria, lekeadewunmi@gmail.com)

Sound is a pervasive and transformative force, shaping our experiences and interactions. This comprehensive overview delves into the fundamentals of sound engineering, exploring the physical properties, technical/science principles, and innovative applications of audio signals. From the simple definition of sound to the basics of frequency, amplitude, and wavelength to advanced concepts in signal processing, and audio perception, this work provides a solid foundation for understanding the science and art of sound. Real-world applications in music production live sound reinforcement, post-production, audio-visual installations, and emerging fields like audio augmented reality are examined. By unlocking the power of sound, this work aims to inspire new possibilities for creative expression, communication, and technological innovation.

MONDAY AFTERNOON, 19 MAY 2025 — STUDIOS 9/10, 1:00 P.M. TO 4:20 P.M.

Session 1pUW

Underwater Acoustics: Measurements of Natural and Man-Made Underwater Sounds

Max Radermacher, Chair
Northeastern University, 360 Huntington Ave, Boston, MA 02115

Contributed Papers

1:00

1pUW1. Diversity and abundance of fish grunts recorded off coastal Guam. Isaac Fong (Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, fong.i@northeastern.edu), Cameron Hallett (Dept. of Ocean Sci., Univ. of Miami, Miami, FL), Max Radermacher (Northeastern Univ., Boston, MA), Lily Moore, Claire B Paris (Dept. of Ocean Sci., Univ. of Miami, Miami, FL), and Purnima Ratilal (Northeastern Univ., Boston, MA)

To assess fish diversity and abundance around Guam Island, a set of two stereo hydrophones were deployed in ca. 5 m of water depth and recorded continuously over a 48-h time period. The soundscape of this nearshore environment was dominated by fish grunts, which are produced by drumming muscles attached to their swim bladder. Here, we provide an analysis of these grunt signals received on the double-channel hydrophone by matched filter time-series analysis. Each 10-min sequence of acoustic data was first analyzed to extract the most significant grunt via amplitude thresholding. Next, each significant grunt signature was used as a matched filter template against significant grunts and a matrix of cross matched filter outputs were used to extract unique grunt signatures present in the dataset. The set of unique grunt signatures indicates the level of biodiversity in measured acoustic signals. These were then used as templates to extract similar grunts in the dataset via second stage matched filtering that quantifies the abundance of signals in each grunt category. Results from the acoustic analysis will be combined with concurrent eDNA sampling analysis in the future.

1:20

1pUW2. Geographic mapping of marine mammal vocalizations from diverse species in the Norwegian and Barents Seas. Arpita Ghosh (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, ghosh.arp@northeastern.edu), Hrafn S. Sigurdarson, Saunak Samantray (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), Hamed Mohebbi-Kalkhoran (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Heidi Ahonen (Norwegian Polar Inst., Tromsø, Norway), Olav R. Godoe (Inst. of Marine Res., Bergen, Norway), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A large-aperture densely populated coherent hydrophone array system was towed at multiple locations off Norway encompassing regions above and below the Artic Circle during an experiment in spring 2014. Vocalization signals up to 4 kHz from diverse marine mammals species were received over multiple diel cycles at each measurement site off Alesund,

Loften, and the Northern Finnmark region of the Norwegian and Barents Seas. Here we compare and contrast the species-dependent marine mammal vocalization temporal and spatial distributions for the three regions. Data from the coherent hydrophone array system are automatically processed to enable signal detections, bearing estimation via beamforming, and time-frequency feature extraction via pitch tracking. Bearing-time trajectories of signal detections after automatic classfication are first extensively verified by visual inspection of select spectrograms. Verified marine mammal vocalization bearing-time trajectories are next localized via the moving array triangulation technique and mapped onto geographic space, including error bounds for range and cross-range estimates. Vocalizations belonging to baleen whale species include fin, humpback and minke, and tooth whale species include sperm and beluga. The species-dependent diel, diurnal and nocturnal call rates, as well as call spatial distributions provide insights into their behavior and interactions in the undersea environment.

1:40

1pUW3. Baleen whale sounds recorded in the Gulf of Mexico with a 192-element coherent hydrophone array. Max Radermacher (Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, radermacher.m@northeastern.edu), Matthew E. Schinault, Arpita Ghosh (N), Saunak Samantray (N), Udit Sankhadasariya, and Purnima Ratilal (Northeastern Univ., Boston, MA)

The Gulf of Mexico is a biodiverse ecosystem that is comprised of a variety of marine wildlife. The 192-element hydrophone array with a sampling rate up to 100 kHz, developed in-house at Northeastern University, underwent an experiment on the R/V Weatherbird II in April 2024. The array recorded acoustic data for roughly 48 h across 5 days, while deployed at an approximate depth of 80 m. Diel cycles were captured in numerous ecological hotspots including the West Florida continental shelf and slope, and the DeSoto Canyon. During the experiment, sounds from numerous marine species were detected, including potential sperm whales, bottlenose dolphins, pantropical dolphins, and baleen whales. The baleen whale calls, composed of low frequency pulses and tonal sounds, are believed to be produced by Rice's whales, the only all year-round endemic baleen whale species. Each of these possible Rice's whale sounds were automatically detected, then manually classified from spectrogram and time-series analysis based upon several features, such as minimum and maximum frequency values, bandwidth, time duration, and slope. These manually classified signals will then be used as templates to extract similar signals in the dataset, providing localization, source level estimation, and bearing time trajectories.

2:00

1pUW4. Analysis of very low frequency wind driven noise at Ocean Observatories Initiative hydrophones. John Ragland (Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu), Minh Phan, and Shima Abadi (Univ. of Washington, Seattle, WA)

In the frequency band between 1 and 10 Hz, ambient sound in the ocean has a global pattern in the power spectral density that is driven by local surface gravity waves and is highly correlated to wind speeds. At low frequencies above 10 Hz, other source mechanisms dominate ambient sound. In this presentation, surface buoy measurements, model estimates, and satellite-based observations of wind speed are compared to spectral levels measured by the five low-frequency hydrophones that are part of the Ocean Observatories Initiative Cabled Array. Features of the spectrogram are categorized and associated with wind observations. The ambient sound spectral levels are linearly separable by wind speed at frequencies below 5 Hz. Lastly, a general frequency-temporal structure of spectral levels due to wind events is presented and discussed. [Work supported by ONR.]

2:20

1pUW5. Dependence of very-low- and ultra-low-frequency ocean ambient noise on wind speed. Anthony I. Eller (Appl. Ocean Sci., Springfield, VA), Kevin D. Heaney (Appl. Ocean Sci., 5242 Port Royal Rd. #1032, Springfield, VA 22151, Kevin.Heaney@AppliedOceanSciences.com), and David L. Bradley (Univ. of New Hampshire, Portsmouth, NH)

A long-held hypothesis is that ambient ocean noise at very-low (2–20 Hz) and ultra-low (0.1–2 Hz) frequencies is the result, at least in part, of wind-generated surface waves, and several researchers have addressed this matter, both in terms of measurement and theory. The present availability of high-sample-rate, long-term noise data from the United Nations Comprehensive Nuclear Test Ban Treaty Organization (CTBTO) hydrophones makes possible an in-depth examination of the wind–noise relationship. This paper presents a detailed data-based analysis of the dependence of noise level on wind speed, focused especially on how this relationship varies with location and with the time periods of the wind and noise variability. The results both support the hypothesis and indicate its limitations.

2:40–3:00 Break

3:00

1pUW6. Gulf of Mexico soundscape dominated by anthropogenic contributors. Vanessa M. ZoBell (Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037, vmzobell@ucsd.edu), Lynne Hodge (Univ. Corp. for Atmospheric Research's (UCAR) Cooperative Programs for the Advancement of Earth System Sci. (CPAESS), Boulder, CO), Sean Wiggins, John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA), Arturo Serrano (Universidad Veracruzana, Veracruz, Mexico), Adolfo Gracia (Instituto de Ciencias del Mar y Limnología, UNAM, Mexico City, Mexico), and Kaitlin Frasier (Scripps Inst. of Oceanogr., La Jolla, CA)

The Gulf of Mexico is a biodiversity hotspot, encompassing diverse habitats ranging from coral reefs to deep-sea seeps and supporting an array of marine species. Anthropogenic activity in the region, however, is widespread, with over 3000 actively drilling oil rigs including associated seismic exploration surveys, and over 10 international ports supporting extensive marine traffic. These activities generate underwater noise, impacting marine species that rely on sound for daily life functions. In this study, we analyzed sound levels from long-term passive acoustic recordings at eight sites across the Gulf to evaluate the influence of anthropogenic activity on the soundscape. A source separation analysis was conducted on hourly spectra to examine prominent subcomponents of the soundscape. Specific frequency bands were extracted to identify variations in sound levels associated with airguns, vessels, and wind noise. Excess noise levels were computed to determine increases in noise from modeled pre-industrial levels. Results reveal that seismic exploration airgun signals were a dominant contributor, significantly altering the low-frequency acoustic environment for the majority of the sites with noise levels at some sites reaching excess levels of over 40 dB. This research underscores the critical role of measuring underwater sound levels in assessing the ecological impacts of human activities on marine environments.

3:20

1pUW7. Method of holographic signal processing for resolving noise signals in shallow oceanic waveguides in the presence of internal waves. Sergey A. Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru) and Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst. of RAS, Moscow, Russian Federation)

The paper presents a method of holographic processing for the separation of several noise sources in an irregular shallow water waveguide. It is assumed that intensive internal waves (IIWs) are present in shallow water.-The proposed holographic method enables the separation of sources with minimal distortion. Criteria for resolving sources are analyzed in the paper. The results of numerical modeling of the resolution of two noise sources in the presence of IIWs, causing horizontal refraction and mode coupling, are presented. The 2-D Fourier transformation (2-D-FT) is used to analyze the interferograms of two sources moving in a waveguide with IIWs. The result of the 2-D-FT of an interferogram—hologram. Hologram is a composite of the focal spot sets corresponding to each source moving in the waveguide and focal spot sets related to IIWs. Sources can be separated when their holograms do not overlap. Information about source parameters is contained in the focal spots of their hologram. Thus, the separated source hologram allows for the estimation of range, velocity, and direction (Pereselkov and Kuz'kin, JASA 151(2), 666–676). [Work supported by the Russian Science Foundation, Grant No. 23-61-10024.]

3:40

1pUW8. Numerical investigation of radiated propeller noise from a model scale ship in the bollard condition. Duncan McIntyre (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, dmcintyre@uvic.ca), Mohammad-Reza Pendar (Univ. of Victoria, Victoria, BC, Canada), Shameem Islam (Ocean, Coastal and River Eng. Res. Ctr., National Res. Council, St. John's, NF, Canada), and Peter Oshkai (Univ. of Victoria, Victoria, BC, Canada)

The bollard condition describes an operating condition in which a ship's propeller is producing thrust, but the ship itself has no speed over ground, analogous to a towing pull from zero speed. The operating condition is important in applications beyond towing, including vehicle loading into ferries. Since it involves a highly loaded propeller operating in a non-optimal regime, it is also believed to produce significant underwater radiated noise; however, study of propeller noise in the bollard condition is limited compared to designed operating conditions. We employ computational fluid dynamics (CFD) at three levels of fidelity— Reynolds Averaged (RANS), Large Eddy Simulation (LES), and a hybrid RANS-LES approach—to examine the hydrodynamics and hydroacoustic sound of a ship propeller at model scale, reproducing a set of towing tank experiments examining a model ship operating in the bollard condition. Through application of Fourier and wavelet transforms of fluctuating pressures in the wake, in conjunction with quantitative analysis of vortex structures, we can identify the contributions of different flow features to overall radiated propeller sound; blade tip vortices and vortex interactions with the rudder dominate sound production. Comparison of the solution fidelities suggests different roles of each in propeller hydroacoustics.

4:00

1pUW9. The hydroacoustics of a Voith–Schneider propeller. Duncan McIntyre (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, dmcintyre@uvic.ca) and Peter Oshkai (Univ. of Victoria, Victoria, BC, Canada)

Voith–Schneider Propellers (VSPs), a type of crossflow propeller, are widely used in tug and ferry designs and are favored for their maneuverability. Their crossflow design is fundamentally different from more common, parallel-flow, propellers, and their orientation and actively pitching blades allow them change thrust direction nearly instantly. That same pitching-blade design appears to give them favorable properties for limiting vortex-induced sound and related cavitation-induced noise by supressing leading- and trailing-edge vortex formation; they are, therefore, proposed as a propulsion solution for quiet marine vessels and have been successfully implemented on research vessels with strict requirements for quiet operation. Currently, high-fidelity numerical simulations of VSPs, namely Large Eddy Simulations (LES), are limited in literature. LES solutions, in contrast to Reynolds Averaged (RANS) solutions widely used in industry, provide the detail necessary to investigate the sources of noise from propellers. We present numerical solutions, both RANS and LES, of a full-scale VSP operating in open-water conditions with and without cavitation. Through comparison with a similar rotor device lacking active blade pitching, we show that the suppression of leading- and trailing-edge vortices is a large factor in the quiet operation of VSPs. We also highlight other features, notably tip vortices, that dominate VSP-generated noise.

Session 2aAAa

Architectural Acoustics, Noise, Speech Communication, and Psychological and Physiological Acoustics: At the Intersection of Speech and Architecture I

Kenneth Good, Cochair
Architecture Acoustics, Armstrong World Industries, 2500 Columbia Avenue, Lancaster, PA 17601

Evelyn Hoglund, Cochair
Speech and Hearing, Ohio State University, 104a Pressey Hall, 1070 Carmack Road, Columbus, OH 43210

Pasquale Bottalico, Cochair
Department of Speech and Hearing Science, University of Illinois at Urbana-Champaign, 901 South Sixth Street, Champaign, IL 61820

Chair's Introduction—7:00

Invited Paper

7:05

2aAAa1. Does virtual reality match reality? Vocal performance across environments. Pasquale Bottalico (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu), Carly Wingfield (School of Music, Univ. of Illinois at Urbana-Champaign, Champaign, IL), Charles J. Nudelman (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Yvonne Gonzales Redman (School of Music, Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Singers naturally adapt their vocal production based on the acoustics of performance venues, yet rehearsal opportunities in these spaces are often limited. These adjustments are shaped by a combination of auditory and visual perceptions as well as the acoustic properties of the environment. This study examined how room acoustics affect five vocal parameters: vibrato rate, vibrato extent, vibrato jitter (Jvib), vibrato shimmer, and quality ratio (QR), an estimation of the singer's formant power. Ten classically trained professional singers (five males, five females) performed the aria da camera Caro mio ben by Giordani in their preferred key under two conditions: in three physical performance venues with varying acoustics and dimensions, and in the same venues simulated through virtual reality (VR) headsets. The findings revealed that vibrato rate, extent, jitter, shimmer, and QR remained consistent across real and VR-simulated conditions, suggesting that VR successfully replicates the immersive experience of live venues. This consistency highlights the adaptability of professional singers to both physical and virtual environments. These results underline the intricate relationship between room acoustics, visual perception, and vocal production, offering insights into how classical singers maintain performance quality despite varying environments. VR technology holds promise as a reliable rehearsal tool for vocal training and performance preparation.

Contributed Paper

7:25

2aAAa2. The handshake between natural and simulated acoustic reflections: Design for a dynamic discussion space. Shane J. Kanter (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, skanter@thresholdacoustics.com), Carl P. Giegold (Threshold Acoust. LLC, Evanston, IL), and Chris Springthorpe (Threshold Acoust. LLC, Chicago, IL)

The Harvard Business School's Klarman Hall presents a unique architectural solution to a complex acoustic challenge: creating a 1000-seat auditorium that can function effectively as an intimate 250-seat discussion space. This paper examines how architectural form and technological integration combine to support Harvard's distinctive case-study pedagogy, where student interaction and clear speech intelligibility are paramount. The hall's design leverages both passive and active acoustic elements. The overhead, marionette-style, articulating ceiling geometry creates natural early reflections that reinforce speech, while a carefully tuned voice-lift system extends these benefits throughout the space. This hybrid approach extends speech intelligibility beyond the capability of natural acoustics alone. The presentation analyzes the specific architectural features that support this acoustic flexibility, including the relationship between room geometry and early reflection patterns. Performance data demonstrate how the voice-lift system complements these architectural elements, particularly in supporting cross-room discussion. This synthesis of architectural and technological solutions offers valuable insights for designing future educational spaces that must balance multiple acoustic requirements while supporting dynamic use cases.

Invited Papers

7:45

2aAAa3. The influence of physical context on speech perception. Melissa Baese-Berk (Linguist, Univ. of Chicago, 1115 E 58th St., Rosenwald Hall Rm. 203, Chicago, IL 60657, mmbb@uchicago.edu), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), and Erica E. Ryherd (Architectural Eng., Univ. of Nebraska–Lincoln, Omaha, NE)

Speech perception is a complex process involving both the acoustic signal (i.e., bottom–up information) and a listener's knowledge and experience (i.e., top–down information). Context is also critically important to speech perception—so important, in fact, that sounds, words, and phrases can morph in how they are perceived depending on the context in which they are heard (e.g., linguistic and social context). One understudied factor that likely impacts speech perception is physical context. Physical properties, including architectural factors including layout and materials and ecological variables such as noise, differ across spaces. Hospitals are distinct from classrooms which are distinct from cars. Because important verbal information is communicated in all these physical locations, it is critically important to understand how physical context impacts speech perception. In this presentation, we will discuss the basic properties of speech perception that make it especially susceptible to context effects. We will then focus on physical context effects on speech perception, including ongoing work on how hospital noise impacts speech perception. The relative dearth of work in this area suggests a need for researchers in architectural acoustics and speech communication to collaborate on future projects to determine how physical context impacts speech perception. [Work supported by the James S. McDonnell Foundation.]

8:05

2aAA4. Rules of thumb and myths related to sound isolation and speech privacy in buildings. Kenneth Good (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrongceilings.com)

Rules of thumb should never replace true engineering but can serve to narrow down a starting point or as simplified conversation with the laity. This paper will discussion some of the rules of thumb related to speech privacy and good design practices. When the rules of thumb might be appropriate and when they may not. We will also take a few moments to explore common myths about sound isolation.

8:25

2aAAa5. Self-reported background noise and communicative health among healthcare workers. Lady Catherine Cantor-Cutiva (Audiol. & Speech Lang. Pathol., East Tennessee State Univ., 258 Lamb Hall, Johnson City, TN 37614, lccantorc@unal.edu.co) and Andrés Carrillo-Gonzalez (Universidad Nacional de Colombia, Bogota, Colombia)

Introduction: Background noise in healthcare settings can significantly impact the communicative health of healthcare workers. This study aims to determine the relationship between self-reported background noise conditions and communicative health among healthcare workers. Methods: This cross-sectional study was conducted in December 2021. Participants completed an online survey and recorded two voice samples, with and without a surgical mask. Results: The multivariate analysis showed that female registered nurses who spent more time commuting, perceived inadequate background noise conditions, experienced monotony at work, and frequently cleared their throats had significantly higher jitter values. For shimmer, factors such as marital status, job position, hospital services, working extra hours, inadequate space and background noise conditions, stress, laryngitis, colds, sleep duration, whispering, and speaking load were significantly associated with changes in shimmer among healthcare workers. Job position, inadequate background noise conditions, higher productivity during COVID-19, high stress, and screaming were associated with decreased scores on the Speech, Spatial, and Qualities of Hearing scale (Sp-SSQ12). Conclusion: These findings highlight that addressing background noise and improving working conditions could significantly enhance the communicative health and overall well-being of healthcare workers.

8:45

2aAAa6. Understanding human adaptation to acoustic spaces towards clinical voice application. Eric J. Hunter (Commun. Sci. and Disord., Univ. of Iowa, 250 Hawkins Dr., 119 SHC, University of Iowa, Iowa City, IA 52242, eric-hunter@uiowa.edu), Lady C. Cantor Cutiva (Dept. of Audiol. & Speech Lang. Pathol., East Tennessee State Univ., Iowa City, IA), Mark L. Berardi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), Pamela Hallam (Educational Leadership and Foundations, Brigham Young Univ., Provo, UT), Adrián Castillo-Allendes, Russell Banks (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), and Ahmed Yousef (Massachusetts General Hospital, Boston, MA)

Effective speech production and general human communication depend on aligning communicative intent and the environment. This presentation will review our research on how individuals perceive and adapt their voices to various communication spaces and how these environments may influence voice analysis regardless of the talkers' perception. While our studies have primarily been motivated by understanding schoolteachers and their vocal use in classrooms, we have examined test–retest variability and general speech production variability in actual and simulated room environments. Our findings indicate subtle dependencies on moderate changes in reverberation and noise, as well as significant dependencies on more substantial acoustic alterations. Beyond speech production dependencies, we have explored the perception of room changes and the sense of vocal effort required in different scenarios. The implications extend to practical applications in architectural design, particularly for spaces like classrooms, performance venues, and healthcare settings, where vocal performance and voice quantification are more critical. By understanding the relationship between human voice and architectural acoustics, we can create environments that support vocal health and communication effectiveness. This research bridges the gap between architectural science and human behavior, advancing our understanding of how built environments shape human communication.

9:05–9:25 Break

9:25

2aAAa7. Achieving clear speech in the Holy Name of Jesus Cathedral in Raleigh. Joseph F. Bridger (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, joe@sacnc.com), Mathew M. George (MMG Acoust. Consultants, Bangalore, India), Siddharth Mahajan (Performed while with Stewart Acoust. Consultants (now with TTI Floorcare), Charlotte, NC), and Julius Elo (Performed while with Stewart Acoust. Consultants, Gainesville, VA)

This study investigates the methods employed to achieve clear speech in the Holy Name of Jesus Cathedral in Raleigh. A comprehensive acoustical model was developed to assess the cathedral's acoustic properties. Ray tracing analysis was utilized to evaluate various sound system designs, focusing on their effectiveness in enhancing speech intelligibility. A detailed review of the speech-related parameters derived from these analyses was conducted. The final sound system design is described, highlighting its essential characteristics. Auralizations were created to illustrate the performance of the implemented system, demonstrating its capability to enhance auditory clarity in the cathedral's acoustically challenging environment. These findings contribute valuable insights into effective analysis and overall sound system design for similar venues. The loudspeaker element design of the sound system was by the late Frederick C. Schafer.

9:45

2aAAa8. Average absorption coefficient as a measure of classroom acoustics sound quality. Andy Carballeira (Acentech, 33 Moulton St., Cambridge, MA 02138, acarballeira@acentech.com)

ANSI S12.60 is currently under active revision, with the final copy scheduled for release in 2026. The room acoustics criteria have been reviewed and modified to reflect the consensus of a multidisciplinary expert group that included audiologists, acoustical consultants, engineers, and academic researchers. The mid-frequency average absorption coefficient is proposed as a measure of sound quality in classrooms, where the minimum design criteria for learning spaces is 0.25. This paper will examine the reasoning behind this fundamental change and explore its implications.

10:05

2aAAa9. Speech intelligibility in restaurants. Keely M. Siebein (Siebein Acoust., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com), Gary W. Siebein, Marylin Roa, Jennifer R. Miller, Abigail Gulley, Nicolas Ospina, and Gary Siebein Jr. (Siebein Acoust., Gainesville, FL)

Speech intelligibility is at the heart of the dining experience. Being able to understand what is being said at the table is critical, whether it is a romantic dinner between two lovers, an important business deal closing, or sharing laughter with family and friends. This paper will explore the intersection of the soundscape method that was used to analyze over 40 restaurants and incorporate data from the selected restaurants to understand the trends in speech perception metrics such as STI, background sound level and Reverberation Time and how they relate to different dining spaces. The idea of analyzing the Near and Far STI is explored in several restaurants. A surprising case study is examined that includes dining rooms at a retirement home, their measured acoustic metrics, hypotheses for the difficulty in speech intelligibility for these rooms, and examines potential strategies to improve.

10:25

2aAAa10. Evaluating the effects of reverberation on speech intelligibility and head movement using spatial room impulse responses. Heui Young Park (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hkp5188@psu.edu), Navin Viswanathan, and Michelle C. Vigeant (The Penn State Univ., University Park, PA)

Listeners must consistently distinguish between voices from different locations and cope with room acoustic effects in everyday listening. To evaluate how listeners cope with such conditions, three acoustic conditions: anechoic, classroom (reverberation time = 0.6 s), and auditorium (reverberation time = 1.3 s) each with two receiver positions, three target-masker separation angles: co-located, 45, and 90, and two target locations: to the front and to the side of the listener, were tested. 36 functionally monolingual American English listeners transcribed English sentences under different conditions reproduced using third-order spherical harmonics from spatial room impulse response measurements. Speech intelligibility scores and head movement were measured. Both the target and masker were female voices. Results show that speech intelligibility is significantly influenced by the room acoustic condition, target-masker separation, and target location. Speech intelligibility decreased with increasing reverberation and smaller target-masker separation. Interestingly, speech intelligibility was significantly higher for target locations that are not to the front of the listener. For head movement, main effects of the acoustic condition and target location were seen, with participants generally having more head movement for more difficult listening conditions. This study highlights the importance of implementing realistic acoustic conditions to better understand and capture how listeners cope in real-world scenarios.

10:45

2aAAa11. Individual differences in speech recognition for older adults in noisy and degraded environments. Daniel Fogerty (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, dfogerty@illinois.edu) and Judy R. Dubno (Medical Univ. of South Carolina, Charleston, SC)

Everyday environments are often noisy and can degrade temporal speech modulations. Furthermore, these environments lead to large variability in speech recognition due to individual differences in auditory and cognitive processing. Older adults with normal (ONH) or impaired hearing (OHI) completed three speech recognition experiments consisting of 15–16 measures of temporally filtered speech with (1) degraded spectral cues, (2) competing speech-modulated noise, and (3) combined degraded spectral cues in speech-modulated noise. Results were compared to other measures of speech-on-speech masking. Speech was spectrally shaped according to each listener's hearing thresholds. Speech recognition thresholds (SRTs) were determined at SRT_{20}, SRT_{50}, and SRT_{80} percent correct recognition and summarized across experiments as a single principal component. Measures of auditory and cognitive function were entered into a dominance analysis separately for ONH and OHI, which determined the relative importance of each predictor in the presence of all other

predictor combinations. Auditory and cognitive measures accounted for 72%–89% of the variance in speech recognition with greater contributions from vocabulary knowledge for ONH and from speech glimpsing abilities for OHI. These results suggest that individual differences in auditory and cognitive abilities and group differences in hearing function significantly contribute to speech recognition in degraded auditory environments.

TUESDAY MORNING, 20 MAY 2025 STUDIO FOYER, 7:00 A.M. TO 11:00 A.M.

Session 2aAAb

Architectural Acoustics: Student Design Competition (Poster Session)

Robin Glosemeyer Petrone, Chair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604

This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significant importance.

Design Scenario: A college with a very strong drama, vocal, and dance program intends to construct a new 700-seat theater primarily for dramatic/spoken word and musicals. Although the main purpose of the hall is to support their drama program, the hall will also be used for speaking engagements by the school's president and other invited speakers.

Entries to the 2025 competition will be posted in this session for viewing. Awards are made possible through a generous donation from the Wenger Foundation to the Newman Student Award Fund.

Session 2aAB

Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Distributed Acoustics Sensing (DAS) in Ocean Acoustics I

Shima Abadi, Cochair
University of Washington, 185 Stevens Way, Paul Allen Center – Room AE100R, Seattle, WA 98195

Léa Bouffaut, Cochair
K. Lisa Yang Center for Conservation Bioacoustics, Cornell University, Cornell Lab of Onithology, 159 Sapsucker Woods Road, Ithaca, NY 14850

Chair's Introduction—7:55

Invited Papers

8:00

2aAB1. Distributed acoustic sensing with ocean applications. Angeliki Xenaxi (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy), Peter Gerstoft (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, gerstoft@ucsd.edu), Ethan F. Williams (Earth and Space Sci., Univ. of Washington, Seattle, WA), and Shima Abadi (Univ. of Washington, Seattle, WA)

Extensive monitoring of acoustical activities is important for many fields, including biology, security, and ocean and Earth science. Distributed acoustic sensing (DAS) is an evolving technique for continuous, wide-coverage measurements of mechanical vibrations across oceans. DAS illuminates a fiber-optic cable with laser pulses and measures the backscattered wave due to small random variations in the refractive index of the material. Specifically, DAS uses coherent optical interferometry to measure the phase difference of the backscattered wave from adjacent locations along the fiber. External stimuli, such as mechanical strain due to acoustic wavefields impinging on the fiber-optic cable, modulate the backscattered wave. Hence, the differential phase measurements of the optical backscatter are proportional to the underlying physical quantities of the surrounding wavefield. Continuous measurement of the backscattered electromagnetic signal provides a distributed sensing modality for the external acoustic wavefield that extends spatially along the fiber. We provide a comprehensive overview of DAS technology and detail the underlying physics, from electromagnetic to mechanical and eventually acoustic quantities. We explain the effect of DAS acquisition parameters in signal processing and show the potential of DAS for sound source detection on data collected from the Ocean Observatories Initiative, DOI: https://doi.org/10.58046/5J60-FJ89.

8:20

2aAB2. Cable response for ocean-bottom distributed acoustic sensing. Ethan F. Williams (Earth and Space Sci., Univ. of Washington, Box 351310, Seattle, WA 98195, efwillia@uw.edu) and Brad P. Lipovsky (Earth and Space Sci., Univ. of Washington, Seattle, WA)

In recent years, widespread application of fiber-optic sensing methods, especially distributed acoustic sensing (DAS), on seafloor telecommunication and power cables has enabled acoustic detection and localization of marine mammals and shipping traffic, inversion of sediment properties from Scholte wave dispersion and shear wave resonance, and quantitative estimation of ocean surface gravity wave statistics in shallow water. However, the effect of cable construction, burial, and sediment characteristics on the measured fiber strain and its relationship to conventional measurands (water-column pressure or seafloor displacement) has not been rigorously addressed. We introduce an idealized model of a fiber-optic cable as a cylindrical, layered rod embedded in a uniform whole space and develop semi-analytical solutions for seismic and acoustic wave forcing. We then discuss several implications for fiber-optic sensing, including Poisson and photoelastic effects. We conclude that the strain measured by DAS on a buried cable is typically on the same order of magnitude as the longitudinal strain in the seafloor, with the difference not exceeding 20% for reasonable elastic parameters. For an unburied cable, the hydroacoustic sensitivity is more strongly dependent on the stiffness of the cladding material. In both cases, the directional response may deviate significantly from conventional theory and warrants further experimental investigation.

8:40

2aAB3. Experimental, theoretical, and numerical investigation of Distributed Acoustic Sensing frequency- and angle-dependent sensitivity. John R. Potter (IES, NTNU, Trondheim, Norway, john.r.potter@ntnu.no), Johan Emil H. Wengle (IES, NTNU, Trondheim, Norway), Ole H. Waagaard (Alcatel Submarine Networks, Trondheim, Norway), and Tor Arne Reinen (Sustainable Commun. Technologies, SINTEF Digital, Trondheim, Norway)

Distributed Acoustic Sensing (DAS) is revolutionizing passive acoustic monitoring, offering the potential to create vast arrays of uniformly spaced virtual hydrophones every few meters. Applications are expanding rapidly and now include marine mammal monitoring, which often requires higher (>1 kHz) bandwidths. DAS is unlike traditional hydrophones since the sensitivity is determined by a

complex stress-to-strain conversion from acoustic pressure into extensional strain in the fiber with a directional response. Furthermore, the nature of the extended physical aperture over which strain rates are measured by the laser pulses injected into the fiber creates a frequency-dependent directivity, both from the spatial weighting of the laser pulse autocorrelation function and from the "gauge length" over which strain rate estimates are averaged to suppress noise. We develop a simple theoretical model and apply finite element numerical modeling to account for these phenomena, generating predicted sensitivity curves as a function of the acoustic frequency and incident angle to the fiber, accounting for interrogator self-noise. These models are compared to experimental observations of several thousand received acoustic signal pulses over a large range of angles at frequencies up to 3 kHz by an OptoDAS interrogator connected to a fiber-optic cable in the Trondheim fjord, Norway.

Contributed Paper

9:00

2aAB4. From strain to pressure, evaluation of Distributed Acoustic Sensing's response to fin whale calls. Léa Bouffaut (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Cornell Lab of Onithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850, lea.bouffaut@cornell.edu), Quentin Goestchel (School of Oceanogr., Univ. of Washington, Seattle, WA), Robin Andre Rørstadbotnen (Ctr. for Geophysical Forecasting, Norwegian Univ. of Sci. and Technol. (NTNU), Trondheim, Norway), Anthony Sladen (Observatoire de la Côte d'Azur, IRD, Géoazur, Université Côte d'Azur, CNRS, Valbonne, France), Arthur H. Hartog (FOSINA, Nanterre, Hauts de Seine, France), and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Distributed Acoustic Sensing (DAS), capable of detecting water-borne sources such as low-frequency baleen whales and ships, holds great promise in underwater acoustics. However, investigating its instrumental response across the virtual channels created along an instrumented fiber is essential to enable quantitative measurements. We estimate DAS's response dependent on source-receiver configuration, incorporating gauge length impact and fiber-cable coupling. We estimate DAS received levels by framing the problem as a passive sonar equation, including the response and a conversion term between strain and acoustic pressure. Simulated received levels are compared to data from three distinct deployments, processed to isolate well-characterized and relatively stable fin whale 20 Hz calls. The datasets we used were collected from different seafloor telecommunication cables in the Northeast Pacific, the Mediterranean, and the North Atlantic with different interrogators. The similarity of the results underlines the strong influence of the grazing angle on the response. Our approach enables the identification of a conversion term applicable across sites, aligning closely with water compressibility. Additionally, we present a sensitivity analysis of selected simulation variables. This research marks a significant step in DAS characterization for marine monitoring and underscores current limitations due to instrument noise floors.

9:20–9:40 Break

Invited Papers

9:40

2aAB5. Distributed acoustic sensing measurements of active and passive sources up to 1 kHz. Alexander S. Douglass (JASCO Appl. Sci. (USA), Inc., 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu), Christina G. Liu, and Shima Abadi (Univ. of Washington, Seattle, WA)

Distributed Acoustic Sensing (DAS) is a technology that utilizes telecommunication fiber-optic cables as dense acoustic sensors by transmitting pulses of light along the cable and measuring backscatter from inhomogeneities in the fiber. The technology provides a means of dense samples (as low as several meter channel spacing) over long ranges (upwards of 100 km). An experiment was conducted in Puget Sound, near Seattle WA, in which a DAS cable and co-located hydrophones collected passive acoustic data for ~9 days, in addition to data broadcast from an active, impulsive source at 1, 5, and 10 m depth from multiple locations on the first and last days. The DAS data sampled the field at 2 kHz, with 6.38 m channel spacing over ~3.5 km of underwater cable, most of which at ~100 m depth. We explore the capabilities of the DAS measurements at frequencies up to 1 kHz from different types of sources, considering the advantages and limitations of measurements on the cable relative to the hydrophones.

10:00

2aAB6. Harnessing seafloor telecommunication cables for enhanced marine mammal monitoring: A comparative study of distributed acoustic sensing and hydrophone arrays. Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., B301, Kent St., Perth, Western Australia 6102, Australia, C.Erbe@curtin.edu.au), Evgeny Sidenko, Alexander N. Gavrilov, Rob McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, Western Australia, Australia), Olivia Collet, Boris Gurevich, Roman Pevzner, Konstantin Tertyshnikov, Roman Isaenkov, Pavel Shashkin (Ctr. for Exploration Geophys., Curtin Univ., Perth, Western Australia, Australia), Henry Debens, and Denise McCorry (Woodside Energy Ltd, Perth, Western Australia, Australia)

Sustainable management of marine industries (e.g., oil and gas, construction, subsea mining, defense, shipping/ports) typically requires baseline assessments and monitoring of marine fauna. In the case of vocal marine fauna such as whales, this is commonly achieved by passive acoustic monitoring using moored acoustic recorders. Recent literature has demonstrated the capability of fiber-optic telecommunications cables (FOC) on the seafloor to detect low-frequency (<100 Hz) sounds of nearby whales by measuring the dynamic strain induced in the cable by sound waves in the ocean [i.e., distributed acoustic sensing (DAS)]. The potential advantage of DAS is the simultaneous sensing along the length of the cable, potentially covering a larger monitoring area and enabling beamforming and thus tracking of sound sources in near or real time. We compared the performance of DAS to that of a conventional hydrophone array in a 2-month field trial off Western Australia. With controlled vessel passes we determined the sensitivity, angular dependency, and detection range of both systems. Migrating pygmy blue and Omura's whales were recorded and tracked. We discuss the pros and

cons of both technologies, and future development needs if existing FOC infrastructure is to be dual-purposed for whale detection for environmental monitoring and management.

10:20

2aAB7. Integrating hydrophone data and distributed acoustic sensing for pile driving noise monitoring in offshore environments. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wfjenkins@ucsd.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), and Wenbo Wu (Dept. of Geology & Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA)

Pile driving in offshore environments generates impulsive, broadband noise that may have adverse effects on marine ecosystems when unmitigated. Passive acoustic monitoring is performed in conjunction with pile-driving operations to measure the affected soundscape, assess the efficacy of noise mitigation, and identify marine mammal vocalizations in the area. Traditionally, this work is performed using hydrophones located in the vicinity of pile driving, as was the case during the construction of the Vineyard Wind project south of Martha's Vineyard, Massachusetts in 2023. During the course of construction at Vineyard Wind, pile-driving pulses were also recorded on a nearby fiber-optic cable capable of distributed acoustic sensing (DAS) providing data connectivity to the Martha's Vineyard Coastal Observatory operated by the Woods Hole Oceanographic Institution. While DAS data have many uses, they are constrained in some applications since they often lack a means by which they can be acoustically calibrated, particularly since the cable is buried in the seabed. By combining hydrophone data, DAS, and propagation modeling, we show that a transfer function can be developed to estimate pile-driving sound level in the water column from DAS data, which suggests that DAS could be a viable method to monitor pile-driving operations.

Contributed Paper

10:40

2aAB8. Analyzing underwater radiated noise from ships using distributed acoustic sensing. Erfan B. Horeh (School of Oceanogr., Univ. of Washington, 1503 NE Boat St. · Box 357940 · Marine Sci. Bldg., Rm. 214, Seattle, WA 98195, erfanbh@uw.edu), Shima Abadi, and William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA)

Distributed Acoustic Sensing (DAS) technology enables continuous monitoring of acoustic vibrations along fiber-optic cables, providing high-resolution spatial and temporal data for marine acoustic applications. This study investigates DAS's capability for analyzing underwater radiated noise from ships, leveraging 4 days of DAS data collected in November 2021 using two cables from the Ocean Observatories Initiative Regional Cabled Array, extending offshore central Oregon. Data were collected using OptaSense and Silixa interrogators with gauge lengths of either 30 or 50 m and sampling rates ranging from 200 to 1000 Hz. Ship signals were identified in DAS data using Automatic Identification System (AIS) information and analyzed across different fibers and interrogators to assess system performance. Our results demonstrate that large ships are clearly visible in the DAS data within the 10–90 Hz frequency band at ranges exceeding 10 km. Low-frequency filtering extends the detection range, enhancing the ability to monitor ship signals. DAS-derived ship signals are compared with hydrophone data and a propagation model to assess the DAS system's detection capabilities. This study highlights the potential of DAS technology for maritime acoustic monitoring and its complementary role to traditional hydrophone systems in capturing underwater radiated noise by ships.

Session 2aAO

Acoustical Oceanography and Underwater Acoustics: Acoustical Oceanography at Deep Water Abrupt Topography II

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943

Ying-Tsong Lin, Cochair
Woods Hole Oceanographic Institution, Scripps Institution of Oceanography, La Jolla, CA 92093

Lauren Freeman, Cochair
NUWC Newport, Naval Undersea Warfare Center (NUWC), 1176 Howell Street, Newport, RI 02841

Contributed Papers

7:20

2aAO1. Evaluation of a machine learning-based algorithmic pipeline for real-time onboard removal of glider self-noise from passive acoustic data. Anna Hu (Univ. of Georgia, Athens, GA), Guoming Li (Dept. of Poultry Sci., Univ. of Georgia, Athens, GA), Erin Meyer Gutbrod (School of the Earth, Ocean & Environment, Univ. of South Carolina, Columbia, SC), and Catherine R. Edwards (Dept. of Marine Sci., Skidaway Inst. of Oceanogr., Univ. of Georgia, Savannah, GA, catherine.edwards@skio.uga.edu)

Gliders are autonomous underwater vehicles that move by changing their buoyancy and center of gravity, making them a relatively quiet and thus attractive platform for collecting passive acoustic data for many applications, including soundscape/fisheries management, real-time monitoring of endangered whales, and acoustic propagation modeling/validation. Hydrophones can be integrated into the glider for automated analysis and identification from a purpose-built library, but broadband self-noise from internally moving motors and components must be removed to enhance precise signal processes for targets of interest. Passive acoustic data from two 2024 right whale monitoring missions in shallow coastal waters of the Georgia/Florida calving grounds were used with glider engineering data to develop signal processors and train a machine learning algorithm to identify and remove glider self-noise. Algorithm performance is evaluated using 2025 monitoring missions to assess potential improvement over current onboard filtering methods. Given the extremely shallow depths of operation for monitoring in the calving areas (15–25 m water depth), careful elimination of self-noise while retaining useful data may improve real-time detection capability. Algorithm performance on deep-water (>1000 m) data from 2022–2023 New England Sea Mounts Acoustics (NESMA) field experiments will be used to explore the viability of onboard, real-time use over a broader range of applications.

7:40

2aAO2. Reshaping and redirection of a tomographic impulse by local and distant undersea mountains. David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, 1959 NE Pacific St., Seattle, WA 98195, dallosto@uw.edu)

For the past 2 years, the 75-Hz pseudo-impulse from the Kauai Beacon has been captured on hydrophones maintained at Wake Island to enforce the Comprehensive Test Ban Treaty, sampling a 3000-km span of the North Mid-Pacific Subtropical Ocean Basin. Considering the curvature of the Earth, the source and receiver sit well below the horizon and the bulk of the acoustic energy is trapped in the deep ocean waveguide. The acoustic propagation becomes diffuse as the sound passes over the Mid-Pacific Mountains, which rise up into the Deep Sound Channel (DSC). The reception at Wake Island is made atop a steep bathymetric feature known as a Guyot—an undersea volcanic mountain that quickly rises from the deep seafloor but terminates well below the sea surface, and in this case below the DSC axis too. Models of the propagation processes are examined, specifically with the goal of interpreting the observed energy distribution at the receiver to correctly track the timing of the bulk-energy arrival, and thus deep ocean temperature. Results, cast in terms of deep ocean temperature changes, show a strong deviation in 2024 (much warmer than seasonally predicted) due to the transition into La Ninota.

8:00

2aAO3. Ambient sound directionality on a deep-water drifting platform: Measurement and model comparisons using hydrophone arrays and vector sensors. Alison B. Laferriere (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92109, alaferriere@ucsd.edu), Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Dieter Bevans (NUWC Keyport, Keyport, WA), Eric Berkenpas (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD), Mike Shepard (Second Star Robotics, Silver Spring, MD), Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Lauren Freeman (NUWC Newport, Newport, RI)

Between 2022 and 2024, multiple deployments of deep-water, drifting, opto-acoustic platforms were conducted near the New England Seamounts (~4000 m depth) and in the Southern California Bight (~1000 m depth). The platforms were equipped with conductivity, temperature, and depth (CTD) sensors, vertical and tetrahedral hydrophone arrays, and a three-axis acoustic vector sensor. For one deployment in Southern California, an 80-m thermistor string was suspended from the platform, providing high-resolution temperature profiles complementing the acoustic measurements. The drifters continuously recorded mid-frequency (500–20 000 Hz) ambient sound for 1–4 days at depths ranging from 100 to 800 m, entering a "hibernation" mode to minimize self-noise. The directionality and spatial coherence between hydrophones were computed and compared to analytical models and numerical simulations of surface-generated noise. Data between the hydrophone arrays and the acoustic vector sensor were also compared. Temporal and spatial variations in the acoustic environment, particularly 15–20 min oscillations in the surface noise directionality, were observed at both sites. The seamounts data displayed more dynamic changes in surface-generated noise directionality but less bio-acoustic activity. These results highlight the utility of these platforms for characterizing complex acoustic environments and cross-checking beamforming and vector sensor measurements. [Work sponsored by ONR TFO.]

8:20

2aAO4. Analysis of bistatic mid-frequency seamount scattering. Chad M. Smith (The Penn State Univ., Appl. Res. Lab., State College, PA 16804, chad.smith@psu.edu) and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

During the Office of Naval Research (ONR) New England Seamounts Acoustics (NESMA) experiment, low- to mid-frequency acoustic measurements were made of scattering from steep and irregular seamount flanks. These measurements included bistatic scattering from a stationary source (operated by Scripps Institution of Oceanography) to a towed horizontal line array (the Penn State Three Octave Research Array). The use of broadband waveforms and horizontal beamforming allows the separation of seabed/surface interaction and the azimuthal extent of scattering from the bathymetrically complex and acoustically rough seamount flank. The impact of these features is seen in the form of anisometric, out-of-plane scattering with an angular extent greater than 30 deg. Acoustic receptions appear to have significant incoherent energy due to seamount roughness, but also contain deterministic components from specific regions of the seamount flank. This talk will discuss acoustic measurements and analysis, the spatial distribution, and statistical characteristics of scattered returns.

8:40

2aAO5. Inferring subsurface sound speed near the Gulf Stream by combining satellite altimetry with a climatology from underwater glider observations. Robert E. Todd (Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS #21, Woods Hole, MA 02543, rtodd@whoi.edu) and Alice S. Ren (Woods Hole Oceanogr. Inst., Woods Hole, MA)

The Gulf Stream separates cooler, fresher waters of subpolar origin from warmer, saltier subtropical waters, with a notably deeper sound speed minimum on the warm side of the front. North and east of its separation from the continental margin at Cape Hatteras, the Gulf Stream meanders substantially, leading to large variability in acoustic properties on time scales of days to weeks. Despite its varying position and orientation, the structure of the Gulf Stream front is known to be quite stable. Here, satellite altimetry is used to determine the location and orientation of the Gulf Stream at particular times, and a seasonal, stream-coordinate climatology developed from nearly a decade of underwater glider sampling is used to infer corresponding subsurface sound speed structure. Resulting predictions are compared to measured sound speed profiles and predictive skill is compared to an operational numerical simulation.

9:00–9:20 Break

9:20

2aAO6. Estimation of source depth from low-frequency acoustic transmissions through the Gulf Stream. Katherine Hoekstra (Woods Hole Oceanogr. Inst., Bigelow Lab., 98 Water St., Woods Hole, MA 02543, katherine.hoekstra@whoi.edu), Julien Bonnel (Woods Hole Oceanogr. Inst., Woods Hole, MA), John A. Colosi (Woods Hole Oceanogr. Inst., Monterey, CA), Matthew Alford, Charlotte Bellerjeau (Univ. of California, San Diego, Scripps Inst. of Oceanogr., La Jolla, CA), Matthew Dzieciuch (Univ. of California, San Diego, Scripps Inst. of Oceanogr., San Diego, CA), and Gunnar Voet (Univ. of California, San Diego, Scripps Inst. of Oceanogr., La Jolla, CA)

The New England Seamounts Calibrated Acoustic Fluctuation Experiment (NESCAFE) is an ongoing experiment supported by ONR to assess how the New England seamounts and Gulf Stream interact to affect low-frequency underwater acoustic propagation. In a spring 2023 pilot study, two transceiver moorings were deployed, on either side of the Gulf Stream and away from the seamounts at a range separation of 153 km. The transceivers sent three consecutive 2-min-long up-down 200–300 Hz FM sweeps every hour, and the signals were recorded on upper/lower arrays nominally from 88 to 343 m and 850 to 1093 m. Due to strong currents, one of the sources designed to be at a depth of 1100 m, was pulled down close to a depth where the mooring buoyancy could fail. A total of 23 transmissions were recorded from this source before the mooring did fail and the aim of this talk is to present source localization estimates derived from the receptions made 153 km away. The localization was aided by high-resolution shipboard Conductivity, Temperature, and Depth (Fast CTD) measurements taken between the moorings but 1 month after the failure. This talk will also investigate the impact of sound-speed profile uncertainty in determining the source depth.

9:40

2aAO7. Subtropical Mode Water as an acoustic waveguide at the Gulf Stream. Nicholas Beaird (Lincoln Lab., Massachusetts Inst. of Technol., 244 Wood St., Lexington, MA 02420, Nicholas.Beaird@ll.mit.edu), Madeline Miller (Lincoln Lab., Massachusetts Inst. of Technol., Lexington, MA), Jeffrey Book (U.S. Naval Res. Lab., Stennis Space Ctr., MS), Catherine R. Edwards (Skidaway Inst. of Oceanogr., Univ. of Georgia, Savannah, GA), Joseph Edwards (Lincoln Lab., Massachusetts Inst. of Technol., Lexington, MA), Donglai Gong (Virginia Inst. of Marine Sci., Gloucester Point, VA), Stephen Lynch (MITRE, Bedford, MA), Travis Miles (Rutgers Univ., Rutgers, NJ), and John Osborne (U.S. Naval Res. Lab., Stennis Space Ctr., MS)

In this work, we describe the distribution and variation of subsurface acoustic ducts formed by North Atlantic Subtropical Mode Water in the region where the Gulf Stream encounters the New England Seamount chain. High-resolution observations were obtained during two cruises (November 2022 and May/June 2023) from ships and teams of up to seven gliders as part of the Predictions of AcousticS with Smart Experimental Networks of GlidERS (PASSENGERS) project. Low stratification mode water layers of varying thickness were widely observed from gliders, shipboard CTD, and towed underway CTD casts. The cruises bracket the wintertime formation of the mode waters and are located near the region of maximum production. Variation in the mode water properties on the scale of the local baroclinic Rossby radius was observed resulting from an eddy field associated with the southern flank of the Gulf Stream. A strong streamwise/spanwise asymmetry in the ducts exists in the region. The seasonal production and erosion of mode water south of the Gulf Stream extension imprints on the subsurface duct, with thicker ducts and lower sound speed minima in the spring relative to the fall.

10:00

2aAO8. Acoustic simulation with data-assimilated ocean models in the New England seamount environment. Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu), John Osborne (U.S. Naval Res. Lab., Stennis Space Ctr., MS), and Ying-Tsong Lin (Scripps Inst. of Oceanogr., La Jolla, CA)

In July of 2024, a real-time acoustics simulation effort on board the RV Roger Revelle supported science operations of the New England Seamounts Experiment (NESMA). Highly range-dependent bathymetry in the experiment region provided a computationally demanding model environment that stressed the importance of balancing compute time and simulation fidelity. Also imperative to the work were data-assimilated ocean forecast models which enabled time-dependent acoustic forecasting. This work presents the time, accuracy, and hardware costs of generating on-demand acoustic forecasts in a strongly 3-D environment, with a preliminary analysis of data-assimilated ocean model impact on the acoustics modeling products as more ship-based data was incorporated. Of particular value in this analysis are signals received by moored hydrophones deployed both on and near the New England seamounts, enable the identification of seamount scattering effects. Signal sources include SUS deployments and regular low-frequency transmissions from network moorings. The analysis includes comparisons with acoustic data and modeled sensitivity to environment uncertainty. [This work was funded by ONR.]

10:20

2aAO9. Active acoustic detection of fish and zooplankton along bathymetric features of the New York Bight. Brandyn M. Lucca (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Henderson Hall, Seattle, WA 98021, brandyn.lucca@gmail.com) and Joseph Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY)

The Hudson Canyon, located along the shelfbreak of the New York Bight (NYB), has been characterized as a “biological hotspot” that attracts

organisms ranging from zooplankton to marine mammals. Since 2017, seasonal surveys have collected multifrequency acoustic backscatter data (38, 70, 120, and 200 kHz) and hydrographic measurements to investigate how bathymetric features, such as the continental shelf and submarine canyons, influence spatiotemporal distributions of marine organisms throughout the NYB. Relationships between overall biological backscatter (fish and zooplankton) and various oceanographic variables were evaluated using a hierarchical model. Fitted relationships were projected onto numerical ocean model outputs to explore how physical oceanographic features, such as the Mid-Atlantic Cold Pool ($\sim$9 °C isotherm) and abrupt bathymetric changes, drive spatial variability in marine organisms. Combined biological backscatter was predominantly offshore in winter before shifting inshore during the summer, with hotspots predicted along the shelfbreak and Hudson Canyon throughout the year. The spatial extent of the cold pool displaced near-bottom fish offshore in spring and summer, concentrating fish backscatter between the shelfbreak front and the southern extent of the cold pool. These findings provide valuable insights into the role of physical features and oceanographic processes in shaping fish and zooplankton distributions throughout the NYB ecosystem.

10:40

2aAO10. Estimation of seafloor properties using sources of opportunity in the 2023 NESMA Pilot experiment. Ernst M. Uzhansky (Phys. Dept., Naval Postgrad. School, SP-120, 1 University Circle, Monterey, CA 93943, ernstuzhansky@gmail.com), Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA), TSUWEI TAN, Matthew W. Walters (Phys. Dept., U.S. Naval Acad., Monterey, CA), John Joseph (Oceanogr. Dept., Naval Postgrad. School, Monterey, CA), and Matthew Dzieciuch (SIO, UCSD, La Jolla, CA)

Naval Postgraduate School operated a network of Moored Autonomous Noise Recorders (MANRs) in the vicinity of the Atlantis II Seamounts during the 2023 New England Seamounts Acoustics (NESMA) Pilot experiment. Each MANR had a single hydrophone located a few meters above the seafloor. Acoustic pressure was recorded continuously by three MANRs for about 2 months. Two of the MANRs were located on steep flanks of the Atlantis II Seamounts and the other MANR was deployed in a deep trench. In addition to ambient sound, MANRs recorded signals from various compact sources of opportunity, including passing ships, chirp signals from tomographic moorings SIO-E and SIO-N of the Scripps Institution of Oceanography, and the powerful impulsive sound generated by the catastrophic demise of the SIO-E mooring. In this work, arrival patterns from SIO-E chirps and the impulsive sound are analyzed to determine time-dependent source depth. Frequency and angular dependence of the amplitudes of identified ray arrivals are employed to estimate roughness and constrain geoacoustic parameters of the seafloor on SIO-E—MANR propagation paths. Additional constraints on seafloor roughness and reflectivity are derived from the observed Lloyd's mirror-type interference pattern of broadband noise of the R/V Neil Armstrong. [Work supported by ONR.]

Session 2aBAa

Biomedical Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Double, Double, Toil and Trouble - Towards a Cavitation Dose I

Christy K. Holland, Cochair
Internal Medicine, Division of Cardiovascular Health and Disease, and Biomedical Engineering, University of Cincinnati, Cardiovascular Center, Room 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Eleanor P. Stride, Cochair
University of Oxford, Institute of Biomedical Engineering, Oxford OX3 7DQ, United Kingdom

Chair's Introduction—7:55

Invited Papers

8:00

2aBAa1. Clinical needs and standardization of cavitation monitoring in focused ultrasound therapies. Frederic Padilla (Focused Ultrasound Foundation, 1230 Cedars Court, Ste. 206, Charlottesville, VA 22903, fpadilla@fusfoundation.org) and J. Brian Fowlkes (Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI)

Cavitation monitoring is a cornerstone of focused ultrasound (FUS) therapies, including applications such as blood–brain barrier opening (BBBo), drug delivery, and histotripsy. This presentation will explore the clinical needs for cavitation monitoring, including assessing technical success (cavitation present, yes/no), enhancing safety, and quantifying efficacy (e.g., BBBo achieved, drug delivery quantified). Standardization is critical for interpreting cavitation data across modalities, devices, and clinical sites. Defining cavitation doses—potentially tailored to specific applications—can help with the comparison of results among treatment centers and the consistency of outcomes. Lastly, we will discuss opportunities for establishing data repositories to facilitate retrospective analyses and advance our understanding of cavitation's role in therapeutic outcomes.

8:20

2aBAa2. Abstract withdrawn.

8:40

2aBAa3. Cellular trouble for cavitation dose monitoring in microbubble-mediated drug delivery? Klazina Kooiman (Erasmus MC, Wytemaweg 80, Rm. Ee2302, Rotterdam 3015 CN, Netherlands, k.kooiman@erasmusmc.nl)

Ultrasound-activated microbubbles have shown great drug delivery potential to treat cardiovascular disease and cancer, both preclinically and clinically [Bouakaz and Escoffre, Adv. Drug Deliver. Rev. 206, 115199 (2024)]. Drug delivery is induced through the mechanical forces that ultrasound-activated microbubbles exert onto tissue, which opens up drug delivery barriers. Although the mechanism remains unclear, it is clear that microbubbles need to oscillate above a threshold (0.9 μm oscillation amplitude for non-targeted microbubbles) to induce bioeffects. Bioeffects include cell membrane pore formation (i.e., sonoporation), which facilitates intracellular drug delivery, and opening of cell–cell contacts, which facilitates transendothelial drug delivery. Interestingly, we found no clear distinguishment in microbubble behavior that predicts the occurrence of sonoporation only or sonoporation plus cell–cell contact opening. However, the cytoskeleton F-actin organization within cells was found crucial for opening cell–cell contacts by oscillating microbubbles [Meijlink *et al*., J. Control. Release. 376, 1176–1189 (2024)]. At the same time, higher peak negative acoustic pressures were needed to induce transendothelial drug delivery than sonoporation in a microvessel-on-chip model, both for 10 and 1000 cycles of ultrasound (2 MHz) [Meijlink *et al*., Small 2407550 (2024)]. Taken together, our findings suggest that dose monitoring is feasible, which may be complicated by the cellular landscape.

Contributed Paper

9:00

2aBAa4. Bubbles behaving badly—Dose versus threshold in cavitation classification. Qiang Wu (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Cameron A. Smith (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Luca Bau (Univ. of Oxford, Oxford, United Kingdom), Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom), Robin O. Cleveland (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

The acoustic emissions generated by oscillating gas bubbles (cavitation) provide an extremely useful means of treatment monitoring in ultrasound-mediated therapy. Metrics such as the relative proportions of harmonics and broadband noise in the acoustic emissions frequency spectra are widely used to categorize cavitation, e.g., as "inertial" or "stable." It is unclear, however, whether these simple categories can adequately describe the wide range of bubble dynamics that may occur or the effects that these may produce in tissue. We have previously shown through simultaneous capture of high-speed video footage and acoustic radiation that even for single bubbles there is no simple correlation between different types of bubble behavior and the frequency content of the acoustic emissions. This suggests that measures such as the onset of broadband noise, or the appearance of sub- or ultra-harmonics are not appropriate for use as universal thresholds in therapeutic ultrasound. Binary categories such as stable and inertial cavitation, or the use of spectral characteristics to infer these, should similarly be avoided. In this talk, the implications of these findings for defining a cavitation dose and applying it to different types of therapy will be discussed.

9:20–9:40 Break

Invited Papers

9:40

2aBAa5. Bubble, bubble, sonic trouble: Cavitation dose and therapeutic close. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Daniel Suarez Escudero (Cardiawave, Levallois-Perret, France), Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586, kevin.haworth@uc.edu), and Curtis Genstler (Boston Sci., Snohomish, WA)

Round about the phantom flow, Infused microbubbles go.
With pulses varied and rarefied, In cavitation's course we bide.

Double, double toil and trouble;
Pressure rise and bubbles rumble.

Energy from echoes seen, Mapped in colors, red and green.
Rarefaction's peak doth tell, Stable hum or inertial spell.
Long or short, the pulse doth play, Cavitation marks the way.

Double, double toil and trouble;
Pressure rise and bubbles rumble.

Metrics dense of dose defined, Energy coursing, fate entwined.
Stable charms or collapse to see, Impacting efficacy.
For venous flow or clots to break, Future schemes for patients' sake.

Double, double toil and trouble;
Pressure rise and bubbles rumble.

Cavitation with image masks, the spell complete,
For healing tasks, the charm is sweet.
To delve the depths of knowledge nigh,
Seek thou this DOI: https://doi.org/10.1016/j.ultrasmedbio.2023.08.002.

10:00

2aBAa6. Cavitation radiated energy density (CRED): An energy-preserving spatiotemporal cavitation dose metric. Cameron A. Smith, Luca Bau, Abigail Collins, Darcy M. Dunn-Lawless, Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7LD, United Kingdom), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, 54 Franklin Rd., Headington, Oxford, Oxfordshire OX3 7SA, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Passive Acoustic Mapping (PAM) makes it possible to qualify and quantify the spatio-temporal distribution of cavitation activity during therapeutic ultrasound treatments, in addition to providing a valuable tool for real-time treatment monitoring and cavitation imaging. To enable meaningful correlation of cavitation activity with the safety and efficacy of associated bioeffects, a cavitation dose metric is required that is energy-preserving, device-independent and adequately normalized to the tissue volume being affected by the reported

cavitation activity. We introduce cavitational radiated energy density (CRED), defined as the total energy of either narrowband or broadband radiated acoustic emissions over a volume, divided by that volume. We show that image blurring caused by the instrumentation-dependent point spread function introduces significant image artifacts and CRED errors when using existing conventional and adaptive beamformers, and propose a novel PAM algorithm that utilizes the Lucy–Richardson deconvolution (LRD) technique to compensate for the point-spread function and thus provide cavitation maps with reduced tail artifacts and improved energy estimates in a computationally efficient manner. Using a combination of modeling and experimental approaches, we demonstrate the quantitative significance of CRED and the potential of the LRD-PAM algorithm to enable energy-preserving quantitative real-time monitoring of cavitation-based therapies, independently of the instrumentation employed.

10:20

2aBAa7. Measure for measure: Diffraction correction for consistent quantification of bubble-related acoustic emissions. T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

To characterize and improve ultrasound-mediated therapies, system-independent measures of cavitation activity are needed. Here, an approach is described to quantify acoustic emissions from ensembles of cavitating bubbles, as measured by passive cavitation detection or imaging (PCD or PCI). An analytic diffraction-correction factor relates measured frequency-dependent pressure to cavitation-radiated power or energy per unit area or volume of a region of interest (ROI). This approach is illustrated via previous experiments modeling two therapeutic ultrasound scenarios. For diffraction-corrected PCD measurements during sonophoresis of *ex-vivo* porcine skin [Rich *et al.*, JASA 144, 3563–3574 (2018)], radiated subharmonic acoustic power per unit ROI area is highly correlated with decreases in skin electrical resistance ($r = 0.823$), across sonication frequencies (0.41 and 2.0 MHz), sonication amplitudes, and degassing conditions. For thermal bulk ablation of liver tissue by unfocused ultrasound [Karunakaran *et al.*, UMB 47, 2360–2376 (2021)], measured thermal lesion dimensions are related to PCD-measured radiated powers and energies of subharmonic and broadband acoustic emissions per unit ROI volume across two sonication conditions (3.1 MHz, 20 min; 4.8 MHz, 10 min) and three overpressure conditions modulating bubble activity. Implications are discussed for defining appropriate cavitation doses based on the therapeutic application and PCD or PCI configuration.

10:40

2aBAa8. Sound and fury, signifying something? Technical challenges in cavitation dosimetry. Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, michael.gray@eng.ox.ac.uk)

A number of factors impede the pursuit of reliable and meaningful dosimetry for cavitation-mediated therapies. This presentation will review major technical challenges in the measurement of biomedical cavitation, along with the resulting uncertainties in cavitation dose calculation across several treatment modalities. Mitigation methods for propagation path, receiver, and processing related effects will then be summarized. Finally, an optimistic outlook for this field will be highlighted based on progress made to date and lessons learned from other areas of medical dosimetry.

Session 2aBAb

Biomedical Acoustics, Computational Acoustics, Signal Processing in Acoustics, Physical Acoustics, and Engineering Acoustics: Technological Developments and Emerging Biomarkers in Elasticity Imaging I

Javier Brum, Cochair
Laboratorio de Acústica Ultrasonora, Instituto de Física, Facultad de Ciencias, Universidad de la República, Igua 4225, Montevideo 11400, Uruguay

John M. Cormack, Cochair
Department of Medicine, University of Pittsburgh, Division of Cardiology, Pittsburgh, PA 15261

Zhiyu Sheng, Cochair
Department of Medicine, University of Pittsburgh, 3550 Terrace Street, Scaife Hall 624, Pittsburgh, PA 15213

Chair's Introduction—8:35

Invited Papers

8:40

2aBAb1. Three-dimensional rotational shear wave elastography for muscle characterization: Opportunities and challenges. Siddhartha Sikdar (Bioengineering, George Mason Univ., 4400 University Dr., MS 1J7, Peterson Hall, Rm. 3303, Fairfax, VA 22030, ssikdar@gmu.edu), Matin Jahani, Cristian Rios (Bioengineering, George Mason Univ., Fairfax, VA), Seiyon Lee, William Rosenberger (Statistics, George Mason Univ., Fairfax, VA), Jay Shah (Rehabilitation Medicine, National Institutes of Health, Bethesda, MD), and Lynn Gerber (Medicine, INOVA Health System, Fairfax, VA)

There is great interest in characterizing the material properties of heterogeneous transversely anisotropic media such as muscle and associated connective tissue for a number of clinical conditions. For example, alterations in tissue mechanical properties are a hallmark of many chronic musculoskeletal pain disorders, such as neck and lower back pain, which are leading causes of disability worldwide. Current diagnostic criteria are based on subjective physical examination, and there is a pressing need to develop reliable and objective image-based biomarkers. Ultrasound shear wave elastography (SWE) is an attractive modality due to its suitability for use in community-based pain management clinics. However, the complex architecture and heterogeneous material properties of muscle make it challenging to develop ultrasound-based biomarkers. This talk will describe the use of 3-D rotational SWE as a method for quantifying the anisotropic material properties of muscle. In this method, the ultrasound transducer is mechanically rotated at different angles and SWE images are acquired at each angle to develop a 3-D representation of the anisotropic shear wave propagation speed. Results from a recent clinical study of 90 subjects with and without chronic musculoskeletal neck pain highlight the opportunities and challenges associated with 3-D rotational SWE in this clinical application and the need for future research in this area.

9:00

2aBAb2. Elastography applications of surface waves: Examples in biomechanics and in the beef industry. Nicolás Benech (Instituto de Física, Facultad de Ciencias, Universidad de la República, Uruguay, Igua 4225, Montevideo 11400, Uruguay, nicolas.benech@fcien.edu.uy)

The use of surface waves for estimating shear elasticity in soft tissues is an emerging method. There are numerous examples of *in vivo* and *ex vivo* applications in literature. Most of these examples base their inversion methods on Rayleigh surface waves, Rayleigh–Lamb dispersion curves, or even the Timoshenko beam model. In this work, we show that there is a critical distance from the source where near-field effects prevent the use of these models. This distance is given by the interference of the Rayleigh surface wave with the so-called leaky surface wave, which arises as a complex solution of the Rayleigh secular equation for materials with large Poisson's ratio [Benech *et al.*, JASA 142(5), 2017]. This leaky wave has been observed experimentally in geophysical applications, and here we show its existence in tissue-mimicking phantoms [Benech *et al.*, J. Phys. Condens. Matter, 34 (2022)]. We then show that a similar behavior is observed in transversely isotropic solids such as skeletal muscle. Finally, we demonstrate two different applications. First, in estimating the elasticity of the concentric contraction of the biceps brachii *in vivo*. Second, in monitoring the enzymatic maturation process by aging in beef samples. We highlight the advantages of surface waves over traditional ultrasound elastography in these two applications.

Contributed Paper

9:20

2aBAb3. Second-harmonic generation in focused shear wave beams in tissue-like media with arbitrary amplitude shading at the source. Philip G. Kaufinger (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78766-9767, pkaufinger@utexas.edu), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Second-harmonic generation in a focused shear wave beam with Gaussian amplitude shading and radial polarization can be described analytically using a perturbation solution [Kaufinger *et al*., JASA 155, A350 (2024)]. Recently, focused radially polarized shear wave beams were generated in tissue-mimicking gelatin phantoms with an oscillating concave circular piston [Cormack *et al*., IEEE TBME 71, 621 (2024)]. To model second-harmonic generation in a focused shear wave beam with source conditions more relevant to the experiments, i.e., without Gaussian amplitude shading, the nonlinear evolution equation is solved numerically by successive approximations in k-space using a fourth-order Runge–Kutta stepping scheme. The linear solution is obtained first, which is then substituted into the quadratic terms to solve for the nonlinearly generated second harmonic. Emphasis is placed on how more realistic source conditions, including super-Gaussian amplitude shading or source conditions estimated from measurements, affect the generation of the second harmonic. Attention is also devoted to the longitudinal displacement field that is calculated locally from the transverse displacement field and the assumption of material incompressibility. [PGK is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

9:40–10:00 Break

Invited Paper

10:00

2aBAb4. Wave propagation in prestressed and transversely isotropic viscoelastic structures: Inverse modeling challenges in elasticity imaging. Alexandra Vorobyeva, Dieter Klatt (Univ. of Illinois Chicago, Chicago, IL), Kenneth Shull, Eric J. Perreault (Northwestern Univ., Evanston, IL), and Thomas Royston (Univ. of Illinois Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, troyston@uic.edu)

The functional role and structure of skeletal muscle results in anisotropy in both material properties and imposed stresses, as well as waveguide effects. Dynamic elastography reconstruction methods for estimating muscle tissue viscoelastic properties that are rooted in assumptions of isotropy and bulk wave motion may produce inaccurate estimates. The superposition of axially aligned orthotropy (transverse isotropy) in material properties and axially aligned prestress conditions due to passive stretch or muscle activation makes it difficult to independently discern how much of the apparent anisotropy is due to the muscle material or the imposed stress field. Furthermore, this stress field may result in large strain conditions that require the use of higher-order terms in the stress–strain relationship. The significance of these confounding conditions and strategies for decoupling material and stress-based anisotropy are investigated with a series of numerical finite element studies based on simple and morphological image-informed geometries, and experimental elastography studies using scanning laser Doppler vibrometry and magnetic resonance elastography.

Contributed Papers

10:20

2aBAb5. Three-dimensional vibration-controlled transient elastography in human subjects with myofascial trigger pain points. Maryam Satarpour (Bioengineering, Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15621, MAS1338@pitt.edu), John M. Cormack, Zhiyu Sheng (Medicine, Univ. of Pittsburgh School of Medicine, Pittsburgh, PA), Yu-Hsuan Chao (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Allison Bean, Ryan Nussbaum (Physical Medicine and Rehabilitation, Univ. of Pittsburgh School of Medicine, Pittsburgh, PA), Jiantao Pu (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Ajay D. Wasan (Anesthesiology & Perioperative Medicine, Univ. of Pittsburgh School of Medicine, Pittsburgh, PA), and Kang Kim (Bioengineering and Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Myofascial pain is associated with chronic lower back pain (LBP) and is a global health problem that affects almost all adults at some point in their lives. The myofascial components of chronic LBP, including trigger pain points (TPP) that are characterized by pain radiating from these points to broader areas, are difficult to diagnose due to a lack of reliable imaging biomarkers. Here, we present preliminary measurements of three-dimensional vibration-controlled ultrasound transient elastography to explore changes in muscle stiffness that accompany myofascial TPP. We scanned 16 subjects with identified myofascial TPP and 16 without any pain points. Elastic waves were excited by applying vibrations to the skin surface using a wide bar driven by a 150-Hz tone burst at four specific locations, including the multifidus and erector spinae muscles, bilaterally at the level of L3 and L4 vertebrae. Propagating waves were captured with the 3-D volumetric image using a row-column array transducer (RCA, Vermon RC6gV, 6 MHz). The propagation speed of the traveling wave was calculated throughout the muscle volume using a time-of-flight approach. Preliminary comparisons suggest that propagation speeds may be higher in lower back muscles with TPP, indicating the potential diagnostic value for myofascial components in chronic LBP.

10:40

2aBAb6. Elastography in fiber-laden materials: Impact of homogenization assumptions. Lara Nammari (Univ. of Illinois Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, lnamma2@uic.edu), Dieter Klatt, and Thomas Royston (Univ. of Illinois Chicago, Chicago, IL)

The functional role of skeletal muscle and the hierarchal microstructure and arrangement of fibers within it result in anisotropy and inhomogeneity in both material properties and imposed stresses. Dynamic elastography reconstruction methods for estimating muscle tissue viscoelastic properties that are based on assumptions of homogeneity, isotropy and only bulk wave

motion may produce inaccurate estimates. Biases may be introduced in reconstruction by homogenizing muscle with axially aligned fibers and approximating it as transversely isotropic. The significance of these biases, and their interplay with imposed stresses and confounding waveguide effects due to small cross-sectional dimensions, is quantified with a series of numerical finite element and experimental elastography studies on fiber-laden phantoms, with varying fiber dimensions. This study reveals how accurate homogenization is informed not only by cross-sectional area or volumetric information but depends on a more detailed multiscale understanding of the heterogeneous material structure.

TUESDAY MORNING, 20 MAY 2025 STUDIO 6, 7:55 A.M. TO 11:00 A.M.

Session 2aCA

Computational Acoustics, Structural Acoustics, and Vibration and Physical Acoustics: Computational Methods for Nonlinear Problems in Acoustics and Vibration

John S. Allen, Cochair
Mechanical Engineering, University of Hawaii Manoa, Holmes 302, 2540 Dole Street, Honolulu, HI 96822

Martin D. Verweij, Cochair
Imaging Physics, Delft University of Technology, Lorentzweg 1, Delft 2628 CD, Netherlands

Samuel P. Wallen, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet RD, Austin, TX 78758

Chair's Introduction—7:55

Invited Papers

8:00

2aCA1. Simulating ultrasound propagation through a contaminated nanobubble population. Nicholas Ovenden (Dept. of Mathematics, Univ. College London, London WC1E 6BT, United Kingdom, n.ovenden@ucl.ac.uk), Mihir Sheth, Qiang Wu (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom)

Recent papers have reported the ability of nanobubbles to exhibit a comparable or superior acoustic response to microbubbles. Such a claim is significant, as the ability of nanobubbles to extravasate, while remaining visible to conventional ultrasound imaging, could lead to vast applications in terms of diagnostic and therapeutic treatments. The paradoxical performance of these nanobubble populations is, however, disputed by classical acoustic theory, indicating that only nonlinear propagation effects in the harmonic content could be responsible for the higher than expected contrast:background ratio observed. An alternative explanation could be that the large scattering detected is caused by a small proportion of larger microbubbles within the nanobubble suspension. In this study, we test these ideas by adapting an efficient fully nonlinear model of wave propagation through bubbly media to model ultrasound propagation through a relatively high concentration of coated nanobubbles and resolve the harmonic content in the backscattered signal. Such computations require an efficient method of computing the radial dynamics of each bubble size present at every spatial location at each time step while incorporating a nonlinear shell that rapidly buckles and ruptures during large oscillations. The theoretical results presented will be discussed together with recent experimental data.

8:20

2aCA2. Evolution of the Iterative Nonlinear Contrast Source method for simulating nonlinear ultrasound: From still water to sparkling water. Martin D. Verweij (Imaging Phys., Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CD, Netherlands, M.D.Verweij@tudelft.nl)

For many medical ultrasound modalities, the inherent nonlinearity of acoustic waves cannot be neglected or is even of prime importance. The original Iterative Nonlinear Contrast Source (INCS) method was developed to accurately simulate the nonlinear propagation of a pulsed ultrasound beam in a large, three-dimensional domain. As a key step, the nonlinear term in the relevant wave equation is considered to describe a contrast source in the linearized background medium. Convolution of the contrast term with Green's function of the linear background yields the nonlinear field contribution. The resulting integral equation is solved by an iterative scheme. The INCS approach was subsequently extended by introducing contrast sources that account for other realistic medium properties, like frequency-dependent attenuation and inhomogeneity of tissue. The most recent extension is the inclusion of microbubble contrast agents. To model the propagation of an ultrasound wave through a large population of microbubbles, each microbubble is represented as a point contrast source. The strength of each point source depends on the bubble radius, which is obtained by solving each bubble's individual Rayleigh–Plesset equation. In this presentation, the versatility of INCS will be demonstrated by discussing results that have been obtained for various types of contrast sources.

8:40

2aCA3. Data-driven acoustic control of an encapsulated microbubble using a Koopman linear quadratic regulator. Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu)

Encapsulated microbubbles (EMBs) are used in biomedicine for both diagnostic and therapeutic purposes that include ultrasound imaging and drug delivery. In this work, we present a data-driven method to acoustically control the oscillations of EMBs using Koopman operator theory, which is a method for transforming nonlinear dynamical systems into linear systems on an infinite-dimensional function space. This linearization allows classical linear control methods to be applied to the underlying nonlinear dynamical system. Here, we apply a Koopman linear quadratic regulator (KLQR) to control the oscillations of a spherical EMB based on the Marmottant model through the applied ultrasound. The control is confounded by the presence of a slow manifold that arises in the phase plane spanned by the bubble radius and radial velocity, which requires special care in formulating the KLQR controller through the judicious selection of Koopman eigenfunctions. Results are presented that demonstrate the effectiveness of the modified KLQR controller in driving an EMB to follow arbitrarily prescribed radial oscillations and stabilize at nonequilibrium radii.

Contributed Papers

9:00

2aCA4. A hybrid immersed boundary-lattice Boltzmann method solver for modeling nonspherical encapsulated microbubbles. Morteza Garousi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, Colorado Springs, CO) and Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu)

Encapsulated microbubbles (EMBs) have biomedical applications that include ultrasound imaging and targeted drug delivery. EMBs are modeled as a gas core surrounded by a thin, viscoelastic coating immersed in a viscous liquid. Thus, numerical simulation of the EMB response to acoustic forcing is a complex multiphase flow problem that involves fluid–structure interaction (FSI). Upon insonation, an EMB can undergo both spherical (volumetric) and nonspherical (shape) oscillations. In this work, we couple the lattice Boltzmann method (LBM) with the immersed boundary (IB) method to numerically simulate large, axisymmetric, nonspherical deformations of EMBs subject to acoustic forcing. A multicomponent multiphase LBM is used to solve for the fluid dynamics of the interior and exterior fluid phases. The IB method accounts for the FSI between the fluid phases and the encapsulation by explicitly tracking Lagrangian markers placed on the interface. The force exerted on the surrounding fluids by the EMB coating is incorporated into the hybrid IB-LBM solver using a viscoelastic constitutive model. Simulations of the EMB response to step and sinusoidal variations in far-field pressure using the IB-LBM solver are validated against the modified Rayleigh–Plesset equation and other benchmarks. The accuracy of the IB-LBM model is analyzed with respect to the stencil choice for the kernel function used for velocity interpolation and force spreading, and also with respect to the choice of time integration scheme for advecting the EMB surface.

9:20–9:40 Break

9:40

2aCA5. Large-scale aeroacoustic simulation and non-linear 1-D modeling of bass-reflex loudspeakers. Ryoya Tabata (Yamaha Corp., 10-1, Nakazawacho, Naka-Ku, Hamamatsu-shi 4300904, Japan, ryoya.tabata@music.yamaha.com), Katsuya Uchida, Yuko Okada, Yoshikazu Honji (Yamaha Corp., Hamamatsu-shi, Japan), and Kin'ya Takahashi (Res. Inst. for Information Technol. (RIIT) of Kyushu Univ., Fukuoka, Japan)

In a bass-reflex loudspeaker, a port (or vent) enhances the low-frequency response through Helmholtz resonance. However, at high sound pressure levels, the airflow near the port tip can generate vortices, leading to noise and distortion. This paper introduces an aeroacoustic computational method to simulate the noise and distortion in bass-reflex loudspeakers, with results showing reasonable agreement with experimental data. The aeroacoustic computational results, based on large eddy simulation, demonstrate that considering the three-dimensional nature of vortices is essential for accurately predicting the associated harmonic distortion and noise. Furthermore, a one-dimensional modeling approach is introduced to capture the nonlinear distortion caused by the vena-contracta effect in the port, which qualitatively matches experimental results. The 1-D modeling results show that accounting for the asymmetry of the flow at the port tips is essential for reproducing even-order distortion.

10:00

2aCA6. Dynamical analysis and computational research on capillary flow of gallium-based liquid metal. Jiao Yu (Liaoning Petrochemical Univ., 1 Dandong Rd. West Section, Fushun, Liaoning 113001, China, yujiaojoy@hotmail.com), Chuanyang Jiang (Peking Univ., Beijing, China), Sheng Yang (Beijing Univ. of Technol., Beijing, China), and Shiqiang Ren (Liaoning Petrochemical Univ., Beijing, China)

Describing the capillary flow of liquid metals is key to answering many biomedical and industrial questions. In previous research [Jiang *et al*., APL Mater. 12, 121119 (2024)], we developed a dynamical model based on the momentum theorem to quantitatively evaluate the capillary flow of gallium-based liquid metal. In the dynamical model, the surface tension, the contact angle, the hydrostatic pressure, and the dynamic viscosity are all considered in analyzing the nonlinear flow behavior of liquid metal. This paper continues to explore the two-phase flow modeling for the evaluation of the flow behavior of the eGaIn–HCl fluid pair in the copper capillary tube. The computational model tracks the dynamic changes in the interfacial displacements of the liquid metal column with respect to time in capillary tubes of varying lengths and cross-sectional geometries and the results are validated through the comparison with experimental measurements. [Work supported by the National Natural Science Foundation of China (Grant No. 12074160) and the Natural Science Foundation of Liaoning Province of China (Grant No. 2024-MS-181)]

10:20

2aCA7. Acoustic simultaneous localization and mapping for drone navigation in complex environments. Hala Abualsaud (ECE, Univ. of California San Diego, 10645 Calle Mar de Mariposa, Apt 6205, San Diego, CA 92130, habualsa@ucsd.edu) and Peter Gerstoft (ECE, Univ. of California San Diego, San Diego, CA)

This paper introduces a framework for 3-D drone localization and sound source mapping using simultaneous localization and mapping and acoustic

techniques. The approach integrates the quaternion method for orientation tracking with a closed-form solution combining the Time Difference of Arrival and Angle of Arrival methods. A drone equipped with three microphone arrays and an inertial measurement unit that captures the drone's linear acceleration and angular velocity simultaneously maps stationary sound sources and estimates its own trajectory. Weighted least squares optimization refines source positions, while particle filtering enhances accuracy in dynamic settings. Simulations show that the method outperforms conventional techniques like dead reckoning, presenting accurate localization and mapping even in complex scenarios. This work highlights the potential of sound-based mapping in challenging scenarios where traditional visual methods fail, such as environments with poor visibility, like underwater or extreme acoustic interference.

10:40

2aCA8. Coupled vibro-acoustic simulation for noise mitigation and reverberation time control in enclosures. Solomon O. Ologe (Mech., Univ. Polytechnic of Catalonia, Calle de Colom, 11, Tarrassa Campus, Barcelona 08222, Spain, ologe.solomon@upc.edu)

Achieving optimal acoustic comfort in enclosed spaces requires understanding the interaction between structural vibrations, noise propagation, and reverberation characteristics. This study employs a coupled vibro-acoustic simulation to predict noise levels and analyze reverberation time in response to structural excitations. Key frequencies and structural configurations that exacerbate noise and increase reverberation time are identified. The research evaluates various noise mitigation strategies, such as damping treatments and structural reinforcements, and their effects on reducing reverberation time. By integrating predictive modeling with analytical validation, the findings provide actionable insights for enhancing acoustic performance in vessels, automotive, building, and industrial applications, focusing on comprehensive noise and reverberation control.

TUESDAY MORNING, 20 MAY 2025 — BALCONY N, 7:35 A.M. TO 11:05 A.M.

Session 2aED

Education in Acoustics: Acoustics Around the World—Part 1: Education Programs at Universities

Likun Zhang, Cochair
National Center for Physical Acoustics and Department of Physics and Astronomy, University of Mississippi, 145 Hill Drive, University, MS 38655

Joao Ealo, Cochair
School of Mechanical Engineering, Universidad del Valle, Ciudad Universitaria Meléndez, Building E49—Of.2014, Cali 760032, Colombia

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—7:35

Invited Paper

7:40

2aED1. Acoustics education to support the future workforce. Jennifer Miksis-Olds (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu)

In 2022, the Office of Naval Research initiated a project through the National Academies of Sciences, Engineering, and Medicine to assess the current and future ocean acoustics education and workforce landscape over the next decade. The report published in 2024—Ocean Acoustics Education and Expertise—captures the continued and still growing value of ocean acoustics to defense, while also highlighting the expanding need for acoustics education in policy, navigation, economic, industry, and environmental sectors. Although the report findings and recommendations were developed from the ocean acoustics education perspective, the content is common to all sub-disciplines of acoustics and highlights challenges to acoustics and interdisciplinary STEM education at large. The report describes how the multidisciplinary nature of acoustics and the low visibility of acoustics career paths have contributed to shortfalls in workforce recruitment and retention. The presentation will provide a historical perspective on how the ocean acoustics and the greater acoustics

community evolved to its current condition, discuss existing ocean acoustics education and training opportunities, identify current and projected workforce needs, share recommendations to recruit and retain a diverse workforce, and strategize on future alignment of training and education components with the projected workforce.

Contributed Papers

8:00

2aED2. Underwater acoustics and ocean engineering at the University of Rhode Island. Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu), James H. Miller, Gopu R. Potty, and Jennifer Amaral (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

The University of Rhode Island is home to the first ocean engineering program in the country and offers both undergraduate and graduate programs focusing on underwater acoustics. All ocean engineering students in the ABET-accredited undergraduate program take a course in underwater acoustics and many are involved in acoustics-focused capstone projects and research. The graduate program offers PhD degrees as well as both thesis and non-thesis Masters degree options. URI is now also offering a Graduate Certificate in Underwater Acoustics. Graduate students are actively involved in research focused on areas such as acoustical oceanography, propagation modeling, geo-acoustic inversion, marine mammal acoustics, ocean acoustic instrumentation, transducers, and signal processing. The program is based at URI's Narragansett Bay Campus, where students have direct access to Narragansett Bay, and which is currently undergoing a renovation that will include state-of-the-art acoustic tank facilities.

8:20

2aED3. Graduate programs in physical acoustics at the University of Mississippi: Choose physics or engineering. Cecille Labuda (Phys. and Astronomy, Univ. of MS, 108 Lewis Hall, University, MS 38677, cpembert@olemiss.edu), Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, University, MS), Joel Mobley (Phys. and Astronomy, Univ. of MS, University, MS), and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

The University of Mississippi provides a unique opportunity to study physical acoustics in two departments: the Department of Physics and Astronomy and the Department of Mechanical Engineering. It is one of the few universities in the United States where students can pursue PhD and MS degrees in Physics with a focus on physical acoustics research. The National Center of Physical Acoustics (NCPA) is an 85 000 square foot standalone facility that houses many laboratories on the campus of the University of Mississippi solely dedicated to the physics and engineering applications of acoustics over the entire frequency range, from infrasound to ultrasound. Students can study a wide range of topics including the atmosphere, ocean, weather, jet noise, porous media, acoustic metamaterials, ultrasonic properties of materials, nondestructive testing, and fluid dynamics. Our alumni work in academia, national labs, industry, and the medical field among other occupations. The University of Mississippi is a PhD granting institution with an R1 Carnegie designation placing it among schools with the highest level of research activity. It is located in Oxford, MS, a cultural mecca of the southern United States, home to William Faulkner (one of the most celebrated American authors), many artistic festivals and sporting events.

8:40

2aED4. How to have an acoustics program without having an acoustics program. Jeff Foeller (Dept. of Eng., East Carolina Univ., Greenville, NC, foellerj@ecu.edu), Matthew Stengrim (Dept. of Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC)

The East Carolina University Department of Engineering is a small program housed in a large public university. The department offers a general engineering undergraduate degree as well as MS programs in Biomedical and Mechanical Engineering. Faculty in the department have engaged in atmospheric acoustics and vibrations research since 2014 and the program has placed multiple graduates in acoustics or acoustics-adjacent fields. This is happening even without formal undergraduate or graduate acoustics courses in the university course catalog largely due to the integration of undergraduate students in the experimental research efforts. This research involvement typically serves as the gateway or "hook" for students seeking additional opportunities to study acoustics. These additional opportunities have been offered in various ways: independent study for class credit, special topics elective courses, Capstone projects, along with foundational summer research experiences. This presentation explores these offerings and reflects upon the success and pitfalls of these methods. [Work supported by Office of Naval Research Award N00014-22-1-2492.]

9:00–9:20 Break

9:20

2aED5. Acoustics education for speech-language pathology courses in Japan. Takayuki Arai (Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

Speech-language-hearing therapists are nationally licensed professionals specialized in speech and language pathology in Japan. Acoustics is one of the required areas of study, and education in acoustics is usually taught by experts. Students are taking courses in related topics, such as acoustic phonetics and speech science. Because students have different academic backgrounds, it is always a challenge for instructors to teach them acoustics. For example, the concept of "source and filter" in speech production is one area students sometimes struggle to understand. Using a set of physical models of the human vocal tract with various sound sources is an intuitive way to assist students' understanding. In addition, videos of real physical phenomena and computer-based animations are also helpful. Finally, an education program in acoustics for such students with those tools is proposed. [This work was supported by JSPS KAKENHI Grant Nos. 24K06423 and 24K06363.]

9:40

2aED6. Acoustics at the University of Massachusetts Dartmouth. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsdb@gmail.com), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), and Paul J. Gendron (ECE, Univ. of Massachusetts Dartmouth, Dartmouth, MA)

The University of Massachusetts Dartmouth (UMassD) offers Acoustics through various graduate courses associated with the Master of Science and PhD degrees in Electrical Engineering. The focus is on applied underwater acoustics, with emphasis on signal processing and analysis, transduction and sensors. Course offerings include Fundamentals of Acoustics, Acoustics and Electromagnetic Waves, Underwater Acoustics, Electroacoustic Transduction, Medical Ultrasonics, Array Processing, Communications, Detection and Estimation. UMassD has unique facilities including an Underwater Acoustic Test Tank, Open-Ocean water access, and Unmanned Underwater Vehicles that support our graduate projects. The expertise to be gained reaches from acoustic array design, environmental monitoring and statistical inference. Many of the alumni go on to work at US Navy laboratories, nonprofit research centers, as well as industry and small companies focusing on marine technology. Acoustics has been offered at UMassD for over five decades and we encourage applicants from diverse STEM backgrounds to apply to our graduate program.

10:00

2aED7. Architectural acoustics and noise control studies at the University of Nebraska's Architectural Engineering program. Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska– Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

The University of Nebraska–Lincoln (UNL) instituted an Architectural Engineering (AE) program in 1999 and celebrated its 25th anniversary last year. Currently, there are only 39 ABET-accredited AE programs around the world, and very few of these include acoustics as a primary area of study in their curricula. Based in the city of Omaha, the Nebraska AE program has trained students in architectural acoustics and noise control from the program's inception, offering opportunities to study acoustics within its multiple degree options (Bachelor of Science in Architectural Engineering, Master of Architectural Engineering, Master of Science in Architectural Engineering, and Doctor of Philosophy in Architectural Engineering). This presentation reviews the UNL AE program's current acoustic courses, research interests, and facilities. Also highlighted are program alumni from the past quarter century. One unique aspect of the Nebraska AE program is that it is the only program to have received the $25 000 grand prize from the National Council of Examiners for Engineering and Surveying (NCEES) Engineering Education Award multiple times in the past decade.

10:20

2aED8. A Master's degree in acoustics online: The acoustics distance education program at Penn State. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

In 1987, the Graduate Program in Acoustics at Penn State offered its first distance education (DE) graduate-level acoustics course via satellite TV to a group of 45 students at the Naval Undersea Warfare Engineering Station (NUWES) in Keyport, WA. Since that first live broadcast in 1987 the methods of course delivery have changed drastically, from one-way satellite TV to Zoom broadcasts from a hybrid multimedia classroom. Our student population has grown from 45 students at one location to more than 100 mostly non-military students each semester from locations and companies across the U.S. and around the world. To date, the program has awarded nearly 300 M.Eng. degrees in Acoustics to students from a wide variety of backgrounds. This talk will describe the acoustics DE program in detail, including highlights from the DE program's history, technological challenges for course content delivery, the divergence of student demographics, methods of instruction and delivery to a combined hybrid enrollment of both resident and remote students, the current curriculum for the M.Eng. degree, including a new offering of certificates, and plans for the future.

10:40–11:05 Panel Discussion

TUESDAY MORNING, 20 MAY 2025 BISSONET/CARONDELET, 11:10 A.M. TO 12:00 NOON

Session 2aID

Interdisciplinary: Plenary Lecture: Acoustics and Wave Physics in Modern Applications of Ultrasound in Therapy

Vera A. Khokhlova, Chair
University of Washington/Moscow State University, Physics Faculty, Moscow State University, Moscow 119991, Russian Federation

Chair's Introduction—11:10

Invited Paper

11:15

2aID1. Acoustics and wave physics in modern applications of ultrasound in therapy. Oleg A. Sapozhnikov (Moscow State Univ., Moscow, Russia, and Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, olegs2@uw.edu)

Ultrasound methods and devices have become widespread in modern medicine for diagnostics and therapy. Therapeutic applications are increasingly common, including tumor ablation, kidney stone fragmentation, targeted drug delivery, physiotherapy, and essential tremor. The success of ultrasound methods has relied not only on engineers and physicians but also on many years of scientific research in wave physics and acoustics. From an acoustics point of view, medical ultrasound utilizes inaudible high-frequency waves that propagate in water-like media. In the context of wave physics, such ultrasound waves are short-wavelength elastic waves that can be effectively focused, reflected, and refracted. Ultrasound is able to propagate over significant distances and thus reach deep parts of the human body. Many general properties of waves, in particular their ability to carry energy and momentum, are used in medical ultrasound for remote thermal or mechanical destruction of tissue, as well as for providing a pushing effect as a result of the emergence of a radiation force. Key features of the wave field include aberration and refraction when passing through inhomogeneous tissue layers, the diffraction limit of energy localization during focusing, reflection from interfaces, and scattering by inhomogeneities and inclusions.

Session 2aMU

Musical Acoustics: Musical Instruments in Jazz I

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Murray Campbell, Cochair
School of Physics and Astronomy, University of Edinburgh, James Clerk Maxwell Building, Peter Guthrie Tait Road, Edinburgh EH9 3FD, United Kingdom

E. K. Ellington Scott, Cochair
Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Invited Papers

8:20

2aMU1. Placing musical instruments in stereo Jazz recordings to create a transparent and intimate sound image. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu) and E. K. Ellington Scott (Rensselaer Polytechnic Inst., Troy, NY)

Throughout time, Jazz has embraced a transparent sound ideal where individual instruments can be heard clearly, while in classical and pop music, the goal is often to fuse musical instruments into a more holistic soundscape. In the late 1950s, Jazz and classical music started to embrace stereo recordings for transparency (Jazz) and spaciousness (classical) reasons, while pop music focused on mono sound into the 1960s. This presentation focuses on best microphone placement practices to support the transparent jazz sound ideal from pre- and early post-WWII one-microphone mono recordings to elaborate multichannel recordings. The 1950s were an interesting transition period because multiple soloist microphones had to be shared on three available tape tracks. This often resulted in excellent recordings (e.g., Miles Davis's Kind of Blue) because microphone placement and levels had to be balanced carefully. Later on, this was no longer necessary because each microphone was recorded on a separate track. Sometimes, limited technology leads to interesting solutions. For example, the reverberation chamber at Columbia Records was still wired up for mono during the early stereo recordings, and spaciousness was provided by leakage between microphones.

8:40

2aMU2. The use of the steelpan in jazz music. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

The steelpan, which originated from the Caribbean islands of Trinidad and Tobago in the early 20th century, is celebrated for its unique timbre. Understanding the steelpan's complicated note vibrations is essential to being able to explain the instrument's distinctive sound. When crafting tenor steelpans, tuners strive to harmonically tune the first three resonances of each note. Effective tuning results in neighboring notes sharing the same frequency for the first two harmonics on the outer ring of the steelpan. This sympathetic excitation of adjacent notes significantly enhances the steelpan's characteristic sound. While the steelpan is often associated with calypso and reggae music, it has also found a place in various other genres, including jazz. This presentation will explore not only the acoustics of the steelpan but also its origins and its use in jazz music.

9:00

2aMU3. Towards a deeper understanding of steelpan timbre. Randall Ali (Inst. of Sound Recording, Music and Media, Univ. of Surrey, Stag Hill, University Campus, Guildford GU2 7XH, United Kingdom, r.ali@surrey.ac.uk)

With a history tied to post-emancipation traditions and origins in Trinidad, the steelpan musical instrument has had a significant influence on the Caribbean community and its culture. It has even spawned its own brand of jazz called pan-jazz, where it features as a central instrument, merging Caribbean rhythms and melodies with jazz techniques. As steelpans are manually tuned, different steelpans, or even different notes within one steelpan have a unique timbre, a particular characteristic that encourages creative exploration within jazz harmony and improvisation. To attribute factors responsible for the steelpan's timbre, mathematical modeling of notes as mode-localized nonlinear oscillators has proven to be quite insightful. In this talk, we discuss how digital sound synthesis of these mathematical models can lead to a deeper understanding of steelpan timbre as it facilitates the generation of sounds in relation to several changing parameters, helping to define a relevant parameter space for a steelpan sound. The approach underscores the importance of nonlinear mode coupling and mode inharmonicity for defining steelpan timbre.

9:20–9:40 Break

9:40

2aMU4. Indian music tabla bols classification using deep learning. Akshay Kumar (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, Uttar Pradesh, India), Sreerag Ashok (Design, Indian Inst. of Technol. Kanpur, Kanpur, India), and Nachiketa Tiwari (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, India, ntiwari@iitk.ac.in)

Tabla is an important percussion instrument in Indian classical music and is characterized by its unique strokes or a combination of strokes known as "*bols*." Different bols are generated by striking the pair of tabla drums at various locations. Accurate identification of these *bols* is essential for automatic tabla transcription. In this work, we have employed a deep learning-based approach for automatic tabla *bols* classification. We implement and compare different architectures: a Convolutional Neural Network (CNN) and a Convolutional Recurrent Neural Network (CRNN). For the CRNN, we use two variations of RNN layers: GRU and LSTM. The Mel spectrogram is used as the input feature for the CNN model, while MFCC is used for the CRNN model. Various kinds of audio data augmentation techniques are employed to improve model accuracy. Our dataset consists of eight types of bols, which thirteen professional tabla players produce on two different tabla sets. Experimental results show that these methods achieve high classification accuracy and can contribute to developing automated tools for Indian classical music instruments.

10:00

2aMU5. Viscoelastic physical modeling synthesis and machine learning analysis of Jazz drum head tapering. Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, r_bader@t-online.de), Chrisoula Alexandraki (Institut of Technol. and Acoust., Hellenistic Mediterranean Univ., Rethymno, Crete, Greece), Cristiam Martínez (Inst. of Systematic Musicology, Univ. of Hamburg, Hamburg, Germany), and Michael Starakis (Institut of Technol. and Acoust., Hellenistic Mediterranean Univ., Rethymno, Greece)

Jazz drummers often damp their snare drums or tom-toms considerably to shorten the drum sound, allowing the performance of complex rhythmic textures and designing the drum timbre. Tapering involves the use of materials such as Mylar or other plastics, attached to the drumheads using gaffa or similar strong adhesive tapes, applied at different positions and in varying amounts. This leads to a considerable change in the eigenmode spectrum of the drumhead and a complex frequency-dependent viscoelastic damping, which is the main reason for the tapering. Using a viscoelastic physical modeling algorithm, a parameter space of possible drum head tapering is computed by varying position, size, and kind of tapered material. Using deep neural networks, this parameter space is trained and used to identify real tapered drum heads concerning the materials. The presented methodology enables the estimation of tapering in historical drum sets from Jazz recordings, leading to a general understanding of the techniques used to design a desired sound. Additionally, it can assist Jazz drummers in selecting particular tapering methods to achieve a desired drum timbre.

10:20

2aMU6. Timbral effects of the right-hand techniques of jazz guitarists Wes Montgomery and Joe Pass. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 9767, Austin, TX 78766-9767, chiragokani@gmail.com) and Preston S. Wilson (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Wes Montgomery and Joe Pass have left a lasting impact on jazz guitar, thanks in part to the distinctive timbre they achieved through their right-hand techniques. Montgomery played with his thumb, and Pass played both fingerstyle and with a pick. This talk explores the timbral effects of the thumb, fingers, and pick by modeling these sources using mathematical functions that serve as initial conditions for the lossy one-dimensional linear wave equation. Of interest is the sliding between the source and string, the dynamics of which were described by Pavlidou and Richardson [Proc. Ins. Ac. 19, 55–60 (1997)] and Cuzzucoli and Lombardo [Comput. Music J. 23, 52–69 (1999)]. Here, sliding is incorporated kinematically through the sources' time dependence $\exp[-(t+\delta)^2/\tau^2](t+\delta)/\tau$, where δ and τ are the duration of sliding and attack, respectively. Setting δ/τ to 0 and $2^{-1/2}$ describes pure striking and plucking at $t=0$, respectively, while intermediate values $0 < \delta/\tau < 2^{-1/2}$ allow for the exploration of the timbral effects due to sliding. The solutions provide a physical explanation of the timbres of Montgomery and Pass. The talk concludes with a live demonstration of their techniques. [CAG was supported by ARL:UT McKinney Fellowships in Acoustics.]

10:40

2aMU7. The interplay between vowels, pitch targets, and voice quality in singing. May Pik Yu Chan (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing Univ. of Pennsylvania, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu) and Jianjing Kuang (Dept. of Linguist, Univ. of Pennsylvania, Philadelphia, PA)

One strategy in singing is to change the resonance space depending on the target pitch to bridge register shifts. However, the systematic relationship between resonance space adjustments and timbre remains understudied. We investigate whether pitch-dependent vowel modification contributes to voice quality changes. Twenty-three lay speakers participated in an articulatory experiment with ultrasound tongue imaging and electroglottography (EGG). Participants were asked to sing five sets of English vowels across their pitch range in ascending semitone steps. Midsagittal tongue images were splined with DeepLabCut, and closed quotients (CQ) were extracted from the EGG signals. Functional principal component analysis was applied to the tongue splines to evaluate their relationship with CQ. Preliminary results show a nonlinear relationship between voice quality and pitch height, with more modal voice quality in a large portion of the lower end of the pitch range, gradually moving toward a breathier voice quality before becoming tenser again at higher pitches. Tongue position adjustments also contributed to the gradual change in voice quality as the pitch increases; an effect most apparent for participants with a vocal range larger than one octave. Findings highlight the complex interplay between tongue position, voice quality, and pitch targets, showcasing the coordination between source and filter structures.

Session 2aNS

Noise, Physical Acoustics and Computational Acoustics: From Boom to Zoom: Department of Defense and Noise I

James M. Potter, Cochair
Department of the Air Force, Department of Defense, 1260 Air Force Way, Arlington, VA 20330

Erica Rohr, Cochair

Kent L. Gee, Cochair
Department of Physics and Astronomy, Brigham Young University, N281 ESC, Provo, UT 84602

Chair's Introduction—7:55

Invited Papers

8:00

2aNS1. From Boom to Zoom: Department of Defense and Noise. James M. Potter (Dept. of the Air Force, Dept. of Defense, 1260 Air Force Way, Arlington, VA 20330, james.potter.15@us.af.mil)

The Department of Defense's mission demands the application of cutting-edge science in all disciplines. Sometimes science provides DoD a superiority over adversaries. This might be the creation or improvement of weapon systems. Other times, we need to understand the impacts of our actions. Nowhere is this concern for impacts more important than while training within the United States. The National Environmental Policy Act requires that impacts to the human and natural environments be assessed as inputs into the decision process for federal government actions. If impacts are significant, they must be avoided, minimized, or mitigated. A major impact of DoD activity is noise. This overview discusses the variety of sound sources. The performance of those sources is literally life and death, so there are practical limitations on avoiding, minimizing, or mitigating negative acoustic effects. Those effects happen at scales from the individual maintainer or operator to whole communities and regions. And they can cause serious health impacts like hearing loss or simple annoyance. The responses are often conflicting and always challenging. Acoustics science serves them all.

8:20

2aNS2. Boom! Zoom! And everything in between: BYU's involvement in Defense-related noise research. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu)

Noise challenges that the Department of Defense (DoD) faces provide an opportunity for collaborative research into sources, propagation, and reception, which includes methods for measurement and analysis. In this presentation, several DoD-related noise projects conducted at BYU are summarized. These range from military jet noise analysis to measurement of C4 explosions, to machine learning models for ambient noise environments, to reusable rocket flyback sonic booms. Additional research challenges and opportunities for DoD, academia, and industry collaboration are discussed.

8:40

2aNS3. The unique characteristics of things that go BOOM. Michelle E. Swearingen (US Army Engineer Res. and Development Ctr., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Michael J. White (US Army Engineer Res. and Development Ctr., Champaign, IL), and William D. Whiteford (Defense Health Agency, Aberdeen, MD)

Noise from live-fire Army training ranges can travel tens of kilometers and still be loud enough to elicit negative community reactions. An understanding of the source characteristics and acoustic propagation effects is needed to effectively manage noise around the Army's installations. This talk begins with an overview of the unique characteristics of live fire Army training noise. A discussion of the acoustic propagation effects for signals that travel these longer distances follows. It closes with methods for effective management and mitigation of these signals.

9:00

2aNS4. Assessing shooting range noise exposure using acoustic modeling software. Austin Szekacs (US Air Force School of Aerosp. Medicine, Air Force Res. Lab., 2510 North 5th St., Bldg. 840, Wright-Patterson AFB, OH 45433, austin.szekacs@us.af.mil), Alan T. Wall (711th Human Performance Wing, Air Force Res. Lab., Wright Patterson Airforce Base, OH), Gregory Bowers, Deneé Jones (BAE Systems, Inc., Wright-Patterson AFB, OH), and Steven C. Campbell (Air Force Res. Lab., Wright Paterson Air Force Base, OH)

Impulse noise from firearms in shooting ranges presents a unique challenge for assessing the risk to workers' hearing. Firearms produce short-duration, high-level noise that cannot be measured with instruments traditionally used by industrial hygienists. Additionally, acoustic reflections in indoor firing ranges add a level of complexity for modeling not encountered in outdoor, free-field shooting range environments. The Air Force Research Laboratory's 711th Human Performance Wing developed Shooting Range Impulse Noise Calculator, or ShRINC, an acoustic modeling software suite specifically optimized for modeling impulse noise exposure from firearms in indoor firing ranges. ShRINC eliminates the need for specialized data acquisition systems and expands industrial hygienists' capability to assess a common health risk at military installations. We provide an overview of the software, its capabilities, and details of use. We then detail the Air Force's efforts to develop ShRINC, including data acquisition and validation. Finally, we detail the software's implementation in assessing Air Force noise exposure. The Air Force Research Laboratory maintains and distributes ShRINC, which is available with versions for both the Department of Defense and the general public to use for assessing impulse noise exposure at firing ranges.

9:20

2aNS5. Toward the validation of sonic boom propagation models for turbulence effects. Joel B. Lonzaga (Langley Res. Ctr., National Aeronautics and Space Administration (NASA), 2 N. Dryden St. (MS 463), Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

Sonic booms produced by supersonic aircraft are influenced by turbulence within the atmospheric boundary layer (ABL) through which they propagate. These turbulence effects result in random variability of the sonic boom waveforms observed on the ground, making their prediction more complex. This paper utilizes a dataset from a flight test to examine the impact of turbulence on sonic booms. The dataset includes multiple recordings from a linear microphone array. For each supersonic pass, the spectral energy density of the mean waveform and the mean spectral density are calculated, and their ratio is analyzed as a function of frequency. This novel approach helps minimize the impact of uncertainties associated with aircraft flight conditions and large-scale atmospheric conditions. To understand the turbulence effects on sonic booms across different frequencies, various variance and correlation lengths of the fluctuations, along with the ABL thickness, are examined. This approach is then employed to validate two NASA-developed sonic boom models that account for turbulence effects. One model utilizes a nonlinear parabolic equation, while the other, as detailed in Lonzaga, JASA 154, 3078-3088, is based on multiple scattering theory.

9:40–10:00 Break

10:00

2aNS6. Department of Defense Aircraft Noise Models. J. M. Downing (Blue Ridge Res. and Consulting. LLC, 29 N Market St., Ste. 700, Asheville, NC 28801, micah.downing@blueridgeresearch.com) and Juliet A. Page (Blue Ridge Res. and Consulting. LLC, Asheville, NC)

The Department of Defense (DOD) supports the continuing development of the science of noise impact analysis and the application and implementation of scientific principles in environmental impact analysis. This development has led to improvements in aircraft noise models both in terms of accuracy and complexity. This presentation provides an overview of the aircraft noise models used in DOD environmental noise analyses. The overview includes descriptions of the various models and their intended application along with their evolution. The descriptions include the computational approach, reference noise data, input requirements, and output options. The noise metrics generated by each model are described in terms of primary and supplemental metrics. The primary metrics are the main metrics generated by the model and generally are used to describe the basic noise generated by the training activity. The supplemental metrics are used to describe various aspects of the noise environment, and they may need further analysis to calculate from an individual model.

10:20

2aNS7. Noise exposure visualization framework. Mihir Rimjha, Brandon Robinette (Harris Miller Miller & Hanson Inc., Anaheim, CA), and Joseph J. Czech (Harris Miller Miller & Hanson Inc., 300 S. Harbor Blvd, Ste. 516, Anaheim, CA 92805-3717, jczech@hmmh.com)

As communities face increasing exposure to diverse aircraft operations, a framework is needed for operational planning and public engagement. This paper presents an innovative framework for visualizing aircraft noise exposure through animated, three-dimensional representations. By processing Advanced Acoustic Model, Version 3.0 (AAM3) outputs into dynamic Google Earth visualizations, our tool transforms complex acoustic data into intuitive, geospatial displays that effectively communicate noise exposure to communities, planners, and policymakers. Initially validated using Advanced Air Mobility vehicles, the framework's architecture readily accommodates any aircraft modeled in AAM3, from eVTOL aircraft to military platforms like the F-35. The system processes acoustic hemispheres and time-based receptor data, along with high-resolution ground elevation and impedance data, to generate detailed noise-animated visualizations, capturing required A-weighted single-event and cumulative noise metrics.

Contributed Paper

10:40

2aNS8. Analysis of noise radiation asymmetry of the Delta IV Heavy. Noah L. Pulsipher (Phys. and Astronomy, Brigham Young Univ., N 283 Carl F. Eyring Sci. Ctr., Provo, UT 84604, npuls@byu.edu), Kent L. Gee, Levi T. Moats, Grant W. Hart, Logan T. Mathews, and Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The Delta IV Heavy rocket played a critical role in deploying high-profile payloads for the Department of Defense and NASA. Its triple-core, three-engine configuration presents an opportunity to study noise radiation patterns arising from its asymmetrical plume geometry. During NROL-70, the final flight of this historic rocket, acoustic data were collected from 22 locations spanning 0.5–9 km from Space Launch Complex 37 at Cape Canaveral. This paper's objective is to investigate asymmetry in noise radiation resulting from the unmerged plumes of the rocket's three distinct nozzles. Overall and frequency-dependent directivity patterns are calculated and discussed relative to jet and rocket noise literature. For example, previous studies of Atlas V and Space Launch System have shown frequency-dependent directivity in clustered nozzle configurations, with high-frequency asymmetries observed. This research contributes to understanding these dynamics, offering insights into how engine configurations impact acoustic behavior and potential mitigation strategies. Such insights could inform future rocket and launch pad design.

TUESDAY MORNING, 20 MAY 2025 — BALCONY J, 7:50 A.M. TO 11:00 A.M.

Session 2aPAa

Physical Acoustics, Education in Acoustics and Engineering Acoustics: Celebrating Steven L. Garrett's Fifty Years of Contributions in Acoustics I

David A. Brown, Cochair
ECE, Univ. Massachusetts Dartmouth, 151 Martine Street, Suite 123, Fall River, MA 027230000

Robert W. Smith, Cochair
PSU/ARL, P.O. Box 30, State College, PA 16804

Chair's Introduction—7:50

Invited Papers

8:00

2aPAa1. Steve Garrett at UCLA. Julian D. Maynard (Phys., Penn State Univ., 104 Davey Lab, Box 231, University Park, PA 16802, maynard@phys.psu.edu)

Steve Garrett was a graduate student in the Physics Department at UCLA when I was a postdoc there. Steve was a brilliant experimentalist, and well knew he was lucky to be learning from a leading acoustician, Isadore Rudnick. Because nearby Hollywood had a lasting influence, this talk will have three episodes. E1. The mystery of the Peruvian whistling bottles. E2. Steve speaks (in "The Unusual Properties of Liquid Helium"). E3. Two-for-one: How Steve's Ph.D. research launched two careers.

8:20

2aPAa2. The Mentor's nest. Robert M. Keolian (Sonic Joule LLC, 732 Holmes St., State College, PA 16803, keolian@psu.edu)

Isadore Rudnick provided a rich, warm, loving environment for his graduate students. It was a wonderful place to absorb his wisdom in acoustics, both linear and nonlinear, and fluids, both classical and quantum, along with that of the then-young professors Seth Putterman and Gary Williams. We students had ample lab space, all the helium we could use, and access to tools, electronics, and facilities accumulated over Izzy's long career. Izzy's demos were a treat that made their physics obvious. We enjoyed the peripheral glow of his ASA Gold Medal, his Fritz London Memorial Prize, and his inclusion into the National Academy of Sciences. He brought us to ASA meetings where we could watch him and Moe Greenspan hold court in the hallways. Izzy treated us like family by including us in family clam bakes and birthday events. His fame brought scientists visiting from around the world, and he would bring them around to spend much of their time in our student labs. Izzy showed us a way of life, apparent from about half of his 32 doctoral students ultimately becoming professors.

8:40

2aPAa3. Nonlinear resonant mode conversion and the Kramers–Kronig relations. Seth Putterman (Phys. Dept., UCLA, 475 Portola Plaza, Los Angeles, CA 90095, puherman@ritva.physics.ucla.edu)

Understanding Acoustics: An Experimentalist's View of Sound and Vibration by Steven L. Garrett is extraordinary for the spectrum of topics covered: both theoretical and experimental. As an example I will discuss Steve's work on resonant mode conversion in liquid helium; a nonlinear process whereby a propagating temperature wave can create a first sound or pressure wave. This research established the theory and demonstrated its measurement. Steve loved to quote Izzy Rudnick's overarching insight that the experiment should work because "Helium is more nonlinear than water." Resonant mode conversion also exists for a sound wave interacting with a spectrum of sound; a topic pioneered by Landau which surprisingly is not in his monograph on Fluid Mechanics. Another topic in Steve's book that is surprisingly missing from Landau's hydrodynamics is the Kramers–Kronig relations. Using the clear exposition in Steve's book as the jumping-off point I will discuss difficulties that appear when one tries to unify resonant mode conversion with the Kramers–Kronig relations.

9:00

2aPAa4. A Physicist and an Engineer walk into a bar. Neil A. Shaw (Menlo Sci. Acoust., Inc., P.O. Box 1610, Topanga, CA 90290, menlo@ieee.org)

Steve Garrett and I met in 1975 at UCLA when we were graduate students. He was the TA for Isadore Rudnick's Low Temperature Physics course, and I thought "that course would be cool." I was a Mechanical Engineering student (Richard Stern and William C. Meecham were my advisors). Despite this, we became good friends. Over the years we met at ASA and other meetings as well as when he visited Los Angeles to see family. He was always interested in, and I was working with, electroacoustics, as well as other areas of acoustics. Audio and books as well as sonic booms are areas of common interest for us. When he learned that I was the custodian of Vern O. Knudsen's library, he reached out and we arranged for the library to be sent to join Cyril Harris' library, already at Penn State. Steve also facilitated the donation of a portion of my library to Penn State.

9:20–9:40 Break

9:40

2aPAa5. Determining the Boltzmann constant and the temperature from sound speed measurements. Michael Moldover (NIST, 100 Bureau Dr., Gaithersburg, MD 20899, michael.moldover@nist.gov)

Extraordinary speed-of-sound measurements in helium and argon gas determined Boltzmann's constant k_B with sub-part-per-million uncertainties. These measurements were conducted in preparation for the 2019 redefinition of the unit of temperature T, the Kelvin. We outline the motivation for these measurements, the principles that enabled them, and their limitations. Similar sound speed measurements revealed errors of order $50 \times 10^{-6}T$ in the consensus International Temperature Scale of 1990 in the range 7 K $< T <$ 550 K. Accurate acoustic thermometry stimulated improved, first-principles calculations of the virial coefficients and transport properties of helium gas. Now, the calculated properties are so accurate that they are often used to calibrate measuring apparatus.

10:00

2aPAa6. Steve Garrett's energetic approach to acoustic energy research. Gregory Swift (Mater. Phys. and Applications, Los Alamos National Lab., M.S. K764, Los Alamos, NM 87545, swift@lanl.gov)

Beginning in graduate school under the guidance of Izzy Rudnick and Seth Putterman, and culminating with his solo textbook "Understanding Acoustics: An Experimentalist's View of Sound and Vibration" in 2020, Steve Garrett has enthusiastically spread a very effective approach to understanding and doing physical acoustics. Examples from his energetic research career (and his influence on my own) illustrate his style and the times in which we have lived.

10:20

2aPAa7. But it never goes to zero. Thomas B. Gabrielson (Penn State Univ., P.O. Box 30, State College, PA 16804, tbg3@psu.edu)

Steve and I first crossed paths in 1984 at the Naval Postgraduate School (NPS). He was building his career as a professor of physics; I was finishing my doctoral dissertation through Penn State while a visiting scientist at NPS. A fortunate intersection for me. Since then, I have been privileged to have learned from his facility and fascination with tools: tools in physics, tools in engineering, and tools in the machine shop. This is one story of many: eye opening for me, business as usual for Steve. After a lengthy conversation about demodulator noise in fiber-optic sensors, I asked about more fundamental issues. In 10 min, he calculated a lower limit to the self-noise based on equipartition, the fluctuation–dissipation theorem, and an approximation to the dynamics of the sense element. While this lower limit was well known in some communities—and, in fact, all the way back to the mirrored galvanometer—publication of the limit caused many developers of new-technology sensors to re-think designs and modify claims of extremely low self-noise. The self-motion of a sensor mechanism does not go to zero with time. Neither does the influence of the master of his craft on the apprentice.

10:40

2aPAa8. Optical fiber interferometry and Bragg gratings for measuring sound. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsdb@gmail.com)

Optical fiber interferometers are intrinsically differential path-length sensors that may be combined with transducers such as flexural disks and ellipsoids that produce push–pull response for wanted actions of sound and common mode rejection to unwanted actions such as vibration or thermal changes. Fiber Bragg gratings (FBG) can be "UV-written" through the sides of the optical fiber to realize internal

wavelength selective mirrors. The FBG gratings can then be combined with the transducers to realize path-length matched interferometric sensors enabling interrogation of lengths of fiber as well as time and frequency division multiplexing. Steven Garrett and collaborators at the Naval Postgraduate School (Hofler, Gardner, Brown, Keolian *et al.*) demonstrated many of these interferometric devices to detect underwater sound and worked on push–pull sensors, thermal noise characterization, acoustic motion sensors, materials characterization, demodulation, and multiplexing four decades ago. See, e.g., *J. Acoust. Soc. Am.* 83, S19 (1988) and 88, 591 (1990) as well as many thesis and dissertations https://discover.dtic.mil/.

TUESDAY MORNING, 20 MAY 2025 BALCONY I, 7:55 A.M. TO 10:40 A.M.

Session 2aPAb

Physical Acoustics and Biomedical Acoustics: Acoustic Radiation Force and Its Applications

Mohamed A. Ghanem, Cochair
University of Washington, 1013 NE Boat Street, Seattle, WA 98105

Oleg A. Sapozhnikov, Cochair
University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—7:55

Contributed Paper

8:00

2aPAb1. Estimation of liquid viscosity using radiation forces from a potential well. Mohamed A. Ghanem (Appl. Phys. Lab., Univ. of Washington, 1013 NE Boat St., Seattle, WA 98105, mghanem@uw.edu) and Oleg A. Sapozhnikov (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Radiation forces provide a remote, non-contact method to evaluate the elastic or mechanical properties of materials. For instance, noninvasive ultrasound techniques can assess the health of tissue by measuring its modulus of elasticity. In this study, we employ radiation forces from a multi-element array to estimate the viscosity of a medium. The array generates an effective potential well with stabilizing radiation forces that levitate spheres in water. An ultrasound pulse induces oscillation of the sphere about its equilibrium position. The sphere's motion is tracked using a laser diode and a photoresistor. By analyzing the exponential decay of the motion, we estimate the viscosity of the medium. Our measurements agree with known viscosity values. Moreover, we observe that the sphere's oscillations exhibit characteristics of nonlinear spring constant under directional drag forces. This approach provides a non-contact method for measuring liquid properties in benchtop settings and demonstrates potential *in vivo* applications. [This work was supported by NIH K25-DK132416.]

Invited Paper

8:20

2aPAb2. Local assessment of visco-elastic properties with acoustical tweezers. Antoine Penneron (Mech. Eng., Univ. of Bordeaux, 351 Cr de la Libération, Talence 33400, France, antoinepenneron@gmail.com), Thomas Brunet, and Diego Baresch (Mech. Eng., Univ. of Bordeaux, Bordeaux, France)

Assessing the mechanical properties of soft matter is a subject of major interest, specifically if it can be performed in a local and non-invasive way. Here, we explore the use of single-beam acoustical tweezers as a novel active rheology approach. Individual microbubbles ($\sim$50–100 μm in size) are generated in soft hydrogels (Carbopol) and pulled by the radiation force of a focused vortex beam. Due to the good acoustic contrast of a microbubble, forces in the micronewton range were applied at low acoustic intensities ($<10\,\mathrm{W/cm^2}$). The microbubble displacement is tracked using optical microscopy, and we are able to detect the small displacements, u, of a few microns induced in the soft hydrogel. Combined with radiation force calculations and a simple elastic model for the displacement, local values of the shear elastic modulus, G, are obtained. Our local measurements are in very good agreement with those obtained using standard bulk shear rheometry. We then further focus on the full visco-elastic behavior of the hydrogel by exploring the microbubble dynamics in response to a time-varying radiation force. Overall this new approach offers real advantages over traditional bulk rheology methods, as it can be considered minimally intrusive, local, and well adapted to probe inhomogeneous, anisotropic, and opaque-to-light materials.

8:40

2aPAb3. Radiation force on inhomogeneous subwavelength scatterers due to progressive waves. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 9767, Austin, TX 78766-9767, chiragokani@gmail.com), Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Gor'kov's result for radiation force on a subwavelength homogeneous sphere in the direction of an incident progressive plane wave with time-averaged intensity$\langle \mathbf{I} \rangle$[Sov. Phys. Dokl. 6, 773–775 (1962)] is generalized to arbitrarily shaped inhomogeneous scatterers. Westervelt's surface integral in the far field [JASA 29, 26–29 (1957)] reduces by energy conservation to $F_{\|} = \langle I/c_0 \rangle \oint |\Phi|^2 (1 - \mathbf{e}_i \bullet \mathbf{e}_r)\, d\Omega$, where Φ is the scattered wave directivity, $\mathbf{e}_i$ and $\mathbf{e}_r$ are the incident and radial unit vectors, respectively, and $d\Omega$ is the differential solid angle. Since the scatterer size a is much smaller than the wavelength $\lambda = 2\pi/k$, Φ can be calculated in terms of the acoustic polarizabilities $\alpha_{\mathrm{m}} = -\int f_1\, dV$, $\alpha_{\mathrm{d}} = \int 3f_2/(2+f_2)\, dV$, and $\alpha_{\mathrm{c}} = k\, [3\mathbf{e}_i \bullet \mathbf{e}_r \int \mathbf{r} f_2/(2+f_2)\, dV - \int \mathbf{r} f_1\, dV]$, where f_1 and f_2 are Gor'kov's contrast factors. For scatterers whose material properties are symmetric about the centroid $\mathbf{r} \equiv \mathbf{0}$, α_{c} vanishes, recovering Gor'kov's $O[(ka)^4]$ force, while material asymmetry contributes at $O[(ka)^6]$. The forces are verified by comparison to solutions based on partial wave expansions and Fourier transforms. [C.A.G. was supported by the ARL:UT McKinney Fellowship in Acoustics.]

9:00

2aPAb4. Effect of homogenization on calculation of acoustic radiation force. Cassidy A. Christie (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 9767, Austin, TX 78766-9767, cchristie@utexas.edu), Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Analytical methods for calculating acoustic radiation force are complicated by the presence of heterogeneity. For a multilayered sphere formed by biological substances having densities and compressibilities similar to those of the host fluid, the simplification provided by the Born approximation of the radiation force exerted by a 1-D standing wave was found by Jerome and Hamilton [JASA 150, 3417 (2021)] to be accurate via comparisons with the full solution by Wang *et al*. [J. Appl. Phys. 122, 094902 (2017)]. An alternative approximation based on replacing the layered sphere with a homogeneous effective medium, achieved by taking volume averages of the densities and compressibilities, was explored briefly by Gokani *et al.* [POMA 48, 045002 (2022)] in connection with particle sorting (acoustophoresis) in a microfluidic device that employs noncollinear plane waves. The present work explores the parameter range for which the simplification due to homogenization provides a reasonable approximation of the radiation force acting on layered spheres. Both 1-D standing waves and plane progressive waves are considered for the incident fields. Different densities and compressibilities of the layers are considered, as well as the ordering and thicknesses of the layers within the sphere. [CAC was supported by the ARL:UT McKinney Fellowship in Acoustics]

9:20–9:40 Break

9:40

2aPAb5. Generation of negative radiation forces for pulling spheres. Mohamed A. Ghanem (Appl. Phys. Lab., Univ. of Washington, 1013 NE Boat St., Seattle, WA 98105, mghanem@uw.edu)

Negative radiation forces have been reported on objects smaller than the wavelength λ; however, for larger objects, the negative forces have been feasible under limited scenarios. Here, we use a multi-element array to generate various acoustic beams that employ non-gradient forces to impart negative forces on solid targets larger than λ. The array operates at 1.5 MHz and creates a null region that surrounds target spheres with zero on-axis pressure. The generated wavefield produces forward scattering while minimizing backscattering, thus enabling the generation of negative forces. Simulations and measurements show a distinct transition between gradient and non-gradient forces. The negative forces generated are distal to the transducer. Calculations and measurements of the field show agreement for various acoustic traps for spheres up to 2λ. Although the acoustic exposure levels exceed current regulatory limits for *in vivo* applications, this technology has the potential utility in the development of acoustic tweezers for non-contact transport applications. [This work was supported by NIH K25-DK132416.]

10:00

2aPAb6. Nearfield forces generated using tunable liquid lenses. Sina Rostami (Phys., Univ. of MS, 108 Lewis Hall, P.O. Box 1848, University, MS 38677, srostami@go.olemiss.edu) and Joel Mobley (Phys. and Astronomy, Univ. of MS, University, MS)

In previous work, we introduced 4- and 6-channel liquid-tunable acoustic lenses, demonstrating their ability to generate ultrasonic beams exhibiting limited diffraction with adjustable focal zones and depths of field. The target phasing was based on that of the fraxicon, a quasi-planar stepped lens, and the beams formed by the liquid lenses had comparable structures. This current study investigates the forces generated in the near field of beams formed using these tunable lenses. The forces are calculated using the Gor'kov potential applied to measured ultrasonic pressure fields. We compare results with those produced by fraxicon, which has demonstrated the capacity to facilitate the extraction, trapping and manipulation of droplets. A variety of phase profiles for the liquid lenses are used to investigate the presence of field structures similar to the trapping zones generated with the fraxicon. The tunable liquid lens has the potential to be adapted to a variety of targets and systems, providing flexibility not afforded by fixed-phase quasi-planar lenses.

10:20

2aPAb7. Numerical study of two-dimensional resonant gas oscillation with shock waves. Takeru Yano (Mech. Eng., Osaka Univ., Yamadaoka 2-1, Suita 5650871, Japan, takeru.yano@gmail.com)

We study the nonlinear resonant gas oscillation and the propagation of shock waves in a rectangular region bounded by solid walls by numerically solving the system of Navier–Stokes equations for compressible flows of an ideal gas. The numerical method is the fourth-order central difference in space and the fourth-order backward difference in time. The resonant gas oscillation is supposed to be excited by the harmonic oscillation of solid walls with a resonant frequency. Our main objective is to study the peculiarities of long-time phenomena caused by the accumulation of small nonlinear effects due to shock waves. In particular, we show that the flow pattern of acoustic streaming is affected by shock–shock and shock–boundary layer interactions.

Session 2aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Best Student Poster Award Session (Poster Session)

Daniel R. Guest, Cochair
Department of Biomedical Engineering, University of Rochester, 601 Elmwood Avenue, MC 5-6483, Rochester, NY 14620

Monica L. Folkerts, Cochair
Communication Sciences and Disorders, University of Central Florida, 3280 Progress Drive, Suite 100, Orlando, FL 32826

All posters will be on display from 8:00 a.m. to 11:00 a.m. Authors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:30 a.m. and authors of even papers will be at their posters from 9:30 a.m. to 11:00 a.m.

Contributed Papers

2aPP1. Harmonicity and voice segregation in polyphonic music: Effects of age and hearing loss. Lisanne G. Bogaard (Psych., Univ. of Minnesota, 75 E River Pkwy, Minneapolis, MN 55455, bogaa002@umn.edu), Juraj Mesik, and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Harmonic relations between components help to perceptually bind them and create the percept of pitch, usually relating to the common fundamental frequency. Speech segregation is somewhat but not drastically impaired when competing talkers are rendered inharmonic. Less is known about the role of harmonicity in polyphonic music. Both age and hearing loss may affect pitch perception, which in turn may influence the role of harmonicity in music and source segregation. This study investigated the role of harmonicity in listeners' ability to count the number of voices in polyphonic music and to follow one voice in the presence of others. Performance was compared across groups of younger normal-hearing (yNH), hearing-impaired (HI), and age-matched normal-hearing (aNH) listeners. Performance declined in all groups as voice count and inharmonicity increased. The yNH group consistently outperformed aNH and HI listeners, with no significant differences found between the aNH and HI groups. Surprisingly, mean performance in all groups remained above chance in inharmonic conditions, suggesting that harmonicity is not critical for source segregation, even in music. [Work supported by NIH grant R01DC005216.]

2aPP2. Variations in audiovisual benefit for speech perception in noise versus multi-talker maskers. Jaeeun Lee (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, jaeeun.lee4531@gmail.com) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Speech perception is a complex multimodal process that often involves auditory and visual information. Studies of audio-visual asynchrony in speech reveal an asymmetry, with listeners performing better when visual cues precede than follow auditory input. The current study investigates how audio-visual information and asynchrony affect speech sentence intelligibility across different acoustic backgrounds. We hypothesized that visual information and synchrony play a more important role under informational masking (e.g., speech-in-speech conditions), than under energetic masking (e.g., speech-in-noise conditions) because visual cues may help in perceptually segregating the target from the masker speech. Experiment 1 tested this hypothesis by comparing audio-only and audio-visual speech intelligibility in backgrounds of noise or competing talkers over a range of target-to-masker ratios. Our findings confirmed a greater audio-visual benefit for speech-in-speech than speech-in-noise conditions. Experiment 2 tested the effects of audio-visual temporal asynchrony. Preliminary results show a surprisingly shallow function in both conditions, relative to previous work using single words or vowels, when performance is plotted as a function of asynchrony. If confirmed, these results suggest that sentence-level stimuli may involve additional (cognitive) processes that mitigate the effects of temporal asynchrony on audio-visual speech intelligibility. [Work supported by NIH grant R01 DC016119.]

2aPP3. Contributions of auditory encoding and cognitive processing to degraded speech perception during adolescence. Jordin T. Benedict (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., Kent, OH 44240, jbened11@kent.edu), Catherine E. Slowey, Adelia R. Young (Speech Pathol. and Audiol., Kent State Univ., Kent, OH), Bruna S. Mussoi (Audiol. and Speech Pathol., Univ. of Tennessee Health Sci. Ctr., Knoxville, TN), and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., Kent, OH)

Previous studies demonstrated prolonged development of degraded speech perception, perhaps due to the maturation of auditory and cognitive brain regions. We investigated the contributions of auditory cortical encoding, measured by the acoustic change complex (ACC), and cognition to speech perception in adolescents (10–17) and young adults (18–23). Speech conditions included the QuickSIN and perception of 6-band noise-vocoded sentences, followed either by silence or multitalker babble. The ACC was measured to a frequency change condition (1000–950 Hz) and gap detection condition (10 ms gap in noise). Cognitive testing was also administered. Separate hierarchical stepwise regressions were completed for each condition. Preliminary results suggest that testing order (silence or babble first) exclusively affected performance on the NV silence condition. Encoding of timing information (ACC for gaps) contributed to the NV silence condition only, whereas encoding of spectral information (ACC for frequency) contributed to both NV speech conditions. Strategy use and sustained attention contributed to both NV speech conditions but age did not explain any unique variance after all factors were included. For the QuickSIN, auditory working memory and vocabulary contributed to performance, and age continued to influence performance after accounting for all other factors. [Funded by NIDCD.]

2aPP4. Changes in psychometric functions on two out of three auditory tasks during adolescence. Serena A. Sereki (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., Kent, OH 44240, ssereki@kent.edu), Samantha Allan (Speech Pathol. and Audiol., Kent State Univ., Kent, OH), Merri J. Rosen (Anatomy and Neurobiology, Northeast Ohio Medical Univ., Rootstown, OH), and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., Kent, OH)

Adolescents continue to develop on some psychoacoustic tasks, but most studies use adaptive tracking procedures to estimate thresholds; therefore, the effect of prolonged maturation on psychometric functions is relatively unknown. Here, we tested adolescents (age 10–18) and young adults (age 19–23) on three conditions: temporal interval discrimination (100 ms standard), gap detection in bandpass noise, and frequency discrimination (1000 Hz standard). Performance was measured using the method of constant stimuli and a 3AFC task. Data were analyzed by measuring threshold (66.7% correct; proxy for sensitivity), the slope of the psychometric function (a proxy for internal noise), and the proportion of listeners who met criteria for outliers (more than 2 SD from the mean), had poorly fit psychometric functions, or were unable to complete the task. Preliminary results suggest that temporal interval discrimination thresholds, slopes, and proportion of excluded data did not change with age. Gap detection thresholds improved and slopes became steeper with development, and the proportion of excluded data decreased with increasing age. Frequency discrimination thresholds did not improve with age, but slopes became steeper and excluded data decreased with maturation. These data suggest that internal noise may be decreasing during adolescence, even when thresholds stay the same. [Funded by NIDCD.]

2aPP5. Spiking neural networks for sound localization: A new perspective on illuminating auditory spatial perception. Qin Liu (Elec. Eng., EPFL, ELB 039 Station 11, Lausanne, Vaud 1015, Switzerland, qin.liu@epfl.ch), Laurent S. Simon (Sonova AG, Staefa, ZH, Switzerland), and Hervé Lissek (Elec. Eng., EPFL, Lausanne, Switzerland)

Humans estimate sound source direction using information from their auditory neural system. Traditional methods use auditory cues [e.g., interaural time differences (ITDs) and interaural level differences (ILDs), etc] to perform sound localization. These cues are extracted from binaural signals or decoded from neuronal firing rates. In contrast, we proposed a new computational model that directly localizes sound sources using the firing rates of auditory neurons, eliminating the need for physical cue extraction and the template-matching process. This model incorporates spiking neural networks (SNNs) and artificial neural networks (ANNs) to emulate auditory spatial perception. To get firing rates, the SNN uses auditory peripheral processing and physiological models of the cochlear nucleus and medial superior olive (MSO). The SNN calculates a database of firing rates from sine tones across varying positions and frequencies to train the ANN. The ANN performs nonlinear regression to predict azimuth and elevation angles, accommodating both narrowband and broadband signals. The integration of dynamic cues resolves front–back confusion by aligning with human auditory perception. We conducted a localization listening test with 10 participants with normal hearing, enabling the refinement of network parameters to closely mimic human behavior. In the future, hearing loss can be simulated by adjusting parameters related to inner hair cell dysfunction, thereby providing a robust framework for real-world spatial hearing.

2aPP6. Binge-watching music videos for joy? Unveiling the therapeutic effects and rules behind watching. Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong, Hong Kong, mhlawaa@connect.ust.hk) and Andrew B. Horner (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Young adults are often captivated by online videos, much like bees to honey. It raises the question of how to strike a balance between promoting healthy viewing habits while respecting individual freedoms. We also wondered whether binge-watching videos truly brings viewers joy. And what is the optimal duration to watch in order to boost happiness? In our study, we used four categories of music videos (MVs) sourced from YouTube: (1) All-Time Fastest Music Videos to One Billion Views, (2) All-Time Most Viewed MVs worldwide, (3) All-Time Top 24 Hour Music Debuts (Kpop), and (4) Recent Top Ranking MVs in Hong Kong. Through an online survey, participants watched the top 5 MVs in each category, with only their preferred category being displayed. The results of the ANOVA test suggest that young adults should view a maximum of four MVs to sustain changes in mood and energy levels. Exceeding this threshold may cause the positive effects to diminish. Moreover, our findings suggest that by selecting MVs that start with a positive mood, and intersperse mood variations throughout the viewing experience, individuals can effectively elevate their mood and energy levels.

2aPP7. Using cognitive measures to understand individual differences in spatial release from masking beyond those explained by perceived spatial separation. Brittany T. Williams (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, brittany.williams@boystown.org), Angela M. AuBuchon (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Spatial release from speech masking (SRM) refers to improved speech recognition when speech maskers are perceived to be separated from target locations. Binaural manipulations, such as conflicting interaural time and level differences, have been shown to reduce SRM and perceived lateral position. We assessed the impact of perceived target-masker separation by measuring spatial judgments of binaurally manipulated speech and correlating those judgments to individual listeners' SRM scores. The results account for 2.5–4 dB of SRM across the measured lateralization range. Individual SRM scores, however, spanned a range of more than 10–12 dB, suggesting that other factors primarily account for variation in speech unmasking. Past studies suggest that differences in working memory and attention could be responsible. Here, we relate variation in SRM and spatial perception to variation in working memory as assessed by an auditory running digit span task and selective attention as assessed by the Stroop Squared, Simon Squared, and Flanker Squared tasks. Results suggest the importance of considering non-acoustic aspects of listening ability, which may limit access to benefits of spatial hearing. [Work supported by NIH R01-DC016643.]

2aPP8. Underpinnings of acoustic feature integration in auditory salience: Insights from behavioral and pupillary responses. Nahaleh Fatemi (Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, sfatemi2@jhu.edu), Hsin-I Liao (NTT Commun. Sci. Labs., Atsugi, Japan), and Mounya Elhilali (Johns Hopkins Univ., Baltimore, MD)

The perception of salient events is shaped by processing dynamic acoustic scenes and integration of fluctuations across acoustic dimensions, signaling the presence of notable deviations. Auditory salience is affected by variations along the pitch, timbre, and sound intensity and arises from complex interactions among these features, which are likely influenced by interwoven representations of time, frequency, and intensity in the auditory cortex. This study compares cross-feature interactions underlying behavioral responses and pupil dilation responses (PDR) to salient auditory events. Participants listened to musical and nature scenes with salient events marked by changes in intensity, pitch, or timbre. A Bayesian tracking framework using the Dynamic Regularity Extraction (D-Rex) model was used to generate predicted salience responses to the different scenes. The model calculated surprisal values for each acoustic feature and used a constrained regression to examine cross-feature interactions that best predicted behavioral and pupillary responses. Behavioral responses showed stronger feature interactions compared to PDR, suggesting that PDR reflected stimulus-driven changes, indicative of lower-level auditory processes, while behavioral responses captured more complex interactions between features, representing higher-level cognitive processing. This finding highlights the hierarchical nature of auditory perception, with PDR reflecting automatic responses to changes and behavior shaped by interactions between acoustic dimensions.

2aPP9. Auditory spatialization approaches differentially recruit sensory-biased prefrontal cortex. Wusheng Liang (Carnegie Mellon Univ., Baker Hall A55A, Pittsburgh, PA 15217, liangws9789@gmail.com), Abigail Noyce (Carnegie Mellon Univ., Pittsburgh, PA), Christopher A. Brown (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA), and Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA)

Past spatial auditory studies have used various approaches to manipulate perceived source location, including pure interaural time differences (ITDs), pure interaural level differences (ILDs), and more natural and realistic head-related transfer functions (HRTFs). Previously, we showed that the effectiveness of spatial attention is strongest for HRTF simulations and weakest for pure ITDs, but the mechanisms explaining such differences remain unclear. The prefrontal cortex (PFC) supports many higher-order cognitive functions, including working memory. Visual-biased PFC regions show greater activation during auditory tasks that require spatial processing than those that do not, suggesting that visual-biased PFC regions play an important role in auditory spatial cognition. To investigate the interaction between spatial cues and spatial processing, we conducted an fMRI study testing different auditory tasks (spatial, non-spatial, and passive) and different spatialization approaches (ITD, ILD, or HRTF). In spatial tasks, HRTFs yielded the best behavioral performance and strongest activation across PFC; ITDs yielded the lowest performance and weakest activation. Visual-biased PFC regions showed greater activation during spatial than nonspatial tasks. These results provide new insights into how spatial cues interact with PFC regions during auditory tasks.

2aPP10. Evaluating the usability of gamified spatial release from masking tasks using the Portable Automated Rapid Testing app. Nicole Dean (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239, nikki.lavee.dean@gmail.com), Frederick J. Gallun, Tess Koerner, Lauren Charney, Karen Garcia, Conner Corbett (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Aaron Seitz (Northeastern Univ., Riverside, CA), and Chris Stecker (Boystown National Res. Hospital, Omaha, NE)

This research presents initial findings from an ongoing study assessing the usability of gamified Spatial Release from Masking (SRM) tasks using the Portable Automated Rapid Testing (PART) app on an iPad. Evaluating usability data is crucial for optimizing task design, ensuring accessibility, and improving user engagement across diverse populations. The SRM tasks used target and masker sentences from the Coordinate Response Measure (CRM) speech database (Bolia *et al*., 2000). Participants completed two gamified SRM tasks and one non-gamified version in randomized order. After each task, participants rated usability through five survey questions on a 5-point Likert scale (e.g., "Please rate your willingness to perform this task again: Strongly Disagree, Disagree, Neutral, Agree, Strongly Agree"). Performance on each SRM task was also compared to usability feedback collected via a 6-question survey administered after all tasks (e.g., "Which of the three tasks did you find most challenging?"). Demographic factors, including age, hearing level, and history of Traumatic Brain Injury (TBI), were analyzed in relation to SRM performance. Overall, usability scores were high, suggesting that gamification is a viable direction for psychoacoustical tests of SRM. Interactions with demographic variables will be discussed.

2aPP11. Interference and independence in the perception of pitch and brightness. Yongtian Ou (Psych., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, ou000036@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Perception of pitch (defined by fundamental frequency, F0) and brightness (mainly affected by spectral centroid, Fc) has been suggested to be interdependent. However, the encoding mechanism of such interdependence is unclear. One hypothesis suggests that, while F0 and Fc are encoded separately, the direction of change in one feature tends to be confused with that of the other. Another hypothesis argues that F0 and Fc are encoded in a single, or at least overlapping, perceptual dimension. We conducted two experiments to disentangle these two hypotheses. First, participants were asked to make an up-down judgment in pitch or brightness as F0 and Fc varied congruently or incongruently. We found that congruent changes in F0 and Fc led to better discrimination, whereas incongruent changes led to worse discrimination, consistent with both hypotheses. Second, participants were asked to judge if there was a change, regardless of feature or direction, when only one feature varied, both varied congruently, or both varied incongruently. We found that variations in both features simultaneously were easier to detect than variations in only one feature, but there was no significant difference between the congruent and incongruent variations, arguing against the second hypothesis. [Work supported by NIH grant R01 DC005216.]

2aPP12. Enhancing spatial release from masking through individual head-related transfer function augmentation. Nils Marggraf-Turley (Audio Experience Design, Imperial College London, 35 Heythorp St., London SW18 5BW, United Kingdom, nm723@ic.ac.uk), Niels Pontoppidan, Martha M. Shiell (Eriksholm Res. Ctr., Oticon A/S, Snekkersten, Denmark), Drew Cappotto, and Lorenzo Picinali (Audio Experience Design, Imperial College London, London, United Kingdom)

Binaural hearing underpins effective communication in complex acoustic environments by increasing listeners' abilities to segregate concurrent sound sources. In certain conditions, augmenting binaural cues has been shown to improve speech-in-noise performance, yet existing methods largely rely on processing short temporal windows, often causing unwanted artifacts. Moreover, the perceptual implications of augmenting an individual Head-Related Transfer Function (HRTF) are not well understood. We revisit a previous augmentation approach applied directly to individual HRTFs and introduce a novel augmentation technique designed to overcome earlier limitations. Both strategies are evaluated using auditory models and behavioral tests employing a speech-on-speech spatial release from the masking paradigm, with a target at the listener's midline and symmetrical maskers at varied azimuths. Initial findings indicate that augmentations enhance spatial release from masking relative to non-augmented individual HRTFs, though auditory models tend to overestimate this improvement. Conventional metrics do not seem to be able to account for differences arising from these augmentations, suggesting that listener familiarity with individual cues influences observed intelligibility gains. Our proposed method also mitigates the spatial spreading associated with previous approaches.

2aPP13. Effects of conflicting spatial and pitch cues on timing judgments in auditory sequences. Tess M. Starr (Univ. of Michigan, 1137 Catherine St., Ann Arbor, MI 48109, testarr@med.umich.edu), Carolyn Kroger, Renee Banakis Hartl (Univ. of Michigan, Ann Arbor, MI), Kelly L. Whiteford (Univ. of Michigan, Minneapolis, MN), and Anahita H. Mehta (Univ. of Michigan, Ann Arbor, MI)

Feature integration is necessary for parsing complex auditory environments. To perceive or identify auditory objects or sound sources, listeners integrate acoustic features such as pitch, spatial location, and onset timing as auditory events unfold. One example of how stimulus features bias auditory perception is the auditory kappa effect, where changes in pitch bias the perceived onset timing among sequential sounds. Larger changes in pitch between tones are perceived as longer time intervals. Our recent work has extended this effect from pitch to physical space, demonstrating that sounds traversing greater spatial distances were perceived as farther apart in time than sounds presented closer together. Together, these studies raise the question of whether pitch and spatial cues interact with bias timing judgments. In the present study, participants heard sequences of three harmonic complex tones that changed in pitch and location and reported the relative time intervals between tones. Pitch changes were presented either congruently with spatial location (e.g., larger pitch differences paired with farther distance in space) or incongruently (e.g., smaller pitch changes paired with farther distance in space). Results will be discussed in terms of competing influences of pitch and spatial cues on timing judgments. [Funding: HHF ERG (AHM), NIH F32DC022162 (CK)]

2aPP14. Auditory spatio-temporal interactions in single-sided deaf patients with unilateral cochlear implants. Carolyn Kroger (Otolaryngol. – Head and Neck Surgery, Univ. of Michigan, 1301 Catherine St., Ann Arbor, MI 48109, carrie.kroger@gmail.com), Deborah Fu, Renee Banakis Hartl (Otolaryngol. – Head and Neck Surgery, Univ. of Michigan, Ann Arbor, MI), Ruth Y. Litovsky (Commun. Sci. & Disord., Univ. of Wisconsin Madison, Madison, WI), and Anahita H. Mehta (Otolaryngol. – Head and Neck Surgery, Univ. of Michigan, Ann Arbor, MI)

There is a growing use of unilateral cochlear implants (CIs) as a promising intervention for patients with single-sided deafness (SSD), especially to improve speech perception in noisy environments and sound localization performance. However, the effectiveness of CIs in improving spatial hearing for this population remains uncertain. Traditional sound localization tasks are often lengthy and cognitively taxing for individuals who struggle to identify sound source locations. This study introduces a new task for evaluating the restoration of auditory spatial cues in SSD-CI users by examining the biasing effects of spatial location on timing judgments for sequential sounds. The auditory spatial kappa (ASK) effect manifests as a bias in perceived relative onset timing between subsequent sounds as a function of the spatial distance between them. Sounds that are farther in space are judged as farther apart in time compared to sounds presented closer in space. We tested the degree of spatial bias on temporal judgments in SSD patients with and without their CI to evaluate the restoration of spatial cues with CI in different frequency ranges. Results indicate that the ASK task may be a useful tool for implicitly assessing spatial hearing in clinical populations who perform poorly on traditional sound localization tasks. [Funding: NIH F32DC022162 (CK)]

2aPP15. Evidence of compensative listening for speech segregation in hearing-impaired adults. Lindsey Kummerer (Univ. of South Florida, Tampa, 4202 East Fowler Ave., PCD1017, Commun. Sci. and Disord., Tampa, FL 33620, lkummerer@usf.edu), Gabriella Brown, Robert A. Lutfi, and Jungmee Lee (Univ. of South Florida, Tampa, Tampa, FL)

Hearing-impaired (HI) individuals vary widely in their ability to segregate the speech of a conversation partner from others speaking at the same time. The role of hearing loss can be difficult to gauge because reduced sensitivity to relevant speech cues may be compensated for by changes in the *relative reliance* listeners place on those cues. Here, we provide an example. Thirteen NH and 15 HI adults (PTA thresholds >25 dBHL from 0.25–8.0 kHz) heard natural recordings over headphones of two talkers speaking sentences concurrently. Talker B (distracter) was present on each trial, the other talker (target) was equally likely to be A or C differing from B and each other in fundamental frequency (F0) and azimuthal location (θ). The listeners' task was to identify the target as A or C. Relative cue reliance, $R = c_{F0}/(c_{F0} + c_{\theta})$, was determined from regression coefficients, c, relating the listener's response to trial-by-trial perturbations simulating natural variation in F0 and θ. Relative cue sensitivity, $S = d'_{F0}/(d'_{F0} + d'_{\theta})$, was determined from listener performance, d′, for F0 and θ cues presented individually. The results show a significantly greater range of S and a positive correlation of S with R for HI compared to NH listeners. [Work supported by NIH R01 DC001262-31.]

2aPP16. The role of self-construal in auditory spatial attention: Neural responses to masking noise in speech-in-noise tasks. Akira Takeuchi (Golisano College of Computing and Information Sci., Rochester Inst. of Technol., Lomb Memorial Dr., Rochester, NY 14623, akira-musico@outlook.jp), Hwan Shim (Dept. of Elec. and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY), Inyong Choi (Dept. of Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), and Sungyoung Kim (Dept. of Elec. and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY)

This study investigates listeners' biological responses to speech under different masking noise conditions during a speech-in-noise task, examining the role of self-construal—a psychological measure of individual cognition, emotion, and motivation. Fifty participants (19 from the US and 31 from Japan) with normal hearing completed a speech identification task with masking noise streams. The target speech was presented through a front-left speaker while masking noise consisting of music and unintelligible speech was played from either a front-right or back-center speaker. Neural responses were recorded using a 21-channel EEG system, and Event-Related Potentials (ERPs) were analyzed for conditions defined by two Signal-to-Noise Ratios (SNRs: −18 and −12 dB) and masker positions. Participants were categorized into interdependent or independent self-construal groups via a survey. ANOVA results showed that interdependent listeners exhibited significantly lower ERP amplitude for front-positioned maskers at lower SNR ($p = 0.0005$). No significant interaction was found between language and self-construal groups ($p = 0.3882$). These indicate that the masker position impacts interdependent listeners, regardless of their languages. These findings highlight the influence of self-construal on spatial auditory responses. Future research will employ realistic, continuous speech stimuli to further refine our understanding of individual auditory cognition in complex spatial environments.

2aPP27. Evaluating dry-electrode electroencephalography system for auditory neuroscience using speech-in-noise tasks. Akira Takeuchi (Golisano College of Computing and Information Sci., Rochester Inst. of Technol., Lomb Memorial Dr., Rochester, NY 14623, akira-musico@outlook.jp), Hwan Shim (Dept. of Elec. and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY), Inyong Choi (Dept. of Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), and Sungyoung Kim (Dept. of Elec. and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY)

This study evaluates the feasibility of using a dry-electrode electroencephalography (EEG) system for auditory cognitive neuroscience research by comparing its performance with a conventional wet-electrode system during auditory tasks. EEG data were collected using the Biosemi Active Two (wet-electrode system) and the Wearable Sensing DSI-24 (dry-electrode system) as participants completed a Speech-in-Noise task based on the Coordinate Response Measure (CRM) corpus. Participants identified the color and number of target sounds while masker sounds varied in direction and signal-to-noise ratio (SNR). Results reveal no significant differences in SNR between the dry- and wet-electrode systems. This indicates that the dry-electrode system can reliably capture auditory neural responses, making it a suitable tool for auditory neuroscience research. The use of auditory stimuli in this study highlights the applicability of dry-electrode systems for investigating brain responses to complex auditory environments, such as speech-in-noise. These findings suggest that dry-electrode EEG systems offer a practical and efficient alternative for auditory cognitive neuroscience, enabling easier setup and improved participant comfort without compromising data quality. This advancement holds promise for applications in auditory attention decoding, brain-computer interfaces, and clinical research.

2aPP17. Context-dependent auditory inference in environmental sound recognition. Keland Moore (Psychol. and Brain Sci., Univ. of Iowa, 340 Iowa Ave., Iowa City, IA 52245, keland-moore@uiowa.edu) and James Traer (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

In everyday listening, humans make multiple perceptual inferences from sounds (e.g., what object was that? Was it heavy? Hard? Metallic?). Such judgments are robust across different listening contexts (e.g., different motions, etc.). Using listening tasks and Ideal Observer Models trained on a large set of acoustic cues, we investigated which cues might underlie human auditory physical inference. Specifically, we asked listeners to identify attributes of objects (e.g., shape, weight, material, etc.) from clattering sounds. Prior studies of speech perception have highlighted the importance of context-dependent inference (i.e., the use of different acoustic cues in different contexts, such as different speakers, accents, etc.), and we investigated its role in such tasks. Our stimuli varied across multiple physical attributes and thus, for any query (e.g., weight) the other attributes (shape, material) served as unknown and variable contexts. Additionally, we presented stimuli in different "acoustic contexts," manipulating spectro-temporal structure and bounce patterns to further elucidate the role of various acoustic cues in shaping human judgments. Models show different acoustic cues best predict human judgments in different conditions, suggestive of context-dependent inference. The use of such mechanisms in non-speech categorization suggests they are not specific to speech perception but may instead be fundamental to hearing.

2aPP18. Investigating spatial and semantic proximity in auditory processing with event-related potentials. Robin Duclermortier (LTDS, ENTPE, 3 rue Maurice Audin, Vaulx-en-Velin 69120, France, robin.duclermortier@entpe.fr), Mathieu Lavandier (LTDS, ENTPE, Lyon, France), and Fabien Perrin (CAP, CRNL, Lyon, France)

We previously demonstrated an interplay between semantic and spatial proximity in auditory processing, with self-relevant stimuli (e.g., one's own name) enhancing distance discrimination in extrapersonal space. Building on this result, we designed an event-related potentials (ERPs) experiment to understand the cerebral bases of this interaction, by varying proximity in distance and/or autobiographical dimensions. Participants listened to equiprobable stimuli consisting of their own name and an unfamiliar name, each convolved with binaural room impulse responses to simulate two distances (20 cm and 8 m, on both sides). In four blocks (randomly distributed), they were asked to count the stimulus that was semantically and spatially near (own name at 20 cm), spatially near only (unfamiliar name at 20 cm), semantically near only (own name at 8 m), or neither near (unfamiliar name at 8 m). We observed modulations in early sensory-evoked and late cognitive ERPs, reflecting acoustical variations (intensity/reverberation or phonetic characteristics) and proximity discrimination (distance from self in spatial or semantic attributes). Early and late interactions in target and non-target stimuli reflected bottom–up and top–down processes, suggesting automatic and selective attention. This study suggests that auditory perception integrates embodied and situated mental representations, enabling early detection of object proximity, i.e., potential relevance/threat.

2aPP19. Decoding ambiguity: Behavioral insights into contextual priors in auditory perception. Rohit Kumar (Johns Hopkins Univ., 3538 Beech Ave., Baltimore, MD 21211, rkumar44@jhu.edu) and Mounya Elhilali (Johns Hopkins Univ., Baltimore, MD)

Contextual priors play a critical role in auditory perception, shaping how we interpret and understand sounds based on expectations informed by previous contexts or prior experiences. While much remains unknown about how past cues influence the processing of current sensory information, ambiguous sounds present a valuable tool to experimentally investigate how stored priors bias perception. Ambiguous sounds are defined as sounds that can be identified as different events and can be readily "pushed" in different perceptual directions depending on the context or cue provided. In two online experiments, human participants were presented with ambiguous sounds in a priming paradigm where each sound was paired with a visual cue that either preceded or succeeded the auditory stimulus. Ambiguous sounds that have multiple interpretations as well as non-ambiguous sounds were paired with either matched or mismatched visual cues. Results revealed that ambiguous sounds were often misidentified when paired with their counterparts' assigned labels, whereas non-ambiguous sounds showed consistent identification across all conditions. This study emphasizes the vital role of contextual cues in perceptual decision-making, especially in ambiguity resolution, and lays a foundation for investigating the neural dynamics of auditory perception.

2aPP20. *A priori* knowledge about the signal and binaural processing capabilities. Morgan Barkhouse (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson University, Towson, MD 21252, mbarkh1@students.towson.edu), Quinn Donarum, Rebecka Livingstone, Brianna Saffran, Jaclyn Carney, Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Various factors play a significant role in determining a listener's effectiveness in understanding the speech of the target talker in complex auditory contexts including similarity and spatial relationship between speech sources, the listener's age, and hearing acuity, among others. Previous research indicates that directing attention to a specific location in space—especially when the listener is aware of the speaker and/or has prior knowledge of the target stimulus—can greatly improve speech understanding in the presence of competing talkers. *A priori* knowledge regarding the spatial location of the target yielded significantly improved speech understanding under conditions of high acoustic ambiguity. However, there is no information about how *a priori* information about the quality of the signal affects speech identification. Here, we present data on reported listening effort and speech understanding while manipulating the spatial location of the target and the maskers and the *a priori* information available to the listeners. Preliminary analyses indicated no effect on speech understanding based on the available *a priori* information for both colocated and separated conditions. However, perceived listening effort varied as a function of spatial location and available *a priori* information. Also, the effects of aging and hearing impairment on these measures will be discussed.

2aPP21. Working memory and spatial processing capabilities. Sadie O'Neill (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, soneil7@students.towson.edu), Morgan Barkhouse, Chhayakanta Patro (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD), and Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., Towson, MD)

Speech understanding in complex listening environments draws heavily on working memory capabilities and is directly impacted by age-related cognitive deficits. In a normally functioning auditory system, when the target and maskers are spatially separated, binaural benefits can be significant and the subsequent improvement in the intelligibility of the target signal can be measured as a spatial release from masking. Here, we investigate the relationship between working memory and implied cognitive load on speech understanding. Working memory capacity was measured using an abbreviated reading span task. Implied cognitive load was varied using a new temporal overlap threshold estimation method (O'Neill *et al.*, 2023) that used a divided attention version of the classic spatial release from the masking task. Initial analyses of the data revealed that the identification of the first target call sign was higher compared to the identification of the third call sign even though the third call sign was presented the last and was unmasked by other speech stimuli. This trend was true for both colocated and separated listening conditions. Additionally, the interplay between age, hearing loss, and reading span measures in relation to speech identification will be discussed.

2aPP22. Foreground to background: Sound identity as a determinant of auditory salience and importance. Yu-Jeh Liu (Johns Hopkins Univ., 3400 North Charles St., Baltimore, MD 21218, yliu436@jhu.edu) and Mounya Elhilali (Johns Hopkins Univ., Baltimore, MD)

Sound identity defines a hierarchy of auditory perception, influencing how sounds in dynamic scenes are prioritized. The current study explores a perceptual continuum that distinguishes between foreground and background sounds. First, a novel scoring mechanism is introduced to quantify the perceived importance of individual sound events in complex auditory scenes. The scores, along with statistics from the behavioral responses, highlight privileged treatment toward specific sound classes. Furthermore, hierarchical clustering exposes perceptual biases tied to sound classes, delineating a continuum from foreground to background sounds. Behavioral responses, auditory salience, and pupillometry data validate this distinction, with more pronounced attentional shifts and rapid pupil dilation for foreground sound classes. In contrast, certain behaviors, like the relationship between identification reaction time and perceptual importance remain sound-class-independent. These findings emphasize the dynamic relationship between sound identity and auditory attention, offering a framework for understanding auditory scene analysis. Despite limitations in dataset size and scope, the study advances our understanding of how sound identity shapes perception and contributes to our understanding of the mechanisms governing auditory cognition.

2aPP23. Harmonic relations among musical notes affect rhythm judgments for chord sequences. Deborah Fu (Otolaryngol. – Head and Neck Surgery, Univ. of Michigan, Ann Arbor, MI, debfu@med.umich.edu), Mason Shields, Carolyn Kroger (Otolaryngol. – Head and Neck Surgery, Univ. of Michigan, Ann Arbor, MI), and Anahita H. Mehta (Univ. of Michigan, Ann Arbor, MI)

The *auditory grouping hypothesis* posits that auditory stimuli sharing similar features, such as pitch and spatial location, are more readily perceived as part of a unified auditory event or object. This hypothesis can explain perceptual phenomena where changes in one auditory feature distort

judgments of a different feature. For example, the auditory kappa effect illustrates a bias in timing judgments among sequential sounds: those sharing more similar features are perceived as occurring closer in time than more distinct sounds. Higher-level contextual cues also influence perceptual grouping. For example, in music, listeners are better at identifying a target note in a sequence of chords when it is harmonically "distant" (e.g., in a different key) from the rest. The present study investigates whether musical harmony affects perceptual timing (i.e., rhythm) judgments similarly to low-level cues like pitch by exploring auditory kappa effects in sequences of musical chords. Participants judged whether the second chord in each 3-chord sequence was closer in time to the first or third chord. Preliminary data suggest that harmonically similar chords, rated higher in subjective musical "fit," are perceived as occurring closer together in time than chords with lower harmonic fit ratings.

2aPP24. The influence of momentary listening effort on accumulated fatigue. Michael L. Smith (Dept. of Speech Lang. Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, smit8854@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

People with hearing impairment report listening fatigue as a major barrier to social communication, but most investigations examine momentary effort without establishing its connection to longer-term fatigue. In this study, listeners completed a 60-min sentence-repetition task with an easy condition (intact sentences) or an effortful condition (sentences that demanded mentally repairing missing words). Pre- versus post-listening tasks were used to measure fatigue, including (1) reaction times, (2) verbal creativity, and (3) subjective report. During listening, tonic changes in pupil dilation, verbal reaction times, and repetition accuracy were measured. We hypothesize that repeated moments of effortful listening result in slower decay in pupil size as well as reduced verbal creativity, and increased reaction times in the later parts of the testing block. Conversely, listeners who hear only easy intact sentences are expected to have equivalent performance before and after the testing block. A lack of differences across conditions would contradict the notion that fatigue is a linear product of repeated moments of elevated effort. The value of effects shown in this paradigm will be to demonstrate the impact of fatigue on other concurrent abilities beyond speech perception.

2aPP25. Age-related temporal processing deficits: Relationship between temporal speech cue discrimination and gap detection abilities. Vanessa Reyes (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., College Park, MD 20742, vreyes13@terpmail.umd.edu), Anna R. Tinnemore, Erin Doyle, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Temporal processing abilities are crucial for encoding auditory information. These abilities decline with age, particularly impacting the ability to perceive brief temporal cues. Furthermore, age-related temporal processing deficits appear to be particularly detrimental for speech processing with degraded auditory input, such as that experienced by cochlear implant users. First, we hypothesized that reducing spectral information in speech would negatively impact temporal processing, with older listeners showing greater deficits. Twenty-two normal-hearing listeners, aged 20–70 years, completed temporal cue discrimination tasks involving silent interval duration using the stimuli dish versus ditch and gap detection tasks using vocoded speech. Results showed that the 50% crossover point for vocoded speech was about 5 ms later for older compared to younger listeners. Second, we hypothesized the individual variability in speech discrimination performance would correlate with gap detection thresholds. Results showed listeners with steeper word discrimination slopes also had better gap detection thresholds for the vocoded stimuli. These results emphasize the role of temporal processing in understanding speech with degraded spectral cues.

2aPP26. Speech perception and self-efficacy in adults cochlear implant users. Jay E. Garcia (5500 Campanile Dr., San Diego, CA 92182, jjgarcia8399@gmail.com)

Hearing loss (HL) significantly impacts individuals' lives, affecting speech communication, social interactions, and quality of life. Cochlear implants (CIs) offer a solution for adults with moderate to profound sensorineural HL, yet speech communication outcomes vary widely among users. While factors like auditory history, device features, and cognitive health contribute to outcome variability, malleable factors remain underexplored. This study examines self-efficacy (SE)—the belief in one's ability to complete tasks or succeed in specific situations—as a key factor affecting CI speech communication outcomes. Thirty-six postlingually deafened adult CI users (ages 35–83 years) completed a self-report SE survey and a multi-talker sentence comprehension task, involving true/false judgments. Results showed that SE was significantly correlated with speech comprehension accuracy, suggesting that higher SE may relate to better engagement in challenging listening environments and more effective communication strategies. Although SE was lower in CI users compared to a normal-hearing control group, the difference was not significant. These preliminary findings highlight the potential role of SE in determining speech communication outcomes among CI users. Future research will investigate the mechanisms underlying this relationship, to guide interventions aimed at enhancing SE and communication success for adult CI users.

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials and Phononic Crystals I

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Alexey Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, Bethesda, MD

Bogdan-Ioan Popa, Cochair
University of Michigan, 2350 Hayward Street, Ann Arbor, MI 48109

Behrooz Yousefzadeh, Cochair
Concordia University, Montreal, 1455 De Maisonneuve Blvd. W., Room EV-4.139, Montreal H3G 1M8, Canada

Hussein Nassar, Cochair

Chair's Introduction—7:55

Invited Papers

8:00

2aSA1. UnitcellHub: An open-source lattice design tool. Ryan Watkins (Jet Propulsion Lab., California Inst. of Technol., 4800 Oak Grove Dr., Pasadena, CA 91109, ryan.t.watkins@jpl.nasa.gov)

It is well known that the multiscale composition of lattice structures can be used to tune the acoustic response of systems, including desirable features like isolation and dispersion. With the advent of 3-D printing, the design domain for lattice structures has increased significantly; however, the design space is nearly infinite and nuanced with complex phenomena. Finding novel lattice structures for acoustics applications is thus challenging and has restricted their adoption. UnitcellHub is an open-source lattice design suite, enabling robust geometry creation, automated finite element simulation, database management, and an intuitive design space exploration interface. The software is composed of three distinct but interconnected submodules: (1) UnitcellEngine combines implicit geometry representation, meshless finite element analysis, and homogenization theory, to provide an automated yet robust pipeline from geometry definition to performance prediction. (2) UnitcellDB is a database of performance metrics for more than 15 000 lattice point designs spanning more than 30 types of unitcells and their many possible geometric realizations. (3) UnitcellApp is a desktop and web-based graphical user interface that builds upon these data and surrogate modeling to enable users to select the ideal lattice geometry for their given application. Together, UnitcellHub provides an end-to-end framework to understand and design lattices.

8:20

2aSA2. The application of acoustic and elastic metamaterials into the building envelope. Andrew Hall (Univ. of Auckland, Auckland, New Zealand, a.hall@auckland.ac.nz), George Dodd, Vladislav Sorokin (Univ. of Auckland, Auckland, New Zealand), and Emilio P. Calius (Computed Materiality, Auckland, New Zealand)

This research addresses two significant environmental impacts of urbanization and mechanization: noise pollution and ventilation. Noise pollution is increasingly recognized as a pervasive physical and mental health concern, linked to a growing array of medical conditions. Meanwhile, ventilation problems in many homes lead to excessive dampness, contributing to respiratory health issues. Current construction technology does not provide affordable solutions to these challenges, and the industry continues to rely on homogeneous materials. Cost-effective lightweight construction methods often fail to adequately reduce noise transmission between dwellings, while mechanical ventilation systems are costly, and passive trickle vents typically provide insufficient airflow. This study explores using metastructures and metasurfaces to improve sound insulation, with a focus on scalable, practical implementation. Consisting of metamaterial systems, they incorporate elements designed to reflect, absorb, and guide acoustic waves. The paper presents applications of locally resonant metamaterials, phononic crystals, subwavelength coiled acoustic resonators, and passive noise-cancellation waveguides utilizing Fano-resonance. Our findings demonstrate the effectiveness of these systems, with both experimental and simulation results showing a strong correlation. Diffuse-field testing indicates significant sound attenuation within the targeted frequency bands. We assess the advantages of each approach and identify the most effective methods.

8:40

2aSA3. A parametric study of metamaterial-based, acoustically insulating and ventilated louvers. Francesco Martellotta (Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy, francesco.martellotta@poliba.it), Chiara Rubino, and Stefania Liuzzi (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy)

The significant environmental benefits resulting from the use of natural ventilation in buildings are often accompanied by undesired side effects related to an increase in background noise due to the minimal sound insulation provided by an open window. To this purpose, many solutions have been identified in the literature, from partially open windows to metamaterials. The latter are among the most promising solutions, collecting significant research efforts, even though at the moment it seems difficult to achieve the desired performance, particularly in terms of bandwidth for building applications. Taking as a reference a louver system, the incorporation of a space-coiling metamaterial capable of originating a Fano-like resonance (due to the interference of monopolar/dipolar modes developing in the coil and the sound propagating through the ventilated opening), allows to obtain a relatively wide band sound attenuation in a range of about 1000 Hz at mid-frequency. Numerical simulation and parametric analysis are used to quantify the sound insulating performance of the proposed device while keeping the unit dimensions within practical limits, exploring how the system should be modified to maximize the acoustical performance without affecting the ventilation rate and possibly extend the response toward the low-frequency range.

9:00

2aSA4. Scalable design and microfabrication of graded metamaterial waveguides. Charles Dorn (Dept. of Aeronautics and Astronautics, Univ. of Washington, Guggenheim Hall 211, 3940 Benton Ln. NE, Seattle, WA 98195, cdorn@uw.edu), Vignesh Kannan (LMS, École Polytechnique, Palaiseau, France), and Dennis Kochmann (Mech. and Mater. Lab., ETH Zurich, Zurich, Switzerland)

While metamaterials have emerged as a powerful tool for manipulating elastic waves, scalability remains a key bottleneck. It is challenging to model, design, and fabricate architectures with large numbers of unit cells, which restricts the design space and achievable functionalities. We present a framework for scalable inverse design of spatially graded metamaterials, accompanied by a novel microfabrication technique to manufacture planar metamaterials spanning hundreds of thousands of unit cells and beyond. To address the scalability of computational design, we present an optimization framework leveraging ray tracing for efficient forward modeling. Using this framework, we design a set of spatially graded tiles, each spanning many unit cells to guide elastic waves in a prescribed way. We then assemble the tiles like puzzle pieces to achieve complex wave-guiding objectives. To realize our designs, we developed a microfabrication technique inspired by chip manufacturing methods. This enables the manufacture of free-standing planar truss metamaterials with tens to hundreds of thousands of unit cells. Wave guiding is experimentally demonstrated using a pulsed laser to excite elastic waves and interferometry to measure the response. Surprisingly broadband wave guiding is observed, demonstrating the promise of our scalable design and fabrication methods for on-chip wave manipulation.

9:20–9:40 Break

9:40

2aSA5. Roughness effects on phononic structures. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov), Rick A. Kellogg, Michael Denison, Dale E. Cillessen, and Ihab F. El-Kady (Sandia National Labs., Albuquerque, NM)

Phononic structures—whether crystals or pseudocrystals—are inevitably manufactured with finite tolerances, which limit their performance. We compare the modeled and measured transmission results for 3-D phononic isolation structures with finite roughness (from manufacturing in the measured case and simulated roughness based on surface scans of the phononic structures in the calculated case) to determine the degree to which surface roughness can explain measured deviations from the design performance of phononic structures. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

10:00

2aSA6. Vibroacoustic metamaterial distribution optimization for increased sound insulation and mass reduction. Klara Chojnacka (AGH Univ. of Krakow, Mickiewicza Av. 30, Cracow 30-059, Poland, klara.chojnacka@agh.edu.pl)

Locally resonant metamaterials enhance sound and vibration reduction within specific frequency ranges when attached to a base plate. Typically, these metamaterials are designed with periodically distributed elements across the surface, maintaining subwavelength spacing. Using periodic boundary conditions simplifies calculations and reduces the computational power needed compared to finite-sized models. However, no other practical background states that the elements should be distributed evenly in the whole available space for maximum metamaterial effectiveness. The effectiveness of each resonant element depends on the displacement amplitude of the base plate, making elements placed in areas with lower vibration amplitude less effective than those in higher vibration regions. This work presents a topological optimization of the distribution of locally resonant elements and experimental validation of the concept. The Method of Moving Asymptotes is employed to minimize the number of resonant elements while preserving the initial sound reduction achieved by periodic metamaterials. Calculations were conducted in COMSOL Multiphysics with MATLAB, using a combined analytical and numerical approach for mechanical impedance and sound insulation simulation. Vibroacoustic measurements, such as Sound Reduction Index measurements in the reverberation chamber, were conducted in order to validate the simulation results.

10:20

2aSA7. Bridging fidelity gaps in the design of TPMS-based metamaterials using neural networks. Andrew B. Fanton (Phys., Penn State, University Park, PA), Yu-Tong Wang (Acoust., Penn State, 201 Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, ybw5392@psu.edu), and Yun Jing (Acoust., Penn State, State College, PA)

Accurate prediction of wave transmission in architected metamaterials remains challenging due to discrepancies between low-fidelity simulations and high-fidelity experimental measurements. Low-fidelity simulations rely on assumptions about material properties and are further influenced by manufacturing uncertainties and fabrication errors, leading to deviations from measured behaviors. To address this fidelity gap, we propose a novel multi-fidelity learning framework that integrates simulation and experimental data for precise prediction of transmission spectra based on design geometry parameters. Triply Periodic Minimal Surface (TPMS) lattices are selected as a proof of concept due to their mathematically defined geometries, which allow precise control of structural properties and their influence on vibration transmission. The proposed framework captures the intricate relationship between lattice type, relative density, and transmission response by fusing low-fidelity simulations with high-fidelity experimental data from vibration testing of 3-D-printed samples. Results demonstrate the framework's ability to predict experimental transmission curves with minimal error across a frequency range of 500 Hz to 12 kHz. This work represents a significant advancement in the predictive modeling of TPMS lattices and establishes a scalable, data-driven methodology for optimizing architected materials in vibration control and beyond.

10:40

2aSA8. Analysis of truncation resonances in 2-D periodic discrete lattices. Yichen Shi (Mech. Sci. and Eng., The Grainger College of Eng., Univ. of Illinois at Urbana-Champaign, 105 S Mathews Ave., Urbana, IL 61801, yichens6@illinois.edu), Vinod Ramakrishnan, and Kathryn Matlack (Mech. Sci. and Eng., The Grainger College of Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Truncation modes (TMs) are localized modes residing in phononic band gaps, and they have recently been explored for passive flow control and energy harvesting applications. However, current knowledge is limited to TMs in 1-D periodic lattices. Motivated by this, here we systematically investigate 2-D TMs of discrete phononic lattices with two types of rectangular unit cells. We present a decomposition method based on the transfer function framework to analyze 2-D TMs. Our study reveals that, even for simple unit cell configurations, the characterization of 2-D TMs can be complicated and unintuitive. Unlike 1-D TMs, a wavenumber–frequency pair may not fully describe the localization characteristics of 2-D TMs, which leads to various types of localization characteristics, such as diagonal localization. Consequently, we explore conditions of meaningful extension of 1-D TM concepts, e.g., decay rate, directionality, to 2-D TMs using our decomposition method, demonstrate how boundary conditions restrain 2-D TMs in certain partial bandgaps or full bandgaps, and show how boundary conditions at edges perpendicular to the TM affect its characteristics through orthogonal coupling. Finally, we show that the proposed decomposition method can be generalized to characterize 2-D TMs in more complex lattices.

TUESDAY MORNING, 20 MAY 2025 — STUDIO FOYER, 9:00 A.M. TO 11:00 A.M.

Session 2aSC

Speech Communication: Speech Production Poster Session I

Tessa Bent, Chair

Speech, Language and Hearing Sciences, Indiana University, 2631 East Discovery Parkway, Bloomington, IN 47408

All posters will be on display and all authors will be at their posters from 9:00 a.m. to 11:00 a.m.

Contributed Papers

2aSC1. Social expectations in speech perception: English-accented Korean in South Korea. Yeojin Jung (East Asian Lang. and Literatures, Univ. of Oregon, Friendly Hall, 1161 E 13th Ave., Eugene, OR 97403, yeojinj@uoregon.edu) and Kaori Idemaru (East Asian Lang. and Literatures, Univ. of Oregon, Eugene, OR)

What factors beyond acoustic features influence native speakers' perception of foreign-accented speech? This study investigated how social information impacts the perception of English-accented Korean using intelligibility and accent rating tasks. Native Korean listeners heard English-accented Korean stimuli paired with one of three guises and corresponding descriptions: a Caucasian guise described as an American learner of Korean, an Asian guise described as a Korean American, or a silhouette as a foreign learner of Korean. Preliminary results ($n = 25$ in each condition) indicated that intelligibility was highest for the Caucasian, followed by the Korean, and lowest for the silhouette condition, suggesting a more positive bias toward the Caucasian guise compared to the Korean-American guise. In terms of accentedness, the silhouette was judged as least accented, followed by the Caucasian guise, and then the Korean guise, again suggesting a positive bias toward the Caucasian guise compared to the Korean guise. These findings suggest that the exemplar model, which predicts compatible results for the Caucasian and Korean guises, does not fully explain the observed results. Factors such as listeners' language attitudes and notions of in- and out-group membership are considered as potential additional influences in the processing of accented speech.

2aSC2. Sociophonetics of mergers and monophthongization in New Orleans English. Dana Serditova (English Dept., Univ. of Freiburg, Rempartstraße 15, Freiburg 79098, Germany, dana.serditova@anglistik.uni-freiburg.de)

Due to its port city history and extensive migration (Garvey and Widmer, 1982), New Orleans is often considered linguistically distinct from the rest of the US South and has been compared to geographically distant dialects like New York City English (Carmichael and Becker, 2018; Dinkin and Carmichael, 2023). Both literature and local ethnographic accounts suggest that the accent remains misunderstood and underrepresented (Author, 2024). In this talk, I will analyze the speech of New Orleanians across three features: the PIN∼PEN and the FEEL∼FILL mergers, and /aɪ/-monophthongization. Drawing on a socially balanced and representative dataset of 115 sociolinguistic interviews, I conducted a detailed acoustic analysis, with advanced statistical methods such as Bhattacharyya Affinity, Generalized Additive Models, and Trajectory Length. I will demonstrate that New Orleans exhibits certain trends found in other urban centers in the US South regarding mergers (Tillery and Bailey, 2004; Koops *et al.*, 2008) but displays an intriguing age-related tendency for /aɪ/-monophthongization. I will also show that ethnoracial affiliation continues to be one of the key predictors of linguistic variation, with notable acoustic patterns observed among Creoles. This research contributes to our understanding of the sociophonetics of New Orleans English and challenges misconceptions and oversimplifications.

2aSC3. Effect of dialectal variation on listeners' perception of English interdental fricatives and lateral liquid sounds. William S. Morales (Div. of Comput. Sci. and Eng., Louisiana State Univ., Baton Rouge, LA 70803, wmoral2@lsu.edu), Hyunju Chung (Commun. Sci. and Disord., Louisiana State Univ., Baton Rouge, LA), and Irina Shport (Dept. of English, Louisiana State Univ., Baton Rouge, LA)

This study investigated the perception of English consonants that show variation in the U.S. South: interdental fricatives /θ ð/ produced with different degrees of stopping and lateral liquid /l/ produced with different degrees of vocalization. The question of whether the categorization of these consonants is racially biased was examined in a task, where participants rated sounds by clicking along a visual analog scale in two conditions: the audio only condition (word recordings) and the audio-visual condition (recordings paired with pictures of White and Black speakers). The scale represented the "th" – "t, "th" – "d, or "l" – "no l" continuum. A hypothesis that all listeners would perceive interdental fricatives as more stopped and the lateral liquid as more vocalized when paired with Black speaker pictures in the audio-visual condition was not supported. /θ/ and /l/ were perceived less as intended sounds (i.e., interdental and lateral liquid) when produced by Black speakers than by White speakers. Listeners' perceptions of /ð/ did not differ greatly by the listener or speaker's racial background. The results suggest an interaction between a dialectally salient sound target and speaker or listener race that is not influenced by additional visual speaker clues.

2aSC4. How talker and listener gender identity impact spontaneous speech descriptors. Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, tbent@iu.edu), Malachi Henry, Rowan Kilgore (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Brooke Merritt (Rehabilitation Sci., The Univ. of Texas at El Paso, El Paso, TX), Tzu Pei Tsai, and Amelia Xanders (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Gender is an aspect of identity that listeners easily extract from speech. Even in relatively unconstrained tasks, such as auditory-free classification, gender frequently emerges as a primary perceptual dimension. However, studies have typically envisioned gender as binary and only included cisgender talkers. Here, we test how talker and listener gender identity impact person's perception of speech using a free descriptor task. Sentences produced by 20 talkers representing five gender identities (cis man, cis woman, transgender man, transgender woman, nonbinary) were presented to cisgender and gender diverse listeners. Listeners described one talker from each gender category using words or short phrases. Responses were separated into descriptor categories: psychological, stimulus, sexual orientation and gender identity (SGI), physical, social, and other. Response distribution across gender categories was similar regardless of talker's gender identity. Gender diverse listeners utilized a greater variety of SGI words than cisgender listeners, suggesting a role for listener experience and identity in spontaneous speech descriptions. SGI descriptors tended to be produced earliest chronologically, suggesting high salience, but density distributions differed across talker genders. This study provides further evidence that gender is a salient speech cue for listeners, but person perception involves a complex interplay between listener and speaker characteristics.

2aSC5. Sociophonetic variation in Korean two-way tense/lax sibilant contrasts. Jeannene Matthews (Linguist, NC State Univ., 2211 Hillsborough St., Campus Box 8105, Raleigh, NC 27606, jrlang2@ncsu.edu)

This acoustic study of a publicly available corpus of Seoul Korean lays out the specifics of variation found in Korean tense and lax [s] inspeakers between the ages of 15 and 46. Previous variationist literature on Korean dialects is scant, but the narrow body that examines [s] manifestations focuses on geographic region (Cho *et al*., 2002; Lee, 2002) in contrast with Seoul Korean but does not differentiate its findings from other basic demographic information, such as age and gender. One recent study of gender- and age-based variation in Korean sibilants by Kong and Kang (2021) finds some evidence for age-graded differences in women's spectral peak of tense /s/ and some patchwork differences in younger men based on phonetic context. This study will provide a description of variation found in the production of the target contrast across 40 speakers of Seoul Korean, 20 men and 20 women, ages 15 to mid 40s. Nearly 46 000 tokens were extracted from the Seoul Corpus and analyzed for the duration, the center of gravity, spectral peak, slope and skew. Results found statistically significant differences in center of gravity between phonemes and several other age- and gender-graded differences in duration and spectral moments; arguably, the most significant findings pertain to duration. Overall, the study confirms the validity and intrigue behind considering sociolinguistic factors across phonemes in Korean.

2aSC6. Social priming of speech perception in Japanese: A preliminary study. Naho Orita (Faculty of Sci. and Eng., Waseda Univ., 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, orita@waseda.jp) and Utako Minai (Dept. of Linguist, Univ. of Kansas, Lawrence, KS)

Previous studies that examined the effect of a speaker's perceived race on speech perception in English have found mixed results (Babel and Russell 2015; McGowan 2015; Kutlu *et al*. 2022; McLaughlin and Van Engen 2024). To pave the way for a fuller account of this phenomenon, it is necessary to conduct experiments in languages other than English and regions other than Western countries. To fill in this gap, this study extends the method of McLaughlin and Van Engen (2024) to Japanese. We recruited undergraduate students who were born, raised, and are living in Japan ($N = 40$) and measured transcription accuracy of L1-Japanese speech in noise while subjects were presented with either an East Asian female's face or a Black female's face (between-subject design). Subjects also completed Affect and Attitude Questionnaires and Language Background Questionnaires after the transcription task. We found no priming effects when comparing the two pictures. Although the interaction between priming and questionnaire scores was non-significant, individual differences in attitudes toward the speaker, Status and Solidarity (approximately competence and warmth), significantly predicted overall performance, suggesting that listeners who rated the speaker's Status lower and Solidarity higher demonstrated better transcription accuracy.

2aSC7. Effect of listener race and gender on the social evaluation of creaky voice. Jayden Hall-Ingram (Speech-Lang. Pathol., North Carolina Agricultural and Tech. State Univ., Charlotte, NC), Samiyah Hart (Speech, Lang. and Hearing Sci., Indiana Univ., Indianapolis, IN), Aryahna Le Grand (Theatre, Univ. of Missouri, Columbia, MO), Monique Maerilyn T. Valdepenas (Speech, Lang., and Hearing Sci., Saint Louis Univ., 3700 Lindell Boulevard, St. Louis, MO 63103, moniquemaerilyn.valdepenas@slu.edu), Sarah R. Bellavance (Communicative Sci. and Disord., New York Univ., New York, NY), Lisa Davidson (Linguist, New York Univ., New York, NY), and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Nonpathological creaky voice is a common and natural voice quality with linguistic function, such as indicating the end of an utterance. However, it is also a target of gender-based prejudice, with studies suggesting that women who use creaky voice sound less hirable and less pleasant. A previous study in our lab suggested that not all listeners have negative attitudes toward naturally produced creaky voice when it appears sentence-finally. Instead, this previous study found that older women were the only group of listeners who rated the creaky voice among young women speakers as less pleasant, suggesting a form of self-group distancing. In the current study, we examined the perception of utterances with creaks present across multiple words and compared perception among Black and White older listeners. For hireability, creaky productions were rated more negatively overall, and an interaction suggested that women listeners had even more negative ratings. For pleasantness, creaky productions were rated more negatively overall, and an interaction suggested that White listeners had even more negative ratings. Furthermore, Black men rated all stimuli more pleasant than the other listener groups. Taken together, these findings provide some additional evidence for self-group distancing, as the speakers in the study were primarily White.

2aSC8. Durational variability of spontaneous and read speech: Comparison between English and Japanese. Yoichi Mukai (Modern Lang. Studies, Vancouver Island Univ., Nanaimo, BC, Canada), Daniel Brenner (Alameda, CA), and Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., 208 E. Pine Knoll Dr., P.O. Box 15045, Flagstaff, AZ 86011, benjamin.tucker@nau.edu)

The present work examines the cross-linguistic effects of speech style and phonetic reduction. Specifically, we focus on the durational variability of vowels and consonants in spontaneous and read speech in English and Japanese. Data were extracted from spoken corpora of English and Japanese and other read speech data for the two languages. The duration of the segments was extracted then for each segment in the dataset to explore differences in durational variability between the two languages and the two speech styles. Differences were found between spontaneous and read speech in English in both vocalic and consonantal measures. In contrast, the Japanese showed less variability, particularly in vocalic elements, with only the consonantal measure showing a difference. The results are discussed in terms of the interplay between speech style and phonetic reduction, suggesting both language-specific and language-independent patterns of reduction.

2aSC9. Listener judgments of perceived age. Carly Alston, Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), and Benjamin V. Tucker (Dept. of Commun. Sci. and Disord., Northern Arizona Univ., 208 E. Pine Knoll Dr., P.O. Box 15045, Flagstaff, AZ 86011, benjamin.tucker@nau.edu)

When listening to a talker, listeners not only hear their message but also make judgments about the indexical properties of the talker, such as their gender or mood. The present experiment involved listening to pairs of speech stimuli produced by a single talker and judging in which sample the talker was older. The speech stimuli were a subset of samples used by Hunter *et al*. (2015), who used recordings of a single talker taken over 48 years. The paired comparisons method was the same as that used by Ozdogan (2023), who had recordings of two talkers taken over 20 years. Thirteen young adult listeners with self-reported normal hearing completed the paired comparisons task. After completing this task, the listeners were asked to identify which acoustic features of the talker's voice caused them to perceive the talker as older versus younger. Contrary to Ozdogan (2023), where listeners performed at chance levels, the present listeners were able to correctly identify the older sample about 70% of the time. This improved performance may be due to the larger time period covered in the current samples (48 vs 20 years) and/or to the previous talkers being professional speakers while the current talker was not.

2aSC10. Introducing a web app for widespread collection of sociophonetic data. Lisa Sullivan (Oklahoma State Univ., Dept. of Linguist, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, lisa.sullivan10@okstate.edu) and Valerie Freeman (Oklahoma State Univ., Stillwater, OK)

Harnessing the power of recent technological developments that make remote collection of high-quality audio recordings feasible, we present a web app for collecting sociophonetic and other linguistic data in an educational, gamified way. It includes four components: (1) an account system allowing for participants' data from multiple tasks to be linked confidentially and for participants to create sub-profiles to collect data from others who would not use the app on their own (e.g., grandparents, kids); (2) linguistic data collection including production, perception, and survey tasks; (3) an entertaining educational component focused on dispelling language myths and engaging the public with linguistic issues; and (4) a reward system where participants earn badges and points by completing tasks and engaging with the educational component. Our first study to use it will map prevelar vowel production across North American English from user-supplied audio recordings. Another will collect peer perceptions of teens with cochlear implants from audio samples. A third will follow students' second-language abilities longitudinally after graduation. As we prepare for full-scale rollout, we invite feedback and suggestions for future expansion into other types of linguistic experiments and the utility for others to run experiments on our platform.

2aSC11. Acoustic and visual properties of vowels in Oakland CA: Preliminary results. Keith Johnson (Dept. of Linguist, UC Berkeley, Berkeley, CA 94720, keithjohnson@berkeley.edu) and Alexandra Pfiffner (Dept. of Linguist, UC Berkeley, Berkeley, CA)

The "Voices of Oakland" project aims to create a large audiovisual corpus of conversational speech produced by people who were born and raised in Oakland, California. To date, we have hour long recordings from about 50 people ranging in age from 19 to 101, and spanning all races and economic strata of this diverse city. In this report, we describe the corpus design and some of our methodological decisions in collecting and analyzing audiovisual data and then discuss some preliminary findings regarding back vowel fronting, a widespread sound change that is also a primary component of the California Vowel Shift. Preliminary analysis suggests that the typical GOOSE fronting ascribed to California English is, for these speakers, stylistically variable and may be a marker of social status. The paper will also present data on lip posture during vowels, exploring the role of lip spreading in the raising of F2 in back vowel fronting.

2aSC12. Comparing methodologies for emotion perception ratings in clear and conversational speech. Cameron Bolinder, Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South, 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), and Whitney Robison (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Several studies have found that "anger" is perceived more often in clear speech compared to conversational speech. To date, however, these studies have all used a forced-choice paradigm for emotion perception, asking listeners to select emotions from a set of six choices ("anger," "fear," "disgust," "happiness," "sadness," and "neutral"). Thus, the results of earlier studies are limited to these emotion categories. The current study expands upon previous results to determine if speaking style differences in perceived emotion are also found using (a) dimensional ratings of emotion, namely arousal and valence, and (b) single-word, free-response emotion ratings. A subset of stimuli that have previously been rated using categorical emotion ratings in Young *et al*. (2024) was presented to 20 healthy young adult listeners. For each stimulus, listeners rated either (a) the emotion category, arousal, and valence, or (b) the emotion they perceived using a single word. It is hypothesized that clear speech will have (a) higher arousal and lower valence than conversational speech using dimensional measures of emotion perception, and (b) that negative emotion words, such as "anger," "annoyance," and "irritation," will be reported more frequently for clear speech compared to conversational speech using free-response measures of emotion perception.

2aSC13. Perceived emotion in conversational, clear, and Lombard speech. Whitney Robison (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT), Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South, 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Elizabeth D. Young, and Cameron Bolinder (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Several studies from this laboratory have shown that when listeners are given a set of six emotions to choose from, they rate semantically neutral sentences as sounding angry more often when spoken in a clear speaking style (clear speech) than when the talker was instructed to speak in a conversational manner (conversational speech). The present study expanded this line of research to a new corpus of conversational and clear speech in which talkers heard their own voice through headphones in four different simulated speaking environments: quiet, two levels of white noise (i.e., "Lombard" speech), and reverberation. In an exploratory perceptual experiment, listeners with normal hearing heard 10 instances of a single semantically neutral sentence ("I looked up the word in the dictionary" with a variety of /bVd/ keywords) spoken by two male and two female talkers in four conditions: conversational speech in quiet, conversational speech in 63 dB SPL of white noise, clear speech in quiet, and clear speech in 63 dB SPL of white noise. Data analyses will determine whether, similar to clear speech, Lombard's speech is perceived as sounding angry more often than conversational

speech, and whether the effects on perceived emotion of speaking style and speaking environment interact.

2aSC14. Realization of just: Speech reduction in a high-frequency word. Ki Woong Moon (Univ. of Arizona, 1200 E University Blvd, Douglass 318B, Tucson, AZ 85721, kiwoongmoon@arizona.edu), Adrian T. Onyx, Miriam L. Kaylor, and Natasha Warner (Univ. of Arizona, Tucson, AZ)

Speech communication in conversational contexts often contains word forms distinct from careful speech, as speakers modify or omit segments in high-frequency words, producing reduced pronunciations (Aylett and Turk, 2006; Bybee, 2001; Fidelholz, 1975; Fosler-Lussier and Morgan, 1999; Hooper, 1976; Munson, 2007; Pluymaekers *et al*., 2005). The present study explores how the high-frequency word "just" in American English is realized with regard to its acoustic features, how the acoustic features of "just" and its reduction differ based on adjacent segment types, and how they vary depending on the predictability of word sequences. Results reveal that "just" is more reduced in duration and vowel articulation when followed by sibilants compared to vowels and other consonants. Preceding sibilants cause less fronting of the vowel's backness but do not significantly affect duration or vowel height. Additionally, higher word predictability correlates with increased reduction, although specific acoustic cues are not affected by the word predictability. These findings emphasize the systematic, context-dependent nature of speech reduction and support its characterization as a gradient phenomenon (Ernestus and Warner, 2011) with the observed variability in the reduced form of "just."

2aSC15. Effects of speaker gender on attitudes towards creaky voice in US English. Stefania Ruiz (Communicative Sci. and Disord., New York Univ., 665 Broadway, New York, NY 10012, sr5847@nyu.edu), Sarah R. Bellavance, and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

In US English, a creaky voice is a natural voice quality used to convey linguistic information. One common use of a creaky voice is to mark the end of a sentence or conversational turn. Despite its linguistic functions, several studies have suggested that listeners have negative attitudes toward creaky voice especially when used by younger women. A previous study in our lab suggested that not all listeners exhibit these negative attitudes when a creaky voice is produced at the ends of sentences. That study found that only older women rated creaky productions to be less pleasant, suggesting a type of self-group distancing. The current study extends these findings by including cisgender men as speakers in addition to cisgender women. Listeners are asked to rate both the hireability and pleasantness of speakers' voices. Utterances are evenly split between those with creaky voice on the final syllable and those without. The study will examine listener attitudes toward both cisgender men and women speakers and explore the scope of self-group distancing by examining listener gender as well.

2aSC16. Goal-oriented speech adaptation: Conversation between Cantonese tone mergers and non-mergers. Ivan Fong (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A1S6, Canada, ivan_fong@sfu.ca), Eunice Wong, Paul Tupper (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Joan Sereno (Linguist, Univ. of Kansas, Kansas City, KS), Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS), Dawn Behne (Psych., Norwegian Univ. of Sci. and Technol., Trondheim, Norway), and Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Adaptation occurs when speakers adjust their speech for their interlocutors. While previous findings reveal segmental adaptations, our study focuses on the suprasegmental Cantonese tones. Some Cantonese speakers ("mergers") merge Tone3 (mid-level) and Tone6 (low-level), and can unmerge them when shadowing "non-mergers." However, it still unclear is whether such changes result from acoustic mimicking or goal-oriented adaptations for intelligibility. This study examines Cantonese tone adaptations between a tone "merger" and a "non-merger" during an unscripted conversation task, utilizing a goal-oriented video game involving Tone3–Tone6 word confusions. Pre/post-game tone-word productions are also included to compare with online changes during the conversation. Acoustic analyses involve normalized average F0 in target Tone3 and Tone6 productions. Preliminary descriptive conversation data from five participant pairs show a trend of a greater Tone3–Tone6 distance in F0 by the "mergers" and a smaller distance by the "non-mergers" as the conversation progresses. Additionally, "mergers" and "non-mergers" tend to assimilate with each other in terms of Tone3–Tone6 distance. However, pre- versus post-game data reveal similar Tone3–Tone6 distance, although global F0 changes are observed post-game for all participants. Statistical analysis involving additional participants is being conducted to determine the significance of these trends.

2aSC17. An acoustic classification of the Chinese dialects. Hua Lin (School of Lang., Linguist and Cultures, Univ. of Victoria, Victoria, BC V8W2Y2, Canada, hualin@uvic.ca)

Dialect classification is one important task for linguistics. One tool that has not been applied extensively in dialect studies is acoustics. Questions that can be asked include if dialects can or should be classified acoustically and how the acoustic classification measures against traditional non-acoustic classification. The Chinese language, which has traditionally (non-acoustically) been classified into seven dialects (Lin, 2001; Norman, 1988), provides a good specimen to study acoustically. Thus, the present study examines acoustically four Chinese dialects: Mandarin, Cantonese, Shanghainese, and Taiwanese, using five acoustic parameters, including vowel quantity and "variance" measures such as the standard deviation of the durations of the consonants and vowels and the variability indexes of both vowel and consonant durations (Ramus *et al*., 1999; Grabe and Low, 2002). Twenty-four participants were recruited, six native speakers each from one of the four Chinese dialects. The participants were recorded reading the same texts in their native dialects. The analysis was done using the speech analysis software Praat. The results show that the acoustic classification, while supporting the traditional classification in some ways, differs from it in others, suggesting that acoustic classification may be used to complement the traditional dialect classification.

2aSC18. Do speakers' beliefs about language impact the variants they produce? Investigating style shifting in the Northern Cities Shift. Kaitlyn Owens (Indiana Univ. Bloomington, 355 North Eagleson Ave., GA 3151, Bloomington, IN 47405, kaitowen@iu.edu)

Whereas previous studies show that language attitudes influence the use of dialectal features in production (see Preston, 2011), little is known about how language attitudes influence language change. This study probes the role of language attitudes on production during dialect attrition by investigating two of the Northern Cities Shift (NCS) vowels in the English of Lower Michiganders: the raising and fronting of /æ/ and the fronting of /ɑ/. The NCS dialect is receding in Lower Michigan (Nesbitt, 2021a) and some Michiganders are aware that it is a regional marker (Nesbitt, 2021b). In this study, 41 participants completed two tasks: (1) production (spontaneous speech and wordlist) and (2) attitudes elicitation (NCS talker versus "Standard American English" talker). Our results show that as participants increasingly rate the NCS talker as speaking a lesser-quality English than the non-NCS talker, participants have increasingly lower /æ/ vowels ($p = 0.0451$) and increasingly posterior /ɑ/ vowels ($p = 0.0057$) in wordlist speech than spontaneous speech. There is no impact of attitudes on /æ/ fronting. This study is the first to suggest that some Michiganders style shift away from the NCS. Furthermore, we argue that speakers with the greatest preference for "Standard American English" are the drivers of this sociolinguistic conditioning.

2aSC19. Vowel duration and race in Raleigh, North Carolina. Stephen G. Black (Linguist, North Carolina State Univ., 2211 Hillsborough St., Box 8105, Raleigh, NC 27607, stgblack3@gmail.com)

Dialects usually have variables that are measurable, but not necessarily perceptible or salient. Duration of phonetic segments, such as vowels, may or may not be salient depending on the degree of duration variation from what a speaker deems acceptable. Despite its lack of association with AAL by most, Black speakers tend to have longer vowel durations than White speakers overall. These patterns hold true when taking into account internal

factors like the following voicing, place and manner of articulation. This study examines these durational differences between Black and White speakers of English in Raleigh, North Carolina born between 1918 and 1996 (n = 227), considering various social and internal factors.

2aSC20. An investigation of variation in the production of glottalized consonants. Richard A. Wright (Dept. of Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195-2425, rawright@uw.edu), Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ), Emily P. Ahn (Univ. of Washington, Seattle, WA), Bella L. Rae, Serene Wong (Linguist, Univ. of Washington, Seattle, WA), and Kyle Wunsch (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ)

In the study of the consonants of the world's languages, consonants with glottal airstream mechanisms are less well studied than their pulmonic counterparts despite being geographically widespread [I. Maddieson, In: Dryer, M. S. and Haspelmath, M. (eds.) WALS Online (v2020.4)]. In particular, there is very little large-scale research about the acoustic variability in their production in connected speech. In the present study, we investigate the acoustic variability present in the realization of ejectives and implosives from the online corpus of Georgian and kiSwahili [Ardila, R. *et al*., Proc. 12th Conf. on Lang. Res. and Eval. (LREC 2020), 4211–4215, 2020]. The Common Voice corpora contain recordings of speakers reading sentences. For the present study, we hand corrected automatically force-aligned recordings of the pulmonic (for comparison) and glottalized stop consonants in each language. We then use the aligned recordings to investigate the variability in a variety of acoustic measures (e.g., closure duration and rise time) that can be easily extracted from the segmental boundaries. We discuss the variability in the acoustic realization of the consonants and describe the articulatory aspects related to these productions.

2aSC21. Speaking fundamental frequency in cisgender and nonbinary speakers assigned female at birth. Yana Dalbir, Line Lloy (Communicative Sci. and Disord., New York Univ., New York, NY), and Susannah Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, New York, NY, svl2@nyu.edu)

Previous research related to fundamental frequency (F0) across speakers has resulted in published norms for vocal pitch by age and sex, which are often used in the field of speech-language pathology to assess whether a person's vocal pitch falls within this normative range. Due to both anatomical and socially learned factors, a large difference has been reported for male versus female speakers (e.g., Graddol and Swann, 1983; Hollien *et al*., 1997; Stoicheff, 1981). These studies, however, assumed either a sex or gender binary, as well as a normative alignment between sex and gender. The current study examines the average F0 of non-binary speakers assigned female at birth (AFAB, n = 15) and age-matched and cisgender women (n = 15). Speakers produced the first six sentences of the Rainbow Passage three times. By-sentence measures of average F0 will be extracted in Praat and used to better understand F0 distributions for each gender group. Based on previous literature showing that social factors affect F0, it is predicted that the average F0 of AFAB non-binary speakers will be lower than that cisgender women. Since F0 is the focus of gender-affirming voice therapy, these findings will contribute data on F0 values for non-binary clients.

2aSC22. Interactions between voice cues in the perception of transgender and cisgender voices. Victoria A. Sevich (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, sevich.1@osu.edu), Terrin N. Tamati (Dept. of Otolaryngol., Vanderbilt Univ. Medical Ctr., Nashville, TN), and Brooke Merritt (Dept. of Speech, Lang., and Hearing Sci., The Univ. of Texas at El Paso, El Paso, TX)

Acoustic voice cues, like fundamental frequency (f0) and vowel space area (VSA), are used by listeners to attribute gender to a talker. However, the variability of f0 and VSA in transgender and cisgender talkers makes it difficult to understand how listeners use these cues to make gender judgments. The goal of this study was to determine how listeners use f0 and VSA when making voice gender judgments in transgender and cisgender talkers. Forty-two cisgender listeners used nine-point scales to assess the perceived gender ("definitely male" to "definitely female") of 30 transgender (man, woman, non-binary) and 30 cisgender (man, woman) talkers. Acoustic measurements of f0 and VSA were computed for each talker, and linear mixed effects models predicted listener judgments from f0, VSA, talker gender identity, and interactions. Results suggest that VSA is an important cue for gender judgments, but that its weighting varies based on f0 and listeners' perception of gender prototypicality. For transgender talkers with non-prototypical f0 values (e.g., low f0 for trans women), listener judgments are highly dependent on VSA. Results reinforce that there is not a straightforward mapping between the selected voice acoustics and perceived voice gender.

2aSC23. Judging by accent or race? Assessing linguistic and racial bias among Brazilian EFL teacher-trainees. Bremdellin Gabriel Ramos (Linguist, Univ. of Iowa, 16 North Clinton St., Iowa City, IA 52245, bremdellingabriel-ramos@uiowa.edu), Hanna Kivistö-de Souza (Linguist, Federal Univ. of Santa Catarina, Florianópolis, Santa Catarina, Brazil), and Ethan Kutlu (Linguist, Univ. of Iowa, Iowa City, IA)

Globalization has spread the English language worldwide, but its colonial roots perpetuate discrimination against marginalized speakers. Speakers are judged based on their racial background and the prestige of their language variety. While studies on language attitudes toward English accents and speakers' race are extensive, Brazil remains underexplored. This paper investigated how English teacher trainees in Brazil evaluate high- and low-prestige English varieties, considering the racial backgrounds (Black/White) of both speakers and listeners. A total of 51 English teacher trainees from public universities across Brazil participated in a matched-guise test, rating General American, British, Southern US, South African, Spanish, and Portuguese English speakers on dimensions of status and solidarity. The results showed that standard native varieties were rated higher for both status and solidarity, whereas the listeners' native variety (Portuguese English) received lower evaluations. Notably, both Black and White English teacher trainees rated Black speakers more favorably for status and solidarity. The discussion explores social desirability bias and post-colonial awareness as potential explanations for the findings, calling for critical reflection on the structure of English undergraduate teaching programs to better prepare trainees for linguistically and racially diverse English language classrooms.

2aSC24. Changing trajectories of New Zealand English vowels. Stefon M. Flego (Virginia Tech, 181 Turner St NW, Blacksburg, VA 24061, stefonflego@gmail.com), Lynn Clark, Jen Hay, Gia Hurring, Elena Sheard (Univ. of Canterbury, Christchurch, New Zealand), Abby Walker (Virginia Tech, Blacksburg, VA), and Joshua Wilson Black (Univ. of Canterbury, Christchurch, New Zealand)

It is well established that New Zealand English vowels have undergone considerable change over the last century and a half (Gordon *et al*., 2004). In this paper, we look at further change in the vowel system by exploring formant trajectories for both monophthongs and diphthongs in 310 speakers in the Quakebox Corpus (Walsh *et al*., 2013, recorded in 2010/11), born between 1920 and 1993. For most age cohorts, we find evidence of a continuation of changes that have already been documented. For example, when compared with data from earlier recordings (i.e., Hay and MacClagan, 2010), we see continued lowering of the second half of MOUTH, where F1 now generally increases over the course of the vowel, /æo/ → /ɛɐ/, even for the oldest speakers in our dataset. We also see evidence of further diphthongization of FLEECE, and LOT and THOUGHT are considerably higher when compared with speakers from the ONZE corpus (see Brand *et al*., 2021). But the most striking pattern is clear reversals in almost all vowel changes for the youngest groups of speakers, born between 1985 and 1993 in our data (ages 18–25 at the time of recording), supporting complementary findings by Ross *et al*. (2023, 2024).

Session 2aSP

Signal Processing in Acoustics, Acoustical Oceanography and Computational Acoustics: Machine Learning in Underwater Acoustics I

Kendal Leftwich, Cochair
Physics, University of New Orleans, 1021 Science Building, New Orleans, LA 70148

Youngmin Choo, Cochair
Sejong University, 209 Neungdong-ro Gwangin-gu, Seoul 05006, Korea

Shaun Pies, Cochair
Physics, University of New Orleans, 2000 Lakeshore Drive, New Orleans, LA 70148

Chair's Introduction—7:55

Invited Paper

8:00

2aSP1. System-informed neural networks with deep ensembles for frequency detection enhancement. Youngmin Choo (Defense Systems Eng., Sejong Univ., 209 Neungdong-ro Gwangin-gu, Seoul 05006, Korea, ychoo@sejong.ac.kr) and Geunhwan Kim (Dept. of Elec., Electron., and Control Eng., Changwon National Univ. (CWNU), Changwon 51140, Republic of Korea)

We propose a deep learning-based frequency analysis framework, referred to as the system-informed neural network (SINN), which integrates the corresponding linear system model. The SINN architecture is based on the adaptive learned iterative soft shrinkage algorithm and incorporates the system model into its loss function. This design ensures good generalization, fast processing time, and solutions that align with the system model, similar to physics-informed neural networks. To enhance performance, we utilize deep ensemble-based uncertainty quantification. This involves training multiple SINNs with different initial weights on distinct training datasets, allowing them to produce diverse predictions for the same test sample. The proposed scheme is evaluated using *in situ* acoustic data containing 43 frequency components and compared against results from established frequency analysis methods.

Contributed Papers

8:20

2aSP2. System-informed neural network for solving linear inverse problem in underwater acoustics. Myoungin Shin (Agency for Defense Development, Jinhae P.O. Box 18, Changwon, Gyeongsangnam-do 51678, Korea, smopower@hotmail.com)

In underwater acoustics, various applications, such as frequency detection, direction of arrival (DOA) estimation, and localization, can be formulated as linear inverse problems. High-performance solvers for linear inverse problems are essential for the detection and tracking of marine objects such as surface ships and submarines. In this study, we propose a two-step training framework, comprising pre-training and fine-tuning, to implement a system-informed neural network (SINN) solver based on the system model of linear inverse problems, ensuring adaptability to various environments. The SINN architecture adopts an adaptive learned iterative shrinkage thresholding algorithm (Ada-LISTA) of linear inverse problem solver. To overcome the challenge of collecting a large volume of high-quality underwater acoustic data, we generate simulation data that reflects the physical characteristics of the system model and use it to train the model during the pre-training step. In the subsequent fine-tuning step, the model is retrained using a limited amount of underwater acoustic data, with the pre-trained weights serving as initial weight values. This two-step approach facilitates rapid adaptation of the model to the target environment under analysis. The SINN solver not only demonstrates superior restoration and denoising performance but also offers the advantage of reduced computational processing time.

8:40

2aSP3. Line spectrum extraction of underwater ship engine sound using graph convolutional network. Kibae Lee (Jeju National Univ., Jejudaehakro 102, Jejusi, Jeju 63243, Korea, kibae0211@gmail.com) and Chong Hyun Lee (Jeju National Univ., Jeju, Korea)

Line spectrum information is crucial for detecting underwater ship engine sounds in passive sonar systems. However, most studies rely on conventional methods or image processing techniques, which are often error-prone. While recent deep-learning approaches show promise, they require precise labels during training, limiting their practicality. To overcome this limitation, we propose a line spectrum extraction method using a graph convolutional network (GCN) that captures correlations between line spectra. The proposed method divides the spectrogram into image patches and represents these patches as a graph, including their connections. Subsequently, a GCN is trained using weak labels that contain significant errors to extract the line spectra. Experiments using publicly available underwater acoustic data demonstrate that the GCN trained with weak labels accurately extracts line spectra while capturing their correlations. Additionally, the proposed method shows superior performance compared to conventional approaches.

9:00

2aSP4. System-informed neural networks for broadband Direction-of-Arrival estimation. Gwiyeong LEE (Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Korea, kylee4549@sju.ac.kr), Mingu Kang, Wooyoung Hong, and Youngmin Choo (Sejong Univ., Seoul, Korea)

This study presents a high-resolution Direction-of-Arrival (DOA) estimation method designed for noise-dominant environments. By integrating physical insights into a machine learning framework, the proposed approach overcomes the limitations of existing methods, including high computational costs and performance degradation in extreme noise conditions. Utilizing a linear system model, incoming signals are expressed as combinations of basis vectors, with the typically complex-valued representation converted into the real domain to ensure compatibility with standard neural network architectures. The approach extends the linear system model from narrowband to broadband signals, further enhancing DOA estimation accuracy. This work offers a practical and efficient solution for DOA estimation in challenging sonar applications. [This work was supported by the Korea Research Institute for defense Technology planning and advancement (KRIT)—Grant funded by the Korea government (DAPA—Defense Acquisition Program Administration) (No. KRIT-CT-23-026, Integrated Underwater Surveillance Research Center for Adapting Future Technologies, 2025)]

9:20–9:40 Break

9:40

2aSP5. Domain-adaptive meta learning for underwater acoustic target recognition. Junho Bae (Sejong Univ., 209, Seoul 05006, Korea, n69bae@sju.ac.kr), Wooyoung Hong, and Youngmin Choo (Sejong Univ., Seoul, Korea)

This study provides a unique opportunity for advancements in underwater acoustic target recognition (UATR). The ShipsEar dataset, comprised of underwater acoustic recordings captured by hydrophones across eight distinct locations, encapsulates diverse environmental conditions and vessel sound characteristics. This study leverages meta-learning techniques for domain adaptation to enhance the generalizability of machine-learning models in cross-location vessel recognition tasks. By analyzing inter-location variations and identifying domain-invariant features, the proposed approach addresses key challenges in underwater acoustic signal processing, such as variability in ambient noise and recording conditions. The findings have potential applications in maritime safety, environmental monitoring, and naval operations. [This work was supported by the Korea Research Institute for defense Technology planning and advancement (KRIT) grant funded by the Korean government (DAPA—Defense Acquisition Program Administration) (No. 21-107-B00-008(KRIT-CT-23-009), The technical research for Underwater surveillance in open-sea and the analytic technique of acoustical environments in deep water, 2025)]

10:00

2aSP6. Underwater ship engine sound classification using Tensor Factorized Neural Network. Hyun Hee Yim (Jeju National Univ., 102, Jejudaehak-ro, Jeju-si 63243, Jeju-do, Korea, hyhyun0817@stu.jejunu.ac.kr), Kibae Lee, and Chong Hyun Lee (Jeju National Univ., Jeju, Korea)

Existing studies primarily utilize spectrograms generated through the short-time Fourier transform (STFT) as features for underwater ship engine sound classification. However, these features are constrained by the parameters defined during the STFT process, potentially restricting the information required for classification. To address this limitation, this paper proposes constructing multi-resolution spectrograms generated using various window sizes into a three-dimensional tensor, which is then classified using a Tensor Factorized Neural Network (TFNN). This three-dimensional tensor combines features across multiple time and frequency resolutions, enabling more efficient learning. Experimental results using a publicly available ship engine sound dataset demonstrate that employing tensor features structured in three dimensions achieves superior classification performance compared to previously used features.

10:20

2aSP7. Underwater target classification using a deep neural network model based on fast Fourier transform. June Hee Park (Ocean Systems Eng., Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, 1216B, Seoul 05006, Korea, junehee29@gmail.com) and Jonghoek Kim (Defense Systems Eng., Sejong Univ., Seoul, Korea)

Detection and classification often involve converting visual images or audio data into images, which are then input into a Convolutional Neural Network (CNN). However, this approach requires significant computational resources, necessitating high-performance hardware and imposing a substantial burden on the system, particularly for real-time detection. This study proposes the use of a Deep Neural Network (DNN) model for real-time underwater target classification through frequency analysis. Preprocessing is necessary to apply audio data to the DNN model. In this study, audio data are transformed into the frequency domain using Fast Fourier Transform (FFT). The values of each frequency domain are separated by commas, and the class is stored at the end of each row to create a text file dataset. By training the model on frequency ranges that are characteristic of each class, unnecessary data can be excluded, allowing for the use of a relatively simple deep-learning model. Based on experimental results, the performance of the proposed DNN model is verified by comparing it with existing CNN models. The DNN model proposed in this study is expected to enable underwater target classification with fast processing and low costs.

10:40

2aSP8. The impact of water sound speed on deep learning seabed classification from shipping noise. Alexandra M. Hopps-McDaniel (Dept. of Elec. Eng. and Comput. Sci., Univ. of Wyoming, 411 S 26th St. Apt 1, Laramie, WY 82070, mcdaniel.alexh@gmail.com), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA), Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Previous studies have shown that ship noise spectrograms can be used for deep-learning seabed classification when the water sound speed profile (SSP) is known. This work addresses the challenge of seabed classification with unknown SSPs. Four synthetic training datasets were generated, each featuring the same catalog of 34 seabeds but with different SSPs. The four sets of SSPs are (1) slightly downward refracting SSPs measured during the Seabed Characterization Experiment (SBCEX) of 2017, (2) SSPs measured during SBCEX 2022, which exhibit greater variation with depth, (3) a combination of the measured SSPs from SBCEX 2017 and 2022, and (4) a collection of SSPs from the World Ocean Atlas database, sourced from locations where the water depth is similar to the experiment location. Each training dataset was used to train ResNet-18 models for seabed classification, employing different combinations of hydrophone depths from vertical line arrays (VLAs). Measured spectrograms of ship noise from both SBCEX 2017 and 2022 are used as testing data for the trained models. Comparison of the seabed predictions from the different cases demonstrated the importance of including sound speed variability in training deep learning models for ocean acoustics applications. [Work supported by ONR.]

Session 2aUW

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Computational Acoustics: Ambient Sound Measurements and Models I

Martin Siderius, Cochair
Portland State University, 1600 SW 4th Avenue, Suite 260, Portland, OR 97201

S. B. Martin, Cochair
Halifax, JASCO Applied Sciences, 20 Mount Hope Avenue, Dartmouth, B2Y 4S3, Canada

Kay L. Gemba, Cochair
Naval Postgraduate School, 833 Dyer Road, Bldg. 232, NPS, Physics Department, Monterey, CA 93943

Jie Yang, Cochair
Applied Physics Lab., Univ. of Washington, 1015 NE 40th St., Seattle, WA 98105

Chair's Introduction—7:00

Invited Papers

7:05

2aUW1. The role of bubbles in ocean ambient sound: Measurement techniques and modeling approaches from whitecaps to melting glacier ice. Grant B. Deane (Scripps Inst. of Oceanogr., Code 0238, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu)

Bubbles play an important role in generating ocean ambient sound under wind-driven seas and in polar regions near icebergs, glaciers, and ice sheets. The acoustic signals produced by bubbles in these environments have been used to estimate wind speed and the mass of calving tidewater glaciers, for example, and may hold significant potential as a remote monitoring tool for receding tidewater glaciers. Combining measurement and modeling approaches offers a robust framework for interpreting ocean ambient sound from bubbles, though each comes with its own challenges. Some of these challenges will be discussed along with advances in our understanding of the fluid dynamics underlying bubble sound generation.

7:25

2aUW2. Overview of wind generated ambient sound modeling techniques with data comparisons. Martin Siderius (Portland State Univ., 1600 SW 4th Ave., Ste. 260, Portland, OR 97201, siderius@pdx.edu) and S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada)

In recent years, there has been a growing interest in understanding ocean soundscapes, with wind-driven surface sound being a major contributor. Acoustic modeling of wind-generated sound has been studied extensively; however, inconsistencies in terminology and definitions have led to discrepancies in the predicted sound pressure levels (SPL). The issue became particularly apparent at the 2022 Ambient Sound Modeling Workshop, held in conjunction with the Effects of Noise on Aquatic Life 2022 Conference, in Berlin, Germany. A well-defined scenario was used to compare results from different models and researchers, revealing significant variations, with discrepancies reaching up to 10 dB in some cases. This presentation will review common approaches to modeling wind-generated sound including the ray-based method from Harrison [J. Acoust. Soc. Am. 99(4), 1996] and the wave-based approach developed by Kuperman and Ingenito [J. Acoust. Soc. Am. 67(6), 1980]. Issues related to incorporating the wind-based source-level model will be discussed. Additionally, the modeling results will be compared to measured data for several sites, including the one used in the Berlin workshop.

7:45

2aUW3. Estimating the source level for wind-driven underwater noise from long-term archival data. S. B. Martin (Halifax, JASCO Appl. Sci., 20 Mount Hope Ave., Dartmouth, NS B2Y 4S3, Canada, bruce.martin@jasco.com), Martin Siderius (Portland State Univ., Portland, OR), and Mojgan Mirzaei Hotkani (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada)

Sound from wind and waves sets the background sound levels throughout the ocean. An accurate source level for wind is needed to estimate the ambient sound levels for sound exposure modeling, environmental assessments, or detection performance of sonars. Previous models had a constant roll-off of the sound levels at −16 dB/decade at all wind speeds, and the source levels at frequencies below ∼1 kHz were flat due to a lack of measurements. Here, we analyzed 14 long-term archival datasets with limited anthropogenic sound sources to estimate the wind-driven source level. We estimated the site-specific propagation loss using a ray-based model, then added the loss to the median received levels at each wind speed to obtain the source level. The results showed that there is a peak in the source level at ∼500 Hz. At high

frequencies, the roll-off increases with wind speed, from ∼12 dB/decade at 3 m/s to >25 dB/decade at 20 m/s. An analytic equation for the aeric dipole source level will be provided that increases as wind speed cubed, like most other air–ocean coupling processes. A comparison of the new model to previous direct source level measurements and the Wenz curves will be provided.

8:05

2aUW4. A review of wind sound source level derived from ambient noise data. Ross Chapman (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

Experimental estimates of source levels of sound generated by local wind at the sea surface are derived from ambient noise data based upon assumptions about the nature of the wind sound source. Generally, the sound source is assumed to be a distribution of monopole sources near the sea surface or a distribution of dipoles at the sea surface. Depending on the assumption made, the source level estimates for low frequencies (<∼400 Hz) can be significantly different by several decibels. These differences have led to confusion about the nature of the sound source in modeling wind noise. This paper presents a critical review of experimental techniques and analysis procedures for deriving wind sound source levels from low-frequency ambient noise data. The study finds that the differences in published estimates from experiments with vertical line hydrophone arrays can be understood through a simple analysis of the source assumptions. The work shows that the source levels agree well for wind speeds greater than 5 m/s. However, knowledge gaps exist about source levels at very low frequencies below 100 Hz and wind speeds less than 5 m/s. New experiments with vertical line arrays are needed to resolve these questions.

8:25

2aUW5. Utility of ocean wave parameters in ambient noise prediction. William E. Rogers (Code 7322, U.S. Naval Res. Lab., NRL Code 7322, Stennis Space Ctr., MS 39529, w.e.rogers.civ@us.navy.mil), Laurie T. Fialkowski, Daniel J. Brooker (Code 7160, U.S. Naval Res. Lab., Washington, DC), Gleb Panteleev (Code 7322, U.S. Naval Res. Lab., Stennis Space Ctr., MS), and Joseph M. Fialkowski (Washington, DC)

This study is concerned with the prediction of the "wind noise" component of ambient noise (AN) in the ocean. It builds on the seminal paper by Felizardo and Melville (1995), in which the authors quantified the correlation between AN and individual wind/wave parameters. Acoustic data are obtained from hydrophones at six diverse locations, and wind/wave parameters are obtained from moored buoys and numerical models. We describe a procedure developed for this study which identifies the correlation of AN with wave parameters, independent of their mutual correlation with wind speed. We then describe paired calibration/prediction experiments, whereby multiple wind/wave parameters are used simultaneously to estimate AN. We find that the improvement from the inclusion of wave parameters is robust but marginal: typically RMSE is reduced by less than 0.3 dB and/or less than 12% of the original RMSE. We interpret the latter outcome as suggesting that wave breaking responds to changes in local winds quickly, relative to, for example, total wave energy, which develops more slowly. This outcome is consistent with a prior study of the dependence of whitecap coverage on wave state, Hwang and Sletten (2008). We discuss this in the context of the time/space response of various wave parameters to wind forcing.

Contributed Paper

8:45

2aUW6. Empirical ambient noise model based on wind speed using two-decade open ocean ambient noise data in the frequency band of 1–20 kHz. Jie Yang (Appl. Phys. Lab., Univ. of Washington, 1015 NE 40th St., Seattle, WA 98105, jieyang@uw.edu), Stephen Riser (School of Oceanogr., Univ. of Washington, Seattle, WA), Jeffrey A. Nystuen, and Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In the frequency band of 1–20 kHz, wind-generated breaking surface waves produce bubbles near the surface that are the dominant ambient noise source. In previous work, two decades of ambient noise data from six deep ocean moorings were used to validate ambient noise models (Yang *et al*., JASA EL 3(3), 2023). Data-model comparisons show a mismatch, as existing models are monotonic in nature, i.e., the modeled spectral level increases with increasing wind speed for all frequencies, while data display a sharp drop-off that creates a "cross-over" as the spectral level for wind speed exceeding 15 m/s and frequency above ∼4 kHz becomes lower than that at lower wind speeds. This mismatch is due to attenuation when ambient sound propagates through the deeper and denser bubble layer under high sea conditions. In this work, an empirical ambient noise model utilizing wind speed only is presented as a baseline prediction with potential fine-tuning parameters such as bubble statistics, current and its direction, and wave height discussed. [Work supported by NOAA, NASA, and ONR.]

9:05–9:25 Break

Invited Paper

9:25

2aUW7. Wind-generated ambient noise in the deep ocean trenches. Michael J. Buckingham (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Under the influence of the wind, breaking waves create bubbles, which oscillate as monopoles thus acting as efficient sources of sound. In the deep ocean trenches, bottom reflections are often negligible and the bubble-generated ambient noise is primarily downward traveling. The Deep Sound instrument platform is designed to record the noise on pairs of hydrophones, aligned vertically and horizontally, to depths as great as 11 000 m. Deep Sound consists of a Vitrovex glass sphere, three recovery antennas, a high-performance data

acquisition system, inertial navigation, and a CTD plus sound speed sensor, with power provided by lithium-ion batteries. It descends under gravity, and releases a drop weight at depth, at which point it returns to the surface under buoyancy, collecting sound speed and acoustic data on the descent and ascent. Deep Sound has been deployed in the Challenger Deep in the Mariana Trench, the Tonga Trench, the Sirena Deep, and the Philippine Sea. One of the conclusions from the data is that the noise field in the deep trenches conforms to the simple Cron and Sherman theoretical model provided that the local sound speed is used in the computation of the spatial coherence of the noise. [Research supported by ONR.]

Contributed Papers

9:45

2aUW8. Comparison of full-wave and ray-based models of surface-generated ambient underwater sound. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu), Nicholas C. Durofchalk, and Matthew Mazzella (Phys. Dept., Naval Postgrad. School, Monterey, CA)

Here, theoretical techniques developed in the context of acoustic noise interferometry are applied to the problem of characterizing the ambient sound field in the ocean. This work is motivated by observations of deep-water ambient sound by a network of moored autonomous acoustic noise recorders in the 2023 New England Seamounts Acoustics (NESMA) Pilot experiment and the 2024 NESMA experiment. Exact results for the intensity of noise generated by random sound sources distributed on a pressure-release surface are derived and compared to ray-theoretical models. Contributions of evanescent waves to the noise intensity are quantified in the presence of sharp interfaces and in a continuously stratified environment. Elementary recipes are proposed to account for the effect of sound attenuation on noise intensity in a few basic scenarios. Using reciprocity relations and, in the case of the ocean with currents, flow-reversal theorem for directional sound sources, a computationally efficient approach is proposed to model individual realizations and statistics of the field due to random noise sources on the ocean surface. The approach seamlessly combines full-field propagation models with the directional noise source models, which were previously reserved for homogeneous water column and/or ray-based descriptions of ambient sound. [Work supported by ONR.]

10:05

2aUW9. Statistical analysis of wind-driven ambient noise impacts to mid-frequency acoustic transmission loss variability in Shallow Water Bay. Grant Eastland (Test & Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

Environmental variations, such as changes in sound speed profile caused by internal waves, lead to fluctuations in transmission loss from an acoustic source at specific ranges and depths. Data collected by NUWC Division Keyport and APL-UW in 2020 using a mid-frequency, vertical aperture receiving array in relatively shallow water ($\sim$180 m depth) allowed for the measurement of both environmental and resultant acoustic fluctuations. The statistical analysis of the variability in 3.5 kHz source transmissions over a 5.5-km range is presented using histograms of the signal vertical angle of arrival and transmission loss over time. Additionally, factors like freshwater runoff, tidal actions, and wind significantly influence the observed 3.5 kHz source arrivals (including direct path and surface bounce) on the vertical array.

10:25

2aUW10. Quantitively measuring differences between ambient sound environments using statistical distance metrics. Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., MC 0206, La Jolla, CA 92093, athode@ucsd.edu), Alison B. Laferriere, Kevin Souhrada, and Océane Boulais (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

A recurring challenge in ambient sound studies is determining whether sensors placed at two different locations (e.g., drifting over a seamount versus abyssal plain) or at different times (e.g., before and after a coral reef degrades) experience the same ambient acoustic environment, or "soundscape." Here, we examine various candidate metrics for quantitatively measuring the difference between two ambient environments. The ambient environment is modeled as a multidimensional probability distribution, with different measurements (sound pressure level, dominant directionality) defining different axes. Temporal patterns (such as periodic variations in intensity or impulse sequences) are mapped as additional axes representing time-delayed measurements. Various metrics and divergences like the inner product, relative entropy (Kullback–Leibler divergence), total variation distance, and Wasserstein are then used to compute the "distance" between a given environment and a "reference" environment. The resulting outputs reduce a multidimensional distribution into a single scalar, easing visualization of changes in broadband (potentially directional) ambient sound data. Examples of applying this approach to bulk data are shown on data collected from both deep-water drifters and shallow coral reefs, including an example that helped flag a sensor noise issue. [Work supported by ONR TFO.]

Invited Paper

10:45

2aUW11. Development of ambient noise measurement technique and its applications on Argo floats over the past two decades. Jie Yang (Appl. Phys. Lab., Univ. of Washington, 1015 NE 40th St., Seattle, WA 98105, jieyang@uw.edu), Stephen Riser (School of Oceanogr., Univ. of Washington, Seattle, WA), Jeffrey A. Nystuen, and Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The distinctive underwater sound generated by raindrops on the ocean surface has been used to detect and quantify rainfall. Knowledge of the intensity and spatial–temporal distribution of rainfall over the ocean is critical in understanding the global hydrological cycle. However, rainfall is difficult to measure accurately over the ocean due to its spatial and temporal variability. To reduce these problems, satellite-based rain-monitoring instruments are used but they do not capture the full *in situ* temporal and spatial variability. The Passive Aquatic Listener (PAL) was developed at the University of Washington's Applied Physics Laboratory (Nystuen, J. Acoust. Soc. 79, 92–98). PAL has been incorporated into Argo floats and deployed over global oceans. PAL-Argos are capable of telemetering back estimated rain rate and wind speed with a temporal resolution of 2–8 min, representing a circular surface footprint of a few kilometer radius (Yang *et al.*, *Oceanography* 28, 124–133). In this work, the major operations of PAL-Argos in NASA's Aquaris Satellite Mission, SPURS field efforts related to the Tropical Rainfall Measuring Mission, NOAA's Tropical Pacific Observing System Initiative, and NSF's Measurements and Modelling of the Indonesian Throughflow are chronicled with selective field data presented. [Work supported by NASA, NOAA, and NSF.]

Session 2pAAa

Architectural Acoustics, Noise, Speech Communication, and Psychological and Physiological Acoustics: At the Intersection of Speech and Architecture II

Kenneth Good, Cochair
Architecture Acoustics, Armstrong World Industries, 2500 Columbia Avenue, Lancaster, PA 17601

Evelyn Hoglund, Cochair
Speech and Hearing, Ohio State University, 104a Pressey Hall, 1070 Carmack Road, Columbus, OH 43210

Pasquale Bottalico, Cochair
Department of Speech and Hearing Science, University of Illinois at Urbana-Champaign, 901 South Sixth Street, Champaign, IL 61820

Arianna Astolfi, Cochair
Energy, Politecnico di Torino, Corso Duca degli Abruzzi, 24, Torino 10129, Italy

Invited Papers

1:00

2pAAa1. Examining the effects of reverberant environments and interaural level differences for bilateral cochlear implant users. Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, jaronoff@illinois.edu) and Prajna BK (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Reverberant environments and background noise each pose significant challenges for cochlear implant listeners, and their combined effects can be even more detrimental. The effect of reverberation may be magnified for bilateral cochlear implant users since reverberant environments can result in a decrease in the interaural coherence of signals to the two ears, which can degrade binaural abilities. Previous research from our laboratory has shown that adding an interaural level difference can help foster binaural fusion in the presence of decreased interaural coherence. This study examined if adding an interaural level difference can also improve speech perception in reverberant environments for bilateral cochlear implant users. Participants were tested using AzBio sentences with a multi-talker babble background, either with or without a simulated reverberant environment and with or without the addition of a large interaural level difference. Preliminary data indicates that performance markedly decreased with the simulated reverberant environment. However, adding a large interaural level difference improved speech recognition in a reverberant environment, but not for all participants.

1:20

2pAAa2. The role of talker sex and body mass index in horizontal speech directivity. Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, monson@illinois.edu) and Allison Trine (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

Human speech directivity plays a role in speech perception, including speech recognition in complex acoustic environments. Individual variability in speech directivity patterns can arise from differences in the physical attributes of the talker. We examined this variability in horizontal speech directivity using anechoic, multi-channel, high-fidelity recordings of 15 female and 15 male talkers. We calculated the frequency-dependent directivity index (DI) for each talker and tested the effects of frequency, sex, and body mass index (BMI) on the DI. There was a nonmonotonic relationship between DI and frequency. Minimum radiation toward 0° and maximum radiation toward 90° occurred around 700–850 Hz, resulting in a spectral boost at 90° at these frequencies. An effect of sex was limited to frequencies near 1 and 6–7 kHz, with male speech being more directional at these frequencies. BMI was associated with the minimum DI magnitude and the frequency locus of the minimum DI at 0°, with higher BMI associated with a higher frequency locus. Talkers' physical characteristics shape speech directivity, which may have implications for speech perception in real-world, multi-talker acoustic environments. [Work supported by NIH grant R01-DC019745.]

Contributed Papers

1:40

2pAAa3. The Oscar Larson Performing Arts Center Theater—Constrained volume and targeted reflections for speech intelligibility. David Kahn (Acoust. Distinctions, Stamford, CT) and Arthur W. van der Harten (Acoust. Distinctions, 400 Main St., Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com)

In 2019, the South Dakota State University Oscar Larson Performing Arts Center opened in 2019 to great acclaim. The 850-seat theater has been a "game changer" for performing arts in Eastern South Dakota. Hosting lectures, road shows, musical theater, dance, and a wide range of other theatrical performance types, the needs of the theater included a speech heavy program with a goal of being able to host unamplified dramas without any sound reinforcement—a challenging program for an 850-seat theater. In this paper, we discuss the design of the theater, and the strategy for the support of speech as well as the other needs of the space, including an overall constraint of the reverberant volume with stringent management of both beneficial (early, frequency-balanced) and destructive (late, focusing) sound reflections using 3-D modeling. We finish with a discussion of test results, and end-user impressions concerning the intelligibility, and overall acoustic functionality of the theater.

2:00

2pAAa4. High-resolution measurement of voice directivity in the horizontal plane. Pablo Abehsera-Morell (Sorbonne Université, CNRS, 4 Pl. Jussieu, Paris 75005, France, pablo.abehsera_morell@sorbonne-universite.fr), Paul Luizard (TU Berlin, Berlin, Germany), David Poirier-Quinot, and Brian F. Katz (Sorbonne Université, CNRS, Paris, France)

The human voice is directional by nature, and its directivity has been the subject of extensive study. Accurate data on voice directivity are fundamental in any work associating natural human speech and its interaction with architectural spaces. In this research, we present and analyze results obtained using a novel high-resolution measurement setup. The system features an array of 180 MEMS microphones arranged on a horizontal circle with a 3-m diameter, enabling measurement of sound sources with a 2° resolution in the azimuthal plane. Test subjects were positioned at the center of the array, with the location of their mouths precisely calibrated using a multi-camera and laser alignment system. Data were collected from 24 talkers (5 female, 19 male) articulating fluent English speech. The analysis, conducted across different frequency bands, explores variations between sexes and compares findings with previous studies. We discuss the relevance of the resolution provided by the system and examine its performance within the context of existing literature. Potential directions for future work are also outlined.

Invited Papers

2:20

2pAAa5. Virtual reality and voice: Sensory integration and feasibility of a voice intervention. Charles J. Nudelman (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., M/C 482, Champaign, IL 61820, nudelma2@illinois.edu), Asritha Tunuguntla, Naomi Ha, Isabella Rogala, Kaliyah House-Henry, Bella Lopez, Ricardo Perez Guerrero, and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

This study assesses the contributions of varying levels of sensory input during voice production in virtual reality (VR). The influence of auralized room acoustics and visual scenes on voice and the feasibility of VR-based voice therapy were investigated. Two groups of participants were involved—(1) 47 young adults (18–27 years) and (2) 10 pre-service teachers (18–19 years). Group one performed speech tasks in varying VR conditions: (i) auralized, (ii) visual-only, and (iii) multisensory (audiovisual), while Group two performed speech tasks in varying real and VR conditions: (i) control, (ii) teaching style, and (iii) VR intervention. Additionally, with Group two, clinician-mediated feedback was provided in the VR. Voice parameters were analyzed. Multisensory VR environments significantly ($p < .05$) influenced voice outcomes. Larger, noisy, and densely occupied VR spaces had more pronounced effects. The VR intervention resulted in a significantly lower time dose ($p < .05$) compared to the control condition. Real-time clinician feedback within VR resulted in reduced SPL ($p < .05$), fundamental frequency ($p < .05$), and time dose ($p < .05$) compared to instances without clinician feedback. The results demonstrate that VR environments can significantly alter voice and may aid voice therapy. [Work supported by Raymond H. Stetson Scholarship in Phonetics and Speech Science (2024)]

2:40

2pAAa6. The impact of noise and dysphonic voice on vowel recognition in 8–12-year-old children. Mary M. Flaherty (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, maryflah@illinois.edu), Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Keiko Ishikawa (Commun. Sci. and Disord., Univ. of Kentucky, Lexington, KY)

This study examined the effects of noise and voice quality on vowel recognition in 8–12-year-old children. Vowels were recorded in an /hVd/ context by six female talkers: three with normal voices and three with mild to severely dysphonic voices. Children completed a forced-choice vowel recognition task, identifying vowels presented in talker-specific speech-shaped noise at two signal-to-noise ratios (SNRs): −6 and −9 dB. Accuracy was analyzed by voice quality, talker, SNR, child age, and their interactions. Preliminary analyses revealed significantly higher vowel recognition for normal voices across both SNRs, with performance declining at −9 dB for both voice types. An age-by-voice quality interaction showed that younger children performed poorly in both conditions, with minimal differences between normal and dysphonic voices. In contrast, older children demonstrated a larger performance gap, benefiting more from normal voices but struggling with dysphonic voices. Vowel recognition also varied by talker and vowel, with certain dysphonic talkers and vowels (e.g., "hawd," "whod) causing greater deficits. Additional analyses will examine voice quality features and formant cues to determine their impact on children's vowel recognition. Findings suggest that degraded voice quality can compound the challenges children face when understanding speech in noisy environments.

3:00–3:20 Break

Contributed Papers

3:20

2pAAa7. Acoustic design in contemporary museums: Balancing architectural aesthetics and auditory experience. Milena J. Bem (School of Architecture, Rensselaer Polytechnic Inst., 1605 Hutton St. Apt 10, Troy, NY 12180, jonasm@rpi.edu), Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), and Samuel Chabot (Rensselaer Polytechnic Inst., Troy, NY)

Despite the critical role of auditory environments, contemporary museum designs often prioritize visual aesthetics—characterized by highly reflective surfaces, high-volume rooms, and open-plan layouts—resulting in acoustic challenges such as excessive background noise, high reverberation, poor speech intelligibility, and lack of privacy. In this study, room impulse responses were measured in prominent contemporary museums, including the Massachusetts Museum of Contemporary Art (MASS MoCA), and used to calibrate acoustic simulations. These simulations informed the development of proposed scenarios aimed at enhancing the acoustic quality of these spaces. Key parameters analyzed include the speech transmission index (STI), distraction distance (r_D), clarity (C_{50}), reverberation time (RT), and background noise level ($L_{p,A,B}$). The findings underscore the critical role of integrating acoustics into architectural design, providing architects and acousticians with practical strategies to balance auditory and visual elements—ultimately creating museum environments that enhance both speech communication and immersive auditory experiences.

3:40

2pAAa8. Investigation of the appropriate sound level of masking sounds with different sound sources in open-plan offices. Chan Hoon Haan (Architectural Eng., Chungbuk National Univ., Chungdaero 1, Seowon-gu, Cheongju, Chungbuk 28644, Korea, chhaan@cbnu.ac.kr)

Sound masking is widely used in open-plan offices for speech privacy. The present work aims to investigate the appropriate sound level of masking sounds in open-plan offices with different sounds. Three sound sources were used including white, pink, and brown noises. Listening tests were undertaken on 30 adults with questionnaire surveys. Speeches recorded in an anechoic room were played at 65 dB to subjects with masking sounds of different sound levels using a ceiling speaker. Five different sound levels were used for each masking sound such as 43, 46, 49, 52, and 55 dB. Subjects were asked to answer three questions about the degree of speech comprehension, annoyance and pleasantness. Results were analyzed using a Python program. As a result, it was found that white noise was evaluated as the most unconformable sound in terms of annoyance and pleasantness. It was also revealed that brown noise is the most appropriate sound source for masking with a sound level of 49 dB (SNR 16). Also, pleasantness was high with the masking sound level of 46 dB (SNR 19).

Invited Papers

4:00

2pAAa9. In-progress updates to ANSI S12.60 Part 1 pertaining to sound isolation. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, Ste. 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

In early 2022, work began to update ANSI/ASA S12.60, Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools, Part 1: Permanent Schools. The current version of this standard is from 2010. When the working group was polled for comments and requested changes for a new version of the standard, concerns regarding how the 2010 version handles building envelope sound isolation requirements were numerous. Ensuing discussions also revealed concerns with interior sound isolation language. The current progress of the working group's actions regarding modifications to the standard's sound isolation language will be discussed.

4:20

2pAAa10. A case study of biophilic sound masking. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, Ste. 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

Atkinson Hall is a new multi-disciplinary academic research building on the campus of Cornell University in Ithaca, NY. Electronic sound masking was incorporated into the building concept from the onset of the design, and early in the design process it was suggested that a "biophilic" masking signal rather than the more common filtered broadband noise be considered. Ostergaard Acoustical Associates developed a design incorporating optional biophilic masking so that different types of masking are possible in different areas of the building. This presentation will discuss the concept of biophilic masking and its potential benefits to building occupants, the process of introducing this concept to the design team and University representatives, the methodology for developing an appropriate biophilic signal, and the technical challenges of implementing this signal effectively throughout the completed building.

4:40

2pAAa11. Speech privacy worksheets, time-tested but mostly forgotten. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Zachery O. L'Italien (McKay Conant Hoover, Westlake Village, CA)

Providing speech privacy is an important goal in architectural acoustics. Studies beginning in the 1960s showed that concentrating only on sound isolation was not enough; speech effort, privacy expectations, and background noise levels were also essential considerations. A method was developed to help guide designers toward reasonably reliable results and simplified to accommodate non-technical input from laypersons, resulting in speech privacy worksheets. This led to further refinements and developments, such as for open-plan spaces, and by extension can help guide designs for non-speech sounds such as for music or equipment.

5:00

2pAAa12. Architectural acoustics standard and expectations for speech privacy in the built enviroment. Kenneth Good (Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrongceilings.com)

This paper will serve to explore the different architectural conditions, expectations, and standards related to speech privacy. What processes to use under what conditions and how to better communicate the experience.

TUESDAY AFTERNOON, 20 MAY 2025 GALERIE 3, 1:00 P.M. TO 5:40 P.M.

Session 2pAAb

Architectural Acoustics, Noise and ASA Committee on Standards: Day of ASHRAE Part III—Research, Education, Certification, and Remediation

Erik Miller-Klein, Cochair
Tenor Engineering Group, 11514 Dayton Avenue N, Seattle, WA 98133

Samuel H. Underwood, Cochair
Durham School of Architectural Engineering and Construction, University of Nebraska-Lincoln, 1110 S. 67th Street, Omaha, NE 68182-0816

Joseph F. Bridger, Cochair
Stewart Acoustical Consultants, 7330 Chapel Hill Road, Suite 201, Raleigh, NC 27607

Kevin Herreman, Cochair
Owens Corning, 2790 Columbus Road, B75, Granville, OH 43023

Invited Papers

1:00

2pAAb1. A history of ASHRAE-sponsored acoustics research. Karl Peterman (3 Keensford Court, Unit 1, Ajax, ON L1Z 0K4, Canada, kpeterman@vibro-acoustics.com)

The American Society of Heating Refrigerating and Air-conditioning Engineers has had a long history of supporting research into all aspects of building systems and their effects on occupants and the environment. This effort started in 1959 and just a few years later, in 1965, the industry benefitted from Uno Ingard's work titled Noise Generation in Ducts, the first of many projects that looked at acoustic effects found in heating, ventilating, and air-conditioning systems. Since that first research report, there have been over two dozen research projects that have contributed to our understanding of HVAC noise. ASHRAE continues to fund research with 37 projects currently in progress worth over $5 million combined including two projects that will add further to our database of knowledge. This presentation will provide a summary overview of the various research projects over the last 60 years and how they have impacted HVAC systems and acoustical design.

1:20

2pAAb2. On ASHRAE RP-1919: Effect of duct size and aspect ratio on flow noise. Raine Stewart (3 Keensford Ct, Unit 1, Ajax, ON L17 0K4, Canada, raine.stewart@swegon.com) and Karl Peterman (Ajax, ON, Canada)

ASHRAE has funded research that has augmented the understanding of sound and vibration in architectural acoustics since the 1960s. Many of the results from these research projects can still be found in handbooks and reference materials for building design. An ongoing research project is Research Project 1919; the goal of RP-1919 is to investigate the regenerated noise and breakout noise of elbow fittings, eventually resulting in an algorithm for a variety of flow rates that considers a duct's size and aspect ratio. This algorithm is intended for more accurate noise prediction of elbow fittings, to better inform building design. The desired outcomes for this project necessitate modified test methods, including variations on several intensity scanning methods, and some unique equipment. Initial

investigations have been undertaken to ensure that the data collection methods produce a reliable representation of the conditions under investigation. This presentation will cover some testing methods, apparatus, and trends from the initial investigation for RP-1919.

1:40

2pAAb3. A summary of two recent ASHRAE research projects in acoustics. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

One of the many great things about ASHRAE is its commitment to research. This presentation will focus on two recently completed research projects sponsored by TC2.6 Sound and Vibration: (1) Annoyance Threshold of Tones in Noise as Related to Building Services Equipment (RP-1707) and (2) Effect of Lining Length on Insertion Loss of Acoustical Duct Liner in Sheetmetal Ductwork (RP-1408). In addition to a final research report, each with over 300 pages, these two research projects have also created software programs available to ASHRAE members at no cost that provide quick and easy answers to questions relating to the research. These two software programs will be demonstrated as part of the presentation.

2:00

2pAAb4. ASHRAE RP 1852 toward a unified metric for speech privacy in high-performance buildings: Determination of a suitable metric for both open-plan and closed offices through measurement. Roderick Mackenzie (Soft dB, 250 Ave., Dunbar, Ste. 203, Montreal, QC H3P 3E5, Canada, r.mackenzie@softdb.com), Rewan Toubar, and Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., Montreal, QC, Canada)

Various metrics have been developed since the 1960s to measure speech privacy (SP) within office spaces, but have diverged between those intended for open-plan (e.g., Articulation Index, ASTM E1130 or Speech Transmission Index, ISO 3382-3) or those for closed offices/rooms (e.g., Speech Privacy Potential, or Speech Privacy Class, SPC, ASTM E2638). Others (e.g., Speech intelligibility Index, ANSI S3.5) do not have a standardized use in offices. No single metric has been validated for both office types. Furthermore, SII, AI, and STI struggle to differentiate between unintelligibility and inaudibility, while SPC's relationship to distraction and use in open plan is underexplored. This ASHRAE-funded research project (RP 1852) aimed to determine and validate a single metric and method for both space types to accurately rate or predict SP (in terms of audibility, intelligibility, and distraction). 308 open-plan workstations and 44 closed rooms (private offices and meeting rooms) across 5 sites were assessed using the aforementioned metrics and associated methods and variants thereof. Statistical analysis of 1312 open-plan and 1532 closed room measurements reveals the relations between SP metrics. Effects of source/receiver locations and room boundaries are described. The strengths and weaknesses are discussed, with ultimately the SPC being the strongest candidate for a unifying metric across spaces.

2:20

2pAAb5. ASHRAE RP 1852 Toward a unified metric for speech privacy in high-performance buildings: Subjective survey results. Joonhee Lee (Dept. of Building, Civil and Environ. Eng. Concordia Univ., Montreal, QC, Canada, joonhee.lee@concordia.ca), Rewan Toubar (Dept. of Building, Civil and Environ. Eng. Concordia Univ., Montreal, QC, Canada), and Roderick Mackenzie (Soft dB, Montreal, QC, Canada)

Speech privacy in office environments is essential for occupant productivity and well-being. However, practitioners and researchers often rely on different speech privacy metrics for open-plan and closed offices, leading to inconsistent evaluations. This study aims to identify a universal acoustic metric that accurately reflects subjective speech privacy responses regardless of office configuration. A total of 82 participants from three office sites in Quebec, Canada were surveyed regarding perceived speech audibility, intelligibility, distraction, and expectations for their open-plan workstations, private offices, and meeting rooms. Acoustic measurements were then used to compute Speech Privacy Class (SPC), Privacy Index (PI), and Speech Privacy Potential (SPP) between the same locations. Among the metrics evaluated, SPC and PI calculated from "most-likely-receiver" locations showed the strongest correlations with subjective responses. Thresholds for descriptive terms for "Normal" and "Confidential" speech privacy were generally validated for both PI and SPC. The findings also suggest that SPC could serve as a promising candidate for a unified calculation metric of speech privacy in both open and closed office environments.

2:40

2pAAb6. ASHRAE handbook: The future of vibration isolation. Erik Miller-Klein (Tenor Eng. Group, 11514 Dayton Ave., N, Seattle, WA 98133, erik.mk@tenor-eng.com)

Chapter 49 of the ASHRAE HVAC Applications handbook has been the foundation of vibration isolation recommendations for decades. This document provides a good basis for design and the Technical Committee 2.6 on Sound and Vibration is in the middle of a major update. This seminar will talk about the future of the vibration isolation section as well as the opportunities for research to further enhance the built environment.

3:00–3:20 Break

3:20

2pAAb7. ASHRAE research—How ASHRAE contributes to advancement within the acoustic field through grants for research. Steve Wortman (Eng., Victaulic Co., 4901 Kesslersville Rd., Easton, PA 18040, steve.wortmann@victaulic.com)

ASHRAE currently funds 37 research projects with a value of over $5M. Several ongoing and many past projects are directly related to the field of acoustics. The product of these projects is used to enhance the knowledge base available to practitioners and provide content and clarification for handbooks and other publications. This session will provide an overview of these projects and provide guidance on how to navigate ASHRAE's research approval process.

3:40

2pAAb8. Sound Power Testing—A crash course in acoustics and ratings for the fan industry. Ralph T. Muehleisen (Energy Systems and Infrastructure Anal., Argonne National Lab., 9700 S. Cass Ave., Bldg. 362, Lemont, IL 60439-4801, rmuehleisen@anl.gov)

For several decades the Air Motion and Control Association (AMCA) has run a 3-day, 16-h, short course entitled "Sound Power Testing." The short course was first developed by Joe Pope in the 1990s and modified, updated, and augmented by Ralph Muehleisen from 2007–2023. The course is specifically designed to provide a useful intro/background for fan manufacturing staff who are doing the equipment sound testing and rating as well as for the application engineers who are frequently asked for help and advice related to noise control of their company's fans. Day 1 is an afternoon that focuses on an introduction to sound and acoustics and the concept of the decibel and basic logarithmic/decibel math, frequency analysis, and loudness. The morning of Day 2 focuses on sound level meters, sound in rooms, and sound power testing methods. The afternoon of Day 2 is a tour of the AMCA sound testing facilities with a demo of the reverberation room testing method. Day 3 is a morning that focuses on fan and silencer basics, advanced decibel math, AMCA sound ratings, and noise control basics.

4:00

2pAAb9. Three case studies of HVAC system troubleshooting and remediation. Jennifer R. Miller (Siebein Assoc., Inc., 625 NW 60th St., Gainesville, FL 32607, jmiller@siebeinacoustic.com), Matthew Vetterick, Abigail Gulley, Nicolas Ospina, and Gary W. Siebein (Siebein Assoc., Inc., Gainesville, FL)

Three case studies highlight important considerations in troubleshooting and remediation work on HVAC systems in buildings. Case Study 1 is a historic church with a large organ. Congregants thought the fan coil units under the stained glass windows were too loud. Acoustical diagnostics revealed that the blower for the antiphonal organ was contributing to most of the excessive sound in the space. Case Study 2 is an existing recording studio where a large HVAC unit on the roof was going to be replaced. Acoustical analysis and diagnostic measurements identified both the unit and the duct work as contributing to the high noise levels in the studio. While the client was willing to add silencers and an isolation curb to the unit, they were not willing to address the high velocities and self-generated noise created by the ductwork. Case Study 3 was an operations center for a large hospital that was in an existing shell space in a parking garage where fresh air access for the new air-conditioning system had to be brought into the building from openings in a curtain wall adjacent to a busy street. Mitigation design for the HVAC system included silencers and duct lagging in addition to upgrades to the facade of the building.

4:20

2pAAb10. Case Study—Improving restaurant sound. Kevin Herreman (Owens Corning, 2790 Columbus Rd., B75, Granville, OH 43023, kevin.herreman@owenscorning.com)

It is no secret that high sound levels in restaurants are a familiar experience. There have been many papers and articles have been written about this subject over the last decade. As design trends have moved toward minimalism with clean lines and a blend of style and functionality acoustically absorbing materials like curtains, acoustic ceilings, and carpeting have been deleted and replaced by glass, steel, and concrete Sound reflected back into the space increases reverberation time leading to higher sound levels making communication between restaurant patrons and staff challenging. A large restaurant and entertainment venue, having received several complaints about the sound levels, engaged The Owens Corning Acoustic Research Center to evaluate the space before and after an acoustic treatment was installed. This study presents the changes to the sound levels in the restaurant that adding acoustic treatment to the space has made.

4:40

2pAAb11. Compelling case studies of noise investigation and mitigation in buildings. Joseph F. Bridger (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, joe@sacnc.com) and Mukun Acharya (Stewart Acoust. Consultants, Raleigh, NC)

This paper highlights a selection of intriguing case studies focused on the investigation and mitigation of sound isolation, privacy concerns, HVAC mechanical noise, and plumbing noise. We will delve into the tools and methods utilized, including acoustic cameras, and will present the results of our mitigation efforts where applicable.

5:00

2pAAb12. Exploring the restorative potential of indoor acoustic environments through real-time auralization. Megan Wysocki (Architecture, Virginia Tech, 1325 Perry St., Blacksburg, VA 24061, mwysocki15@vt.edu) and Alaa Algargoosh (Architecture, Virginia Tech, Blacksburg, VA)

Indoor acoustic environments hold significant potential for promoting well-being and cognitive restoration, yet their impact remains underexplored. This study investigates the restorative effects of real-time auralization—a process enabling users to interactively experience a room's acoustic characteristics in virtual or augmented spaces. Using a controlled experimental design, participants' brain activity and performance on attention tests are analyzed to quantify the benefits of this interactive approach. By establishing psychoacoustic metrics and evidence-based design guidelines, this research aims to advance the understanding of how immersive acoustic environments can enhance attention restoration, productivity, and well-being. The findings have broader implications for architectural practices and therapeutic applications, particularly for neurodivergent populations and stress recovery initiatives.

5:20

2pAAb13. Robust 3-D localization of early and late reflections in concert halls using the Sound Field Scanning method. Thomas Rittenschober (Seven Bel GmbH, Hafenstrasse 47-51, Linz 4020, Austria, thomas.rittenschober@seven-bel.com)

Reverberation time measurements form the basis for room acoustic optimizations of existing building structures. During the verification of the achieved room acoustic improvements, anomalies may appear in the reverberation time signal which may be hard to spatially localize, especially in spaces with demanding acoustic requirements such as concert halls or large, open workspaces. This contribution focuses on the application of Sound Field Scanning technology to the fast spatial localization of room reflections. In this process, an omnidirectional sound source is positioned at an observation point in the room and periodically excited with band-limited pulses. An acoustic camera system consisting of a rotating linear microphone array is oriented toward the preferred spatial direction. The emitted pulses and associated room reflections are captured on the measurement surface of the rotating microphone array. Acoustic images with high-depth resolution are generated in parallel planes to the measurement surface. In complex situations, the task of spatially localizing anomalies in the reverberation time signal, may it be early or late reflections, can be reduced to a few measurements from different perspectives, thus, significantly accelerating the problem-solving process with high confidence. The method is exemplarily described through the room acoustic analysis of a concert hall.

TUESDAY AFTERNOON, 20 MAY 2025 GALERIE 4, 1:20 P.M. TO 4:40 P.M.

Session 2pAB

Animal Bioacoustics, Acoustical Oceanography and Signal Processing in Acoustics: Distributed Acoustic Sensing (DAS) in Ocean Acoustics II

Shima Abadi, Cochair
University of Washington, 185 Stevens Way, Paul Allen Center – Room AE100R, Seattle, WA 98195

Léa Bouffaut, Cochair
K. Lisa Yang Center for Conservation Bioacoustics, Cornell University, Cornell Lab of Onithology, 159 Sapsucker Woods Road, Ithaca, NY 14850

Contributed Papers

1:20

2pAB1. Southern Resident Killer Whale calls in Puget Sound analyzed by hydrophone for feasibility with distributed acoustic sensing. Isabelle Brandicourt (School of Oceanogr., Univ. of Washington, 1501 NE Boat St., #206, Seattle, WA 98195, imcbrandicourt@gmail.com), Samantha Juber (School of Oceanogr., Univ. of Washington, Seattle, WA), Scott Veirs (Orcasound, Seattle, WA), Brad P. Lipovsky (Univ. of Washington, Seattle, WA), William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), and Shima Abadi (Univ. of Washington, Seattle, WA)

Pushing distributed acoustic sensing (DAS) methods from typical seismic frequency ranges into higher frequencies (500 Hz to 1 kHz) will greatly advance the tools available for broad spatial scales of marine mammal monitoring. Recordings from an Orcasound hydrophone stationed near a pre-existing fiber optic cable in Puget Sound were analyzed throughout the fall of 2024 for Southern Resident Killer Whale (SRKW) calls. Simultaneously, the cable was illuminated and interrogated to gather DAS data. The acoustic calls were corroborated with citizen science data of visual sightings to filter the calls spatially relevant to the DAS system. The hydrophone data was processed in a method designed to mimic the DAS system specifications. Determining the exact power spectra of each call for specific frequencies (0–1 kHz) provides a much stronger framework for identifying killer whale calls on pre-existing cables with unknown armoring or positioning. This study allows for exploration and expansion of DAS into higher frequencies to monitor SRKW and other marine mammals with limited fieldwork required compared to studies that may require laying a specialized cable. Ultimately, using hydrophone data will accelerate the movement of DAS into the marine mammal observation realm for species with higher-frequency vocalizations. [Work supported by Orcasound.]

1:40

2pAB2. Automatic detection of acoustic signals off Svalbard using distributed acoustic sensing. Josephine N. Schulze (Ctr. for Geophysical Forecasting, Norwegian Univ. for Sci. and Technol., Høgskoleringen 1, Trondheim 7034, Norway, josephine.schulze@ntnu.no), John R. Potter (Ctr. for Geophysical Forecasting, Norwegian Univ. for Sci. and Technol., Trondheim, Norway), and Léa Bouffaut (Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

Distributed acoustic sensing (DAS) is transforming acoustic monitoring, using existing fiber optic infrastructure to create vast listening arrays. DAS provides uniformly spaced virtual channels at intervals of a few meters over hundreds of km, with applications from geophysics and weather to vessel and marine mammal monitoring. A problem arises in that the large number of channels generates immense data volumes that overwhelm manual analysis. Hence, the need for automated approaches, especially for infrequent signals of interest. We develop and evaluate an automated algorithm for generic transient acoustic signal detection in DAS data. Our algorithm exploits the extensive available spatial aperture with multiple channels in the frequency–wavenumber domain to estimate signals against background noise over consecutive time windows. Signal energy estimation metrics (average, maximum, weighted average, and 95th percentile) are compared to a time-varying background noise estimate to derive a custom signal-to-noise ratio used for detection. We evaluate our algorithm performance on whale and ship signals using several months of data from a 130-km cable off Svalbard, taken in 2020 and 2023. The algorithm provides robust support for the creation of labeled DAS datasets that can be used in further processing for automated marine mammal and vessel monitoring at scale.

2:00

2pAB3. Automated association and localization of fin whale calls recorded with distributed acoustic sensing. Quentin Goestchel (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98115, qgoestch@uw.edu), William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), and Shima Abadi (Univ. of Washington, Seattle, WA)

Understanding the behavior and distribution of marine mammals such as fin whales is essential for conservation efforts and population recovery assessments. Distributed acoustic sensing (DAS) offers an innovative and scalable solution by transforming submarine fiber optic cables into dense arrays of acoustic sensors. Over 4 days in November 2021, a public DAS dataset was collected using the Ocean Observatories Initiative Regional Cabled Array off the coast of Oregon, recording tens of thousands of fin whale calls during their breeding season. Previous work demonstrated the feasibility of automated detection using matched filtering, Gabor filtering, and signal-to-noise ratio estimation. Building on these methods, this study aims to obtain a full set of localizations for the experiment. To achieve this, detected calls are automatically associated across DAS channels through a coarse grid search, followed by a least-squares time difference of arrival (TDOA) method to estimate call positions. These techniques are tested on representative subsets of the dataset, providing a foundation for long-term studies on fin whales using DAS. The resulting catalog of localizations and tracks can address questions such as fin whale distribution across the shelf and upper continental slope, net movement direction, and spatial organization of calling individuals. [Work supported by ONR.]

2:20

2pAB4. Simultaneous tracking of a vessel and marine mammal from a distributed acoustic sensing enabled telecommunications fiber: A case study. Calder L. Robinson (Electron. Systems, NTNU, Ctr. for Geophysical Forecasting, Gamle Elektro 2, Gloshaugen, Trondheim 7034, Norway, calder.robinson@ntnu.no), Robin Andre Rørstadbotnen (Electron. Systems, NTNU, Trondheim, Norway), Ana Sirovic (Norwegian Univ. of Sci. and Technol. (NTNU), Trondheim, Norway), and Martin Landrø (Electron. Systems, NTNU, Trondheim, Norway)

Distributed acoustic sensing (DAS) provides opportunities for long-term and large-scale monitoring of underwater environments. These data are of ecological significance and can contribute to behavioral assessments of marine mammals, reducing negative interactions with shipping due to exposure, disturbance or impact. This case study demonstrates the simultaneous tracking of a marine mammal and a vessel using DAS-enabled telecommunications cables deployed in a coastal inlet of the Svalbard Archipelago, over several hours. The system captured the distinct acoustic signatures of the whale's vocalizations and vessel noise attributed to the engine and cavitation, with marine mammal scale spatial resolution and high temporal fidelity. Signal processing techniques including adaptive DFT beamforming were applied to isolate and classify these sources, revealing detailed movement patterns, which may in the future be used as a proxy for behavior. The results highlight the DAS system's ability to contribute to behavioral ecology studies, maritime traffic management, and mitigation of anthropogenic impacts on marine life in an environment. Potential challenges to future real-time implementation are identified and discussed.

2:40–3:00 Break

Invited Papers

3:00

2pAB5. Denoising Distributed Acoustic Sensing (DAS) for shallow water acoustic observations. Wenbo Wu (Woods Hole Oceanogr. Inst., Woods Hole, MA, wenbo.wu@whoi.edu), Xuancheng Huang (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Ying-Tsong Lin (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Distributed Acoustic Sensing (DAS) offers a cost-effective solution for long-term acoustic monitoring in shallow water environments. However, field data from multiple DAS experiments reveal significant noise, particularly at frequencies above 1 Hz, which challenges its use for acoustic studies. Notably, these high-frequency noises (>1 Hz) are strongly correlated with 0.1–0.5 Hz ocean gravity waves, with noise levels increasing at the peaks and troughs of the gravity waves' amplitudes. To address this issue, we developed a curvelet-based machine learning method to remove noise and enhance the signal-to-noise ratio (SNR). Using 2 years of DAS data collected at the Martha's Vineyard Coastal Observatory, we trained and validated the denoising model. The method was applied to signals from wind farm pile-driving and whale calls, resulting in significant SNR improvement for both cases. The denoised results were benchmarked against data from a co-located hydrophone and a hydrophone array deployed at the wind farm site. Comparisons demonstrated strong agreement between the hydrophone and denoised DAS data. This highlights the potential of the denoising approach to uncover signals masked by noise in DAS data and enable the detection of signals previously hidden in noisy data, significantly enhancing the utility of DAS for shallow-water acoustic studies.

3:20

2pAB6. Observing coastal ocean and ice physics with seafloor distributed acoustic sensing. Maddie Smith (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #12, Woods Hole, MA 02543, madisonmsmith@whoi.edu), Jim Thomson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Christian A. Stanciu, Robert E. Abbott, Michael G. Baker (Sandia National Labs., Albuquerque, NM), and Jacob Davis (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

When distributed acoustic sensing (DAS) is applied to seafloor fiber optic cables, the observations of cable strain record oceanographic signals including acoustic waveforms in the water column and pressure variations from ocean waves. In comparison to other traditional sensors, such as seafloor moorings with pressure sensors and hydrophones, DAS provides much higher spatial resolution — up to 2-m spacing on 10s of kilometers of cable — with the trade-off of reduced signal-to-noise. Here, we particularly focus on what we

can learn about ocean physics in seasonally sea ice-covered coastal oceans with seafloor DAS. Experiments have been completed on a variety of cable types and deployments ranging from seafloor telecommunication cables and scientific communication cables to purpose-deployed lightweight cables. Ocean surface waves can be observed in high resolution in a phase-resolved manner and converted to bulk statistics with the use of *in situ* calibration. In the presence of sea ice, the attenuation rate of surface waves can also be derived to provide additional insight into ice distribution and concentration. More bespoke deployments of fiber optic cables, such as on top of sea ice and vertically through the water column, can provide more targeted information on environments and processes of interest.

3:40

2pAB7. Distributed acoustic sensing instrumentation for oceanic sonic acquisition. Arthur H. Hartog (FOSINA, 23 rue du Port, Nanterre, Hauts de Seine 92000, France, arthur.hartog@fosina.fr)

The paper explores the challenges of the instrumentation created by the need for designing and operating DAS in the ocean. The objectives of very long range, fine spatial resolution, low measurement self-noise and broad acoustic bandwidth all work against each other. Broadly, these metrics define the boundaries of the system's performance. Moreover, the current trend to use fibers on existing infrastructure telecommunications or energy cables (spare fibers, or disused cables) limits the options for improving the performance. We discuss some of the recent achievements in the technology and options for taking it further. This includes pre-existing cables, but new dedicated cables offer much wider opportunities for extending the range, including dedicated interrogation for individual sections, the use of optical amplification that is optimized for DAS and advanced probe signal synthesis. A further area of exploration includes the miniaturization of the interrogation to fit in autonomous underwater vehicles or in stationary nodes. The placement of stationary nodes is particularly relevant to early warning of devastating events such as earthquakes and tsunamis.

4:00–4:40 Panel Discussion

TUESDAY AFTERNOON, 20 MAY 2025 — STUDIOS 7/8, 1:00 P.M. TO 5:00 P.M.

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Acoustical Oceanography at Deep Water Abrupt Topography III

John A. Colosi, Cochair
Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943

Ying-Tsong Lin, Cochair
Woods Hole Oceanographic Institution, Scripps Institution of Oceanography, La Jolla, CA 92093

Lauren Freeman, Cochair
NUWC Newport, Naval Undersea Warfare Center (NUWC), 1176 Howell Street, Newport, RI 02841

Contributed Papers

1:00

2pAO1. Machine learning-enabled retrieval of ocean current speeds from acoustic flow noise in the 2023 NESMA Pilot Experiment. Tsuwei Tan (Phys. Dept., Naval Postgrad. School, 1 University Cir, Monterey, CA 93943, tsuwei.tan1.tw@nps.edu), Oleg A. Godin (Phys., Naval Postgrad. School, Monterey, CA), Matthew W. Walters (Phys., US Naval Acad., Monterey, CA), John Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA), and Ernst M. Uzhansky (Phys., Naval Postgrad. School, Monterey, CA)

The Naval Postgraduate School deployed a network of Moored Autonomous Noise Recorders (MANRs) in the Northwest Atlantic near the Atlantis II Seamounts during the 2023 NESMA Pilot Experiment. This study examines the retrieval of ocean current speeds from flow noise recorded over a 52-day deployment. A strong correlation was observed between acoustic noise intensity at infrasonic frequencies (<20 Hz) and current speeds. Distinct spectral properties enabled differentiation between flow noise and ambient sound, including shipping noise. A regression tree machine-learning model trained using data from MANR #1, equipped with both a hydrophone and a current meter, facilitated the inference of current speeds with 1-min resolution at MANR #2, which was equipped solely with a hydrophone. MANRs #1 and #2 were located on steep seamount flanks at depths of 2573 and 2994 m, respectively. The regression tree model estimated current speeds of up to 107.7 cm/s at MANR #2, which was validated by spectral comparisons of flow noise at the two moorings. These findings underscore the potential for hydrophones to function as effective tools for long-term current monitoring, offering critical insights into deep-sea currents and their impact on seafloor dynamics and sediment transport. [Work supported by ONR.]

1:20

2pAO2. Ray trace modeling of three-dimensional propagation at the New England Seamounts. Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, thomas.jerome@arlut.utexas.edu), Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Julien Bonnel (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Out-of-plane propagation resulting from acoustic interactions with abruptly varying bathymetry is often difficult to predict and can make the disentanglement of noise and reverberation from multipath propagation challenging. As a case study of these effects, an airgun survey conducted during the New England Seamounts Acoustics Experiment is analyzed for the presence of out-of-plane arrivals associated with prominent bathymetric features of the Atlantis II Seamount Complex. Time delay analysis is applied to a set of three near-bottom acoustic recorders to assess the directionality of distinct arrivals in the received time series. Back propagation using three-dimensional (3-D) ray tracing facilitates the identification of bathymetric features associated with the arrivals. Predicted propagation paths are recreated by combining two-dimensional ray traces from the source to the bathymetric feature and from the bathymetric feature to the receiver to obtain modeled arrival times for comparison with observations. Multiple examples of out-of-plane reflections from specific bathymetric features on the slope and near the base of the seamount are identified, providing insight into likely contributors to the complex propagation in seamount environments. Forward modeling of the identified propagation paths using 3-D ray traces highlights challenges for ray-based acoustic modeling of propagation in the presence of highly variable bathymetry. [Work supported by ONR.]

1:40

2pAO3. Extension of measured sound speed profiles to the upper ocean using altimetry-informed gravest empirical modes. WIlliam R. Harris (AOSE, MIT-WHOI Joint Program, 77 Massachusetts Ave., Cambridge, MA 02139, harriswr@mit.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., La Jolla, CA), William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA), and Magdalena Andres (Physical Oceanogr., Woods Hole Oceanogr. Inst., Woods Hole, MA)

During the New England Seamount Acoustics (NESMA) 2023 experiment, the majority of temperature and salinity measurements from fixed moorings did not reach the surface due to a mooring blowdown caused by the Gulf Stream. To provide a statistical model of upper ocean properties, NESMA 2023 mooring measurements are extended to the surface using the altimetry-informed gravest empirical mode. The altimetry-informed gravest empirical mode is a tool that allows for the determination of interior, three-dimensional water column profiles from surface measurements of sea surface height, as well as the depth-dependent uncertainties of these properties. Sound propagation models based on the parabolic equation method will be implemented using the augmented sound speed profiles. Outputs from the acoustic models will be compared with sound propagation measurements to evaluate the accuracy of the sound speed profile augmentation. [Work supported by the Office of Naval Research.]

2:00

2pAO4. Measured bottom-topography effects on mid-frequency acoustic propagation. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375, altan.turgut@nrl.navy.mil) and Jeffrey Schindall (Naval Res. Lab., Washington, DC)

Abrupt bottom topography near a shelf break has direct and indirect effects on 2-D and 3-D acoustic propagations. To measure the effects of indirect (oceanographic) processes, two Vertical Line Arrays and one source moorings were deployed near the shelf break in the Western Barents Sea from October 12–19, 2022. The source mooring transmitted several Linear Frequency Modulated (LFM) and Continuous Wave (CW) signals, covering a frequency band of 0.7–4.2 kHz. Travel-time analysis of the cross-shelf acoustic data showed larger semidiurnal travel-time fluctuations as compared to those of the along-shelf acoustic data. This was explained by tidal effects on the Polar front and verified by acoustic simulations using both measured oceanographic data and 3-D HYCOM data. To measure the direct effects of abrupt topography, a Vertical/Horizontal Line Array mooring was deployed in the DeSoto Canyon from April 24–29, 2023. A source was towed along the shelf break at 40 m depth transmitting 350–950 Hz LFM and 361 Hz CW signals. Measured effects of abrupt topography such as sudden appearance of LFM signals at a 50-km range, earlier arrival of bottom-bounced signals, etc. can be explained by 2-D acoustic propagation. Topographic effects of 3-D acoustic propagation in the experimental area are also discussed using measured and simulated data. [Work supported by the ONR.]

2:20

2pAO5. Shaping the acoustic field in the Gulf of Mexico: Marine mammals linked to topography and oceanographic features. Alba Solsona Berga (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., San Diego, CA 92093, asolsonaberga@ucsd.edu), Lance Garrison (NOAA NMFS Southeast Fisheries Sci. Ctr., Miami, FL), Matthieu Le Henaff (NOAA Atlantic Oceanographic and Meteorological Lab. (AOML), Miami, FL), Melissa Soldevilla (NOAA NMFS Southeast Fisheries Sci. Ctr., Miami, FL), and Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Marine mammals shape underwater soundscapes through their vocalizations in ways influenced by their habitat use and behavior. Understanding the drivers of their movements and behaviors within the Gulf of Mexico (GoMex) can enhance our ability to predict local acoustic field variability in response to mesoscale oceanographic processes. To investigate these drivers, we derived daily species density estimates from an 18-station gulf-wide passive acoustic array from 2020–2023. We investigated the spatiotemporal distribution of sperm and beaked whales in relation to topography and oceanographic processes, including depth, slope, sea surface temperature, salinity, chlorophyll-a, upwelling, eddy dynamics (surface and depth), and Loop Current influence. We applied Boosted Regression Trees to learn and predict spatial distributions of these deep divers across the GoMex, finding that goose-beaked whales are associated with deep eddies near steep slopes, Gervais' beaked whales follow surface and midwater eddies, and sperm whales frequent freshwater-influenced regions, avoiding Loop Current waters. These findings highlight how dynamic physical oceanographic conditions at the surface and depth interact with topography to drive cetacean occurrence. Mechanisms may include prey aggregation and compression. Marine mammals can provide indirect acoustic cues to help us understand deep processes tied to specific habitat features, and predict sound field variability.

2:40

2pAO6. Episodic deep-water currents observed in NESMA and their implication on acoustics near the New England Seamounts. John Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA, jejospeh@nps.edu), Oleg A. Godin, Tsuwei Tan (Phys., Naval Postgrad. School, Monterey, CA), Matthew W. Walters (Phys., U.S. Naval Acad., Monterey, CA), and Ernst M. Uzhansky (Phys. Dept., Naval Postgrad. School, Monterey, CA)

Naval Postgraduate School deployed a network of Moored Autonomous Noise Recorders (MANRs) in the vicinity of the Atlantis II Seamounts for 2 months during the 2023 New England Seamounts Acoustics (NESMA) pilot experiment and for 5 months during the 2024 NESMA field experiment. In each experiment, two MANRs were placed on the steep flanks of the Atlantis II Seamounts and two were placed along a deep trench. Strong, episodic near-bottom flows were observed a few meters above the seafloor during both experiments at water depths between 2550 and 4450 m. An assemblage of measurements from tilt current meters, flow noise over the MANR hydrophones and variations in *in situ* temperature were used to identify the unexpected flows. In this work, we examine the characteristics of the irregular flows, their spatial and temporal relationship with the Gulf Stream, and the implications of the flows for sound propagation and acoustic measurements in this region of complex bathymetry and oceanography. [Work supported by ONR.]

3:00–3:20 Break

3:20

2pAO7. Effects of airgun operations on targets in the deep scattering layer. Camille Wardlaw (AOPE, Woods Hole Oceanogr. Inst., 98 Water St., MS #9, Woods Hole, MA 02543, camille.wardlaw@whoi.edu), Julien Bonnel, and Andone C. Lavery (AOPE, Woods Hole Oceanogr. Inst., Woods Hole, MA)

Seismic surveys, commonly used for oil and gas exploration, rely on airgun operations that produce intense, low-frequency acoustic impulse signals. These signals have the potential to disrupt ocean ecosystems by altering targets' spatial distribution and triggering behavioral changes. To better understand these potential impacts, we conducted a joint airgun/REMUS operation focusing on biological targets in the deep scattering layer (DSL). A single airgun (210 c.u.) was fired at controlled intervals while the REMUS autonomous underwater vehicle—equipped with a broadband echosounder (38 and 70 kHz), hydrophones, and precise navigational tools—collected data at two specific depths (300 and 550 m) and relatively short slant range (<500 m) from the airgun source. Acoustic target strength (TS) data were recorded to observe potential responses, including changes in individual target behavior at different depths as well as overall layering or patchiness of the DSL. This presentation shares preliminary observations on potential target movement dynamics, startle responses, and vertical or horizontal aggregation shifts due to airgun sounds. [Work supported by the Office of Naval Research.]

3:40

2pAO8. Analysis of frequency-dependent acoustic enhancement from a seamount. Johnathan J. Todd (Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, jjt6090@psu.edu), William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA), Daniel C. Brown, and Chad M. Smith (Penn State, State College, PA)

During the 2023 New England Seamounts Acoustics (NESMA) pilot experiment, continuous measurements were made of underwater acoustic propagation and scattering near a seamount from close range through the first deep-water acoustic convergence zone. Measurements were made by positioning a stationary mid-frequency source at a shallow depth over the plateau of the Atlantis II seamount and then towing Penn State's Three Octave Research Array (THORA) to a range of 85 km from the source. Low- to mid-frequency acoustic data were beamformed to improve signal-to-noise ratio and reduce own-ship noise, and transmission loss was estimated using narrowband processing techniques. Comparison of the data with ray and parabolic equation-based models helped to develop an understanding of the influence of the seamount bathymetry, roughness, and sound speed profile. This talk will discuss details of these unique measurements, modeling, and data analysis confirming a frequency-dependent change in acoustic enhancement within the first deep-water shadow zone.

4:00

2pAO9. Experimental investigation of surface acoustic duct propagation variability near the Atlantis II Seamounts using autonomous surface vehicles. Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., Atlanta, GA), Davis Rider, Matthew McKinley (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Arnaud Le Boyer, Laurent Grare, Matthew Alford, Luc Lenain (Scripps Inst. of Oceanogr., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Detection of underwater targets in the upper ocean is often influenced by surface acoustic ducts, which enable long-range guided sound propagation. The physics of surface acoustic duct propagation, notably its minimum cutoff frequency, has been extensively studied in past studies. However, quantifying the influence of environmental parameters on the temporal and spatial variability of the sonic layer depth (SLD) and the associated energy leakage below the SLD remains under investigation. A collaborative experiment was conducted in August 2024 during the NESMA-IOP 2 cruise to address this topic. A dual-frequency band (∼130 Hz and ∼1.3 kHz) acoustic source was deployed ∼15 m deep from the drifting R/V Revelle while continuously profiling the water column down to 350 m with a fast CTD system to capture the local sub-mesoscale and internal-wave ocean variability. Simultaneously, two autonomous surface vehicles, called Wave Gliders (WG), each instrumented with a towed acoustic module, located above and below the effective sonic layer depth (∼50 m) at ∼12 and ∼100 m, recorded the source transmissions at ranges between 1 and 10 km. A third WG profiled the water column down to 150 m with a CTD assessing the spatial variability of the SLD. Experimental detection ranges are compared to numerical predictions. [Work sponsored by ONR.]

4:20

2pAO10. Effect of bottom topography on sound propagation near the Atlantis II seamount. Pratik Prashant Aghor (Earth and Atmospheric Sci., Georgia Inst. of Technol., 311 Ferst Dr., Atlanta, GA 30332, paghor3@gatech.edu), Matthew McKinley (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Annalisa Bracco (Earth and Atmospheric Sci., Georgia Inst. of Technol., Atlanta, GA)

This work investigates through a modeling exercise how and how much capturing the details of the bathymetry matter for establishing the physical environment in which sound propagates. Two simulations are considered, one with a realistic bathymetry of the New England Seamount region (NESM run) and a second with a single Gaussian seamount of comparable width and height to the Atlantis II seamount (SEAMOUNT run). Both runs are submesoscale permitting (1 km horizontal resolution) with 100 vertical levels. The idealized SEAMOUNT simulation captures a broad range of physical phenomena modeled by the NESM run including a mixed barotropic-baroclinic mechanism responsible for topographic eddy generation, and bottom boundary layer (BBL) separation on the anticyclonic side arising from an overturning instability when the North Atlantic Current flows atop Atlantis II. The scattering of internal waves from the small-scale rough bottom topography and consequent vertical mixing is however underestimated in the idealized SEAMOUNT run. Internal waves can cause attenuation of the propagating sound waves due to acoustic mode coupling. We discuss how dynamical processes like internal waves affect sound propagation near the Atlantis II seamount.

4:40

2pAO11. Water column sound speed estimated from ambient sound measurements recorded on a vertical array of hydrophones. Robert T. Taylor (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, rtaylor119@utexas.edu), Megan Ballard, Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Leah Johnson, Luc Rainville, and Harper Simmons (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In the ice-free Arctic waters near the island of Jan Mayen (Nordic Seas), wind and wave-generated sound is used to estimate the sound speed between vertically separated hydrophones. The experimental data were recorded on a 52-element evenly spaced vertical line array moored on Jan Mayen Ridge, an undersea volcanic ridge with a depth of approximately 420 m at the array location. The coherence between vertically separated hydrophone pairs, sampled at a frequency of 8096 Hz and averaged over the 24-min recording period, provides sufficient resolution in the cross-correlation shape to determine features related to the acoustic travel time between hydrophones. The impact of hydrophone-pair spacing on sound speed estimates is explored, from the smallest separation distance of 7.25 m, determined by array design, to distances exceeding 100 m in high-wind conditions. The sound speed profile was estimated every 4 h over the year-long experiment, and the fluctuations compared well with measurements made from thermistors mounted along the array. The method is robust, enabling accurate estimates (RMSE < 1 m/s) in a wide variety of environmental conditions over the course of the experiment, including low wind conditions, flow noise, and array tilt.

Session 2pBAa

Biomedical Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Double, Double, Toil and Trouble— Towards a Cavitation Dose II

Christy K. Holland, Cochair
Internal Medicine, Division of Cardiovascular Health and Disease, and Biomedical Engineering, University of Cincinnati, Cardiovascular Center, Room 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Eleanor P. Stride, Cochair
University of Oxford, Institute of Biomedical Engineering, Oxford OX3 7DQ, United Kingdom

Invited Paper

1:00

2pBAa1. Using acoustic emissions to monitor and control ultrasound-mediated blood–brain barrier disruption. Nathan J. McDannold (Radiology, Brigham and Women's Hospital, 75 Francis St., Boston, MA 02115, njm@bwh.harvard.edu)

Using ultrasound and preformed microbubbles to disrupt the blood–brain barrier (BBB) is a promising approach to enable the delivery of drugs that do not normally extravasate into the brain. The safe window where BBB disruption is possible without microhemorrhage is small, and due to large variability in skull properties, it is critical to monitor the ultrasound exposures to ensure a safe and effective outcome. Monitoring the acoustic emissions produced during sonication is an effective method to achieve this goal. Due to nonlinear effects, the acoustic emissions produced when circulating microbubbles are introduced are at frequencies different than the driving frequency. At low exposure levels, the microbubbles oscillate stably and produce increases in harmonic emissions. At higher levels, the microbubbles collapse rapidly and violently and produce vascular damage. This inertial cavitation results in a large increase in broadband emissions. These signatures can be used in real time to control the acoustic exposure level and ensure that only stable, and not inertial cavitation, occurs. This talk will be an overview of our experience using acoustic emissions to control the exposure levels during ultrasound-mediated BBB disruption in pre-clinical and clinical studies. Challenges and pitfalls will be discussed, along with potential strategies to mediate them.

Contributed Paper

1:20

2pBAa2. Acoustic emission-based estimation of weakly nonlinear microbubble dynamics. Hohyun Lee (Mech. Eng., Georgia Inst. of Technol., 901 State St NW, Atlanta, GA 30332, hlee649@gatech.edu), Reza Pakdaman Zangabad (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA)

Monitoring microbubble (MB) dynamics is critical during MB-enhanced focused ultrasound (FUS) interventions; our ability to understand distinct mechano-biological effects related to MB dynamics hinges on our ability to accurately estimate the temporal MB radius changes during ultrasonic excitation. Here, we hypothesize that acoustic signatures recorded from real-time passive cavitation detection (PCD) can be used to estimate weakly nonlinear MB dynamics. To test our hypothesis, we employed numerical simulations, based on Rayleigh–Plesset modeling, followed by experimental validation, using calibrated PCDs with concurrent optical imaging of MB dynamics using high frame rate microscopy. Our method, termed linear acoustic wave propagation and superposition (LAWPS) algorithm, is derived using Fourier-series-based linearization of the relationship between MB (which is considered as a monopole source, $R_0 + \Delta R < \lambda$) oscillation and pressure propagation. In our Rayleigh–Plesset-based numerical simulation, LAWPS algorithm estimation closely corroborated the current state-of-the-art non-linear estimation. Crucially, the LAWPS algorithm provided a method to estimate the MB oscillation radius using AE. Moreover, experimental observation using monodisperse MBs and high frame rate microscopy supported the potential of the LAWPS algorithm to accurately estimate the temporal changes in MB radius during 0.5 MHz ultrasonic excitation for small and stable ($\Delta R/R_0 < 30\%$) oscillations.

Invited Paper

1:40

2pBAa3. Inertial cavitation dose applied to two systems: Hemolysis of red blood cells and killing of bacteria cells. Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu), Pratik A. Ambekar (Univ. of Washington, Seattle, WA), Vera A. Khokhlova (Dept. of Medicine, Univ. of Washington, Moscow), and Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Inertial cavitation has been shown to generate pits in hardened metals, so it comes as no surprise that inertial cavitation can cause bioeffects. One might expect that the amount of cavitation would correlate to a bioeffect. The more cavitation there is, or the more intense the cavitation is, the greater the bioeffect. In previous work, hemolysis of red blood cells due to inertial cavitation correlated well with the inertial cavitation dose, or ICD, defined as the cumulated root mean squared broadband noise amplitude in the frequency domain between two harmonics [W. S. Chen *et al*., Ultrasound Med. Biol. 29 (2003)]. However, detection sensitivities and experimental approaches make ICD a relative, rather than absolute, measure of cavitation. Indeed, the cited paper states: "there is no absolute basis by which to specify how much cavitation has occurred. Nonetheless, these dose estimates yield valuable insight into mechanisms." Perhaps even more importantly, ICD, or some other cavitation index, would be valuable to predict the end of treatment. We will examine this via the application of inertial cavitation to two simple *in vitro*- systems, red blood cells and bacteria cells, to illustrate the point. [Work supported in part from NIH grants R01AR080120, R01EB023910, R01GM122859, and R01EB031788.]

Contributed Papers

2:00

2pBAa4. Effect of the point spread function: Deconvolution strategies for accurate spatiotemporal cavitation doses by passive acoustic mapping. Abigail Collins (Inst. of Biomedical Eng., Univ. of Oxford, Botnar Inst. for Musculoskeletal Sci., Old Rd., Headington, Oxford OX3 7LD, United Kingdom, abigail.collins@eng.ox.ac.uk), Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom)

The ability to quantify and compare different cavitational treatments is essential given their accelerating clinical adoption. Passive acoustic mapping (PAM) can reconstruct quantitative maps of cavitational radiated energy density (CRED) in real time, providing a powerful platform for cavitation dosimetry. However, when compared to optical imaging, the relatively longer acoustic wavelengths and longitudinal nature of 2-D PAM result in greater variability and a more significant spatial impact of the point spread function (PSF). The associated imaging artifacts limit the spatial localization of cavitation dose and introduce errors in source energy estimation that vary across source parameters and between setups. Deconvolving with a constant PSF can reduce many of these errors and minimize the setup dependence of CRED estimates, but can introduce iteration-dependent errors in the estimated source location. This work explores the challenges of deconvolving images with large and variable PSFs using algorithms designed for optical applications and proposes strategies for mitigating errors. These approaches will be evaluated with respect to their quantitative performance, computation time, and robustness to noise both in silico and *in vitro*, with the aim of unlocking the full capabilities of PAM as a spatially resolved, energy preserving, setup independent platform for cavitational dosimetry.

2:20

2pBAa5. Practical considerations in the calibration of array probes for quantitative cavitation imaging: Lessons learned from 16 attempts. Darcy M. Dunn-Lawless (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7LD, UK, Botnar Inst. for Musculoskeletal Sci., Windmill Rd., Oxford OX3 7LD, United Kingdom, darcy.dunn-lawless@magd.ox.ac.uk), Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7LD, United Kingdom, Headington, Oxford, Oxfordshire, United Kingdom), and Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7LD, United Kingdom)

Passive Acoustic Mapping (PAM) is a leading approach for the quantification of cavitation energy and can inform the safe and effective use of cavitation in medicine. To achieve quantitative measurements, the sensitivity and directivity of each element in a PAM array must be calibrated, and any issues in the calibration can propagate into poor resolution, incorrect magnitudes, and location errors in all subsequent PAM images. Accurate array calibrations are therefore crucial in the pursuit of a "cavitation dose" to correlate with bioeffects. The most commonly used calibration method, reported by Gray and Coussios in 2018, is a substitution approach using a fine wire scatterer to approximate a point source. Despite its apparent simplicity, this experiment features a number of complicating factors that can prohibit successful measurements, particularly for newer acousticians. Here, we describe the calibration experiment in detail, along with the impacts, causes, and solutions of a range of issues identified through the authors' hard-won experience. Special consideration will be given to the selection of sources and hydrophones, alignment, management of signal-to-noise ratio, and data processing. We will also discuss common assumptions and their validity, measurement uncertainty, comparing results to theory, and future directions for this area of ultrasound metrology.

Invited Papers

2:40

2pBAa6. Chop, grind, puree, liquefy—How to set the right dose on your new histotripsy blender. Jonathan R. Sukovich (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd, Ann Arbor, MI 48109, jsukes@umich.edu), Timothy L. Hall (Univ. of Michigan, Ann Arbor, MI), Scott Haskell, Mahmoud Komaiha, Joseph Lynch, and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Exposure to cavitation during histotripsy incrementally mechanically fractionates targeted tissues and, given sufficient exposure, ultimately reduces them to liquefied acellular homogenates. While one-size-fits-all max exposure treatments can be utilized to ensure complete liquefaction in nearly all cases, recent evidence from animal tumor models suggests this is not ideal and indicates there may be dose-dependent anti-tumor immune responses. The "puree" setting (complete cellular necrosis, but with intact cell nuclei remaining) has

generally been found to produce the strongest immune responses. Monitoring the level of damage generated during histotripsy, particularly quantitatively and in the sub-liquefied regime, has remained a challenge. While ultrasound (US) imaging-based methods have been demonstrated for detecting the presence of histotripsy ablations, and in select cases the binary identification of complete liquefaction, they generally only work for US-accessible and/or for in-plane targets. Here, we describe methods for measuring acoustic shockwaves emitted during cavitation nucleation and collapse using transmit-receive capable histotripsy transducers to assess the lifespans of generated cavitation (and localize it in 3-D), and quantitatively correlate lifespans with damage outcomes generated in *ex vivo* tissues, even in targets inaccessible to US imagers (e.g., through the skull). Preliminary results and challenges experienced in *in vivo* studies are also presented.

3:00–3:20 Break

Contributed Papers

3:20

2pBAa7. Vortex cavitation: Applications of active acoustical scanned array. Chenzhe Wang (Mech. Eng., Univ. of Michigan, G.G. Brown Bldg., 2350 Hayward St., Rm 2651, Ann Arbor, MI 48105, czwang@umich.edu), Ellen Yeats, Mahmoud Komaiha, Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Chengzhi Shi (Univ. of Michigan, Ann Arbor, MI)

Cavitation means rapid transformation of liquid to vapor phase due to a sudden and violent pressure drop. The evolutions of vaporization of the vapor phase have been investigated since the early 19th century. Recently, people have realized that it might be possible to control and utilize cavitation energy. Various approaches have been implemented to trigger cavitation intentionally at specific locations and at specific times. The energy from the collapsed bubble will blast the designated area. The challenge is to provide as much as cavitation energy will keep the pressure low. Here, we introduce acoustic vortex beams in cavitation, demonstrating how they enable the release of more energy while maintaining controlled negative pressure. These acoustic vortices bring in additional in-plane momentum, facilitating bubble evolution. This process leads to the twisting, convergence, and thus the shaping of bubble clouds. Compared to conventional ultrasound-induced cavitation, our vortex-induced cavitation releases significantly more energy into the surroundings through the collapse of gigantic bubble clusters. This makes it a promising technique for enhancing current industrial processing methods and enabling innovative bio-clinical applications.

3:40

2pBAa8. Combination of Gemcitabine with regulated inertial and stable cavitation on a 3-D *in vitro* pancreatic tumor model. Maxime Lafond, Adrien Rohfritsch, Andrew Drainville, Marine Simonneau, Jacqueline Ngo, Magali Perier (LabTAU, INSERM, Lyon, France), and Cyril Lafon (LabTAU, INSERM, Lyon, France, cyril.lafon@inserm.fr)

The prognosis of pancreatic cancer is poor because of resistance to current treatments due to its dense tumoral microenvironment. Cavitation has been proposed to favor the penetration of chemotherapies. In this work, both inertial and stable cavitations were applied to 3-D *in vitro* pancreatic tumor cell model in combination with Gemcitabine. Inertial cavitation aimed at softening the dense fibrous tumors while stable cavitation is aimed at enhancing drug diffusion. Conditions for regulated inertial cavitations are generated with transducers operating at 1.1 MHz, variable PNP up to 7 MPa, 5% DC and 250 Hz PRF. The activity of inertial cavitation was quantified acoustically using a cavitation index (CI) calculated from the average acoustic amplitude of inharmonics frequency bands between 2.475 and 5.775 MHz recorded with a PCD. Chemical dosimetry of cavitation activity was assessed using terephthalic acid by measuring the induced fluorescence. Experiments were performed using two different but geometrically similar systems for which the sensitivity of the PCD was measured. Both systems generated similar cavitation activity when using the CI expressed in Pa. This method of quantifying inertial cavitation may provide a useful metric of comparison between systems in order to allow for direct comparison of experimental conditions.

4:00

2pBAa9. Multiple signal classification in the time domain for passive cavitation imaging. Nathan Caso (Bioengineering, Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, caso.n@northeastern.edu) and Tao Sun (Bioengineering, Northeastern Univ., Boston, MA)

The field of passive acoustic imaging for cavitation control has investigated adaptive and non-adaptive beamforming algorithms to optimize the trade-offs among imaging resolution, artifact reduction, and computational speed. These parameters are critical for clinical translation, where precise control of cavitation location and dosage is essential to ensure the safety and efficacy of focused ultrasound (FUS) therapies while minimizing off-target effects. Conventional beamforming methods, such as Delay-Sum-Integrate (DSI) and Robust Capon Beamformers (RCB), are fast and effective but have limitations including significant artifacts and the need for extensive parameter tuning to achieve optimal performance. This work introduces an adaptation of a Multiple Signal Classification (MUSIC) algorithm to the time domain and evaluates its performance in artifact reduction, resolution enhancement, and practical implementation compared to DSI and RCB. Our MUSIC algorithm achieves a 3.5× improvement in lateral resolution and a 1.3× improvement in axial resolution compared to RCB, while introducing a novel parameter that quantifies the richness of frequency content in the received signal. This parameter enables differentiation between cavitation modalities based on their spectral characteristics. MUSIC is a strong candidate for precise cavitation localization, with scalable artifact reduction, simplicity, and computational speed. Its physically meaningful input parameter further enhances its utility in clinical and research applications.

4:20–5:00 Panel Discussion

Session 2pBAb

Biomedical Acoustics, Computational Acoustics, Signal Processing in Acoustics, Physical Acoustics, and Engineering Acoustics: Technological Developments and Emerging Biomarkers in Elasticity Imaging II

Javier Brum, Cochair
Laboratorio de Acústica Ultrasonora, Instituto de Física, Facultad de Ciencias, Universidad de la República, Igua 4225, Montevideo 11400, Uruguay

John M. Cormack, Cochair
Department of Medicine, University of Pittsburgh, Division of Cardiology, Pittsburgh, PA 15261

Zhiyu Sheng, Cochair
Medicine, University of Pittsburgh, 3550 Terrace Street, Scaife Hall 624, Pittsburgh, PA 15213

Invited Papers

1:20

2pBAb1. Reconstruction of shear moduli in nearly incompressible transversally isotropic media with phase and group velocities of Rayleigh waves. Gabriel Regnault (Université de Lyon, Lyon, France), Agathe Marmin, Matthew O'Donnell, Ruikang K. Wang (Bioengineering, Univ. of Washington, Seattle, WA), and Ivan Pelivanov (Bioengineering, Univ. of Washington, 616 NE Northlake Pl, Benjamin Hall bld, Rm. 363, Seattle, WA 98105, ivanp3@uw.edu)

Evaluating deformational properties in soft tissues is crucial to manage their dysfunctions and help guide surgeries. Wave-based Optical Coherence Elastography (OCE) is a relatively new modality in the elastography field. Although it can probe only anterior organs, such as cornea or skin, it can potentially serve major benefits over USE and MRI due to its non-contact manner, real-time acquisition and high spatial resolution. Additional benefits come from OCT and provide reach, 3-D, micron-scale resolution structural and functional information. Different experimental techniques in air-coupled acoustic micro-tapping OCE have been recently proposed to better match clinical environments. However, OCE has not yet become quantitative in mechanical moduli reconstruction mainly due to (i) wave guidance and (ii) the complicated anisotropic structure of anterior tissues. Mechanical moduli of most anisotropic organs probed by OCE can be locally characterized by a nearly incompressible transversally isotropic (NITI) model. To reconstruct all three mechanical moduli of a NITI medium, angular dispersion of Rayleigh-wave phase velocity should be computed. However, it is not easily achievable due to a difference in propagation directions of wave phase and energy. This talk describes spatial dispersion effects in anisotropic soft media for Rayleigh waves. We consider the main geometries: point-like and line sources, and analytically and experimentally show how to process measured velocity spatial dispersion for correct reconstruction of elastic moduli.

1:40

2pBAb2. Towards characterizing soft tissue viscoelasticity using ultrasound-based harmonic shear wave elastography and physics-informed neural networks. Tuhin Roy (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, tr2738@columbia.edu) and Elisa Konofagou (Columbia Univ., New York, NY)

Toward the characterization of viscoelasticity of the soft tissue, which is an important biomarker, this study aims to investigate the effectiveness of an ultrasound-based novel approach called the Harmonic Shear Wave Elastography (HSWE) framework. We developed and validated a 3-D finite-element simulation framework to model tissue displacement under a multi-frequency HSWE setting. The HSWE results were compared with those from the widely used Pulsed Shear Wave Elastography (PSWE) method through both simulation and phantom experiments simulating breast tumor problems. We focused on analyzing frequency-dependent phase velocity maps within the 300–800 Hz range, examining the effects of inclusion size, stiffness, and viscosity. The HSWE framework demonstrated accurate measurements of group and phase velocities, with mean differences from PSWE of 5.47% and 1.04% in simulations, and 9.72% and 2.8% in phantom experiments. Notably, HSWE excelled in identifying the size and location of smaller inclusions and effectively captured variations in inclusion elasticity and viscosity. Additionally, we applied a physics-informed neural network approach to the HSWE problem as an inverse problem toward the estimation of viscoelasticity of soft tissue, and corresponding results will be presented in the talk, highlighting the potential of these methods in clinical applications.

2:00

2pBAb3. Full shear wave inversion for super-resolution shear wave elasticity. Guoyang Li (College of Eng., Peking Univ., No. 60 Yannan Yuan, Beijing 100871, China, lgy@pku.edu.cn)

Shear wave imaging can obtain images of the shear modulus of media as a form of contrast, and it has widespread applications in medical imaging modalities such as ultrasound imaging, optical coherence tomography, and magnetic resonance imaging. The viscoelastic properties of biological soft tissues have a strong dissipative effect on shear waves, resulting in exponential decay similar to evanescent waves, which are generally measurable only within the near field. Utilizing near-field full shear wave inversion to assess the biomechanical properties of tissues is a key challenge in shear wave imaging. This report mainly introduces our progress in the inversion methods for shear wave imaging. Our inversion algorithm is based on a physics-informed deep neural network, which can efficiently and accurately perform near-field full shear wave inversion. The study reveals the intrinsic super-resolution characteristics of shear wave imaging, indicating that the imaging resolution can be significantly lower than the shear wave wavelength. We further investigated the impact of data signal-to-noise ratio (SNR) on imaging contrast, finding that the SNR significantly affects the imaging contrast, leading to deviations in the identified shear modulus. In summary, this research lays a theoretical foundation for developing super-resolution, high-precision shear wave imaging methods.

Contributed Paper

2:20

2pBAb4. Robotic optical coherence elastography system for *in vivo* 3-D structural imaging and elasticity mapping in skin. Agathe Marmin (Bioengineering, Univ. of Washington, Seattle, WA, agathem@uw.edu), Raveeroj Bawornkitchaikul (Bioengineering, Univ. of Washington, Seattle, WA), Gabriel Regnault (Université de Lyon, Lyon, France), Tam Pham, Russell Ettinger (Surgery, Univ. of Washington, Seattle, WA), Matthew O'Donnell, Ruikang K. Wang, and Ivan Pelivanov (Bioengineering, Univ. of Washington, Seattle, WA)

Evaluating biomechanical properties is crucial to manage skincare and help guide skin cosmetic and reconstruction procedures. Recent development in wave-based optical coherence elastography (OCE) has shown promise for noncontact stiffness measurements in skin and ocular tissues. Integrating multiple OCT modalities—structural and polarization-sensitive OCT, angiography, and optical OCE enables detailed assessment of tissue structure and function at micron-scale resolution. Previous approaches to anisotropic elasticity imaging involved noncontact shear-wave sources and multi-angle acquisitions, but manual rotation limited resolution and increased acquisition time. This study presents a fully automated, fiber-optic robotic system for noncontact monitoring of skin graft surgeries, integrating four OCT modalities. A 1310 nm swept-source OCT system was mounted on a robotic cart with a custom cylindrical acoustic micro-tapping transducer, launching Rayleigh waves in skin through air. The system achieves a 16 μm lateral and 12 μm axial resolution of structural images, with a 9 mm $\times$ 9 mm field of view. The robotic arm provides autofocus, automates imaging head rotation, and acquires waveforms along 19 in-plane angular positions to reconstruct anisotropic mechanical moduli using a nearly incompressible transversely isotropic (NITI) model. *In vivo* measurements on healthy volunteers demonstrated the system's ability to capture structural, functional, and mechanical properties of skin, providing promising results for clinical translation.

Invited Papers

2:40

2pBAb5. Flexural pulse wave detection in coronary arteries using X-ray. Stefan Catheline (INSERM U1032, 151 cours albert thomas, Lyon 69003, France, stefan.catheline@inserm.fr), Sibylle Grégoire, Gabrielle Laloy-Borgna, Bruno Giammarinaro, and Olivier Rouvière (INSERM U1032, Lyon, France)

Dynamic elastography uses an imaging system to visualize the propagation of elastic waves, the speed of which is directly related to the elasticity felt by palpation. Very few studies have focused on X-ray elastography because of the technical challenges it poses: a planar image of an integration volume at a very slow sampling rate. We demonstrate that tracking a slow elastic wave guided along a one-dimensional structure is the solution. The recently discovered flexural pulse wave (Laloy-Borgna, *et al.*, 2023), which is naturally generated by heartbeats and propagates along arteries, is the perfect candidate for X-ray elastography. As it reflects the cardiovascular health of patients, arterial elasticity is a biomarker of high clinical interest.

3:00–3:20 Break

3:20

2pBAb6. 3-D high-frame-rate imaging of natural shear waves in the heart. Annette Caenen (KU Leuven, Herestraat 49 box 911, Leuven 3000, Belgium, annette.caenen@kuleuven.be), Konstantina Papangelopoulou, Laurine Wouters, Jens-Uwe Voigt, and Jan D'hooge (KU Leuven, Leuven, Belgium)

Shear wave elastography images the heart using high-frame-rate ultrasound to detect mechanical waves traveling along the cardiac wall as a result of aortic and mitral valve closure. The propagation speed of these waves depends on myocardial stiffness, rendering shear wave elastography an interesting diagnostic tool for evaluating chamber compliance and diastolic filling properties. Traditionally, shear wave elastography has been studied in a 2-D parasternal view, and it remains unclear how the wave propagates in 3-D compared to the alignment of the 2-D imaging plane. Initial 3-D studies focused on shear wave elastography in the apical view, tracking a different component of tissue motion (longitudinal versus transverse) compared to the parasternal view. Therefore, this work deploys 3-D high-frame-rate imaging in six healthy volunteers (mean age of 31 $\pm$ 4.4 years) using the same echocardiographic view as currently used in 2-D. This allows to better understand the impact of excitation source location and wave propagation trajectories on wave speed estimation in 2-D. We will discuss the wave physics and cardiac mechanics underlying our results, considering recent advancements in the

field. Additionally, we will present preliminary results in older healthy volunteers (>65 years) and patients with hypertrophic cardiomyopathy.

Contributed Papers

3:40

2pBAb7. Finite element characterization of the effect of flow on acoustic radiation force-induced waves in blood vessels. Charles Capron (Biomedical Eng. and Physiol., Mayo Clinic, 200 1st St. SW, Rochester, MN 55902, Capron.Charles@mayo.edu) and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Ultrasound wave-based elastography has emerged as a promising technique for measuring the mechanical properties of blood vessels *in vivo*. It is well known that blood flow modulates the velocity of the symmetric natural pulse wave, but little has been reported about how it may affect the velocity of induced waves using acoustic radiation force (ARF). We used a fluid–structure interaction model in FEBio to study the effect of flow in elastic tubes after applying a simulated ARF. We varied the vessel wall Young's modulus ($E = 200$, 400, and 600 kPa) and wall thickness (WT = 0.5, 1.0, and 1.5 mm) and applied mean flow rates of ±500, 1000, and 1500 mm/s with either parabolic or plug flow using an inviscid fluid model. Analysis of the phase velocities reveals a flow dependence which diminishes with frequency. Plug flow produced greater changes in dispersion curves than parabolic flow. Thinner tubes are more sensitive to flow than thicker tubes. Increasing E tends to increase sensitivity to flow slightly. These results indicate that flow rate may be an important variable to consider when estimating vascular properties from wave-based acquisitions.

4:00

2pBAb8. Comparative assessment of arterial wave velocities measured by Arterial Dispersion Ultrasound Vibrometry (ADUV) and clinical arterial stiffness metrics. Md Aktharuzzaman (Dept. of Radiology, Mayo Clinic, Ultrasound Res. Lab., College of Medicine and Sci., 200 First St SW, Rochester, MN 55905, aktharuzzaman.md@mayo.edu), Charles Capron (Biomedical Eng. and Physiol., Mayo Clinic, Rochester, MN), Tuhin Roy (Civil Eng., North Carolina State Univ., Raleigh, NC), Murthy Guddati (NC State Univ., Raleigh, NC), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Increased arterial stiffness is an early indicator of various cardiovascular diseases (CVD). Arterial Dispersion Ultrasound Vibrometry (ADUV) employs acoustic radiation force to induce arterial wall motion, enabling estimation of arterial wave velocity (i.e., phase velocity). This study investigates the correlation between ADUV-derived wave velocity and traditional arterial stiffness metrics, including distensibility, compliance, and stiffness index in the right and left common carotid arteries (CCAs). In a cohort of 59 subjects, categorized as (a) confirmed CVD ($n = 27$), (b) individuals with cardiovascular risk factors ($n = 20$), and (c) healthy controls ($n = 12$), we assessed arterial wave velocity across the cardiac cycle, split into 10 stages, to capture variations in mechanical properties. Univariate analysis demonstrated a statistically significant inverse linear relationship between ADUV-derived wave velocity and distensibility for the right CCA during the systolic phase (cardiac stages 3–5) and diastolic phase (cardiac stages 7–9) ($-0.35 < \rho < -0.50$, $p < 0.02$). The wave velocity showed associations with compliance in some of these phases. No significant correlations were observed between wave velocity and the stiffness index. Our findings provide insights into the clinical relevance of ADUV-derived wave velocity as a biomarker for arterial stiffness and its potential to complement or enhance traditional stiffness assessments.

Session 2pEA

Engineering Acoustics: Acoustic Transducers and Sensors

Ahmed Allam, Chair
Mechanical & Materials Engineering, University of Cincinnati, 2851 Woodside Drive, Rhodes Hall 500, Cincinnati, OH 45221

Contributed Papers

1:00

2pEA1. Lithium niobate microphone with high SNR potential. Xiaoyu Niu (Elec. and Comput. Eng., The Univ. of Texas at Austin, 2501 Speedway, EER 4.822, Austin, TX 78712, xyniu@utexas.edu), Vakhtang Chulukhadze, Zihuan Liu, Ehsan Vatankhah, Yinan Wang (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Yuqi Meng (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Lezli Matto, Mark S. Goorsky (Material Sci. and Eng., Univ. of California Los Angeles, Los Angeles, CA), Ruochen Lu, and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

We present a lithium niobate (LN) microphone prototype, the first to explore LN as a transducer material for high signal-to-noise ratio (SNR) applications. The design utilizes a bimorph configuration with opposing polarizations, fabricated through a wafer bonding process. The prototype features a 300-μm thick LN diaphragm, which enables lateral electric field sensing from acoustic pressure-induced deformation. Laser Doppler vibrometry and pitch-catch acoustic tests validate diaphragm performance, supported by finite element and lumped element modeling. While initial measurements demonstrate proof-of-concept only, modeling predicts a potential SNR higher than 70 dB for microphones in a 3.76 mm $\times$ 2.95 mm $\times$ 1.1 mm package. Future work will focus on realizing such a microphone. This, in-turn, requires thinning the LN diaphragm to 2 μm to enhance sensitivity, while optimizing electrode design to maximize performance.

1:20

2pEA2. Packaging of directional microphones inspired by the ears of the fly—*Ormia ochracea*. Junpeng Lai (Mech. Eng., Binghamton Univ., Vestal, NY) and Ronald Miles (Mech. Eng. Dept, Binghamton Univ., 85 Murray Hill Rd., Binghamton, NY 13902, miles@binghamton.edu)

Pressure-sensing ears typically incorporate a tympanal membrane that is backed by an enclosed air space so that sound mainly drives only the exterior tympanal surface. Pressure-sensing microphones also typically consist of some sort of pressure-sensing diaphragm structure along with an enclosure that shields the interior surface of the diaphragm from incident sound. The volume of the enclosed air space determines the sensitivity and frequency response and is, perhaps, the first and most important parameter to determine in any microphone design. Microphones that respond to pressure gradients can also be important because of their dependence on the direction of sound propagation. In this case, the proper design of the microphone enclosure also has a marked influence on performance but its design is less well understood. An enclosure design inspired by the coupled ears of the parasitoid flies, Ormia ochracea, and *Emblemasoma* spp. is examined here to determine the dimensions of an air-filled enclosure that facilitates the detection of pressure gradients or acoustic particle velocity. Measured results are in close agreement with those of both a simple analytical model and a more detailed finite element model. It is shown that the resulting design is compatible with silicon microfabrication processes.

1:40

2pEA3. Using speakers as sensors: Detecting acoustic loads with dense neural networks and impedance features. Noori Kim (Purdue Univ., 401 N. Grant St., Rm. 115, West Lafayette, IN 47907, kim4147@purdue.edu), Keisuke A. Nakamura, and Max Chen (Purdue Univ., West Lafayette, IN)

The use of speakers as sensors to detect ear canal conditions has been previously demonstrated using variable syringe lengths attached to earphones. Building on this foundation, our research explores the potential of a single speaker to function as both an actuator and sensor by leveraging electrical impedance measurements under varying acoustic loads, analyzed with dense neural networks (DNN). Electrical impedance data, including magnitude and phase, were collected from four speakers across fourteen distinct acoustic load conditions, yielding a dataset of 5600 samples. The raw data were processed through the DNN model, achieving an 87% accuracy in length prediction independent of speaker types, which improved to 91% when incorporating speaker-specific characteristics. This innovative approach demonstrates the viability of using electrical impedance as machine learning features, paving the way for smart, sensor-integrated systems for ear health monitoring and broader biomedical applications.

2:00

2pEA4. Conformal Cymbal array design. Yishuang Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St., Harbin, Heilongjiang, Harbin 150001, China, 1608899517@qq.com) and Yongjie Sang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

Cymbal transducers are characterized as thin, lightweight, and miniaturized flextensional transducers that can be readily adapted to conform to curved surfaces, thereby facilitating the formation of conformal arrays. These transducers serve as array elements that are strategically arranged on the surface of Unmanned Underwater Vehicles (UUVs), extending from the hemispherical bow to the cylindrical sides. The conformal array is intended for the integrated design of sonar detection and perception systems. This study examines the impact of array layout on the acoustic performance of the array and presents a virtual prototype of a conformal array utilizing Cymbal transducers as its elements. The conformal array is capable of achieving high-performance acoustic emissions in multiple directions, with a source level exceeding 220 dB. Furthermore, it is observed that the grating lobe generated by the wavelength array on the cylindrical side significantly enhances the source level at the front of the conformal array, resulting in an amplification of over 20 dB, while also improving the uniformity of directivity in the circumferential direction.

2:20–2:40 Break

2:40

2pEA5. Calculating the source directivity index for developing hydrophones. Benjamin L. White (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, bw392@byu.edu) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

When designing new acoustic transmitters, an important feature is the source directivity index. Source directivity is often computed using measurements obtained within a water tank environment. The goal of this work is to determine the frequency range over which the water tank in the Underwater Acoustics Lab at Brigham Young University is a suitable environment to determine the directivity index of an acoustic transmitter. A source transducer with a known directivity index is introduced into the tank environment. A receiver with known sensitivities records linear chirps and pulses along a spherical arc. The pulses are timegated based on their position within the tank environment and analyzed in the frequency domain. Overall sound pressure levels are compared with the one-third-octave band levels, and the directivity index is computed. After verifying its feasibility, the procedure is repeated with different transmitters. This work will prepare the tank environment for future studies pertaining to source directivity. [Undergraduate Research supported by the College of Computational, Mathematical, and Physical Sciences at Brigham Young University.]

3:00

2pEA6. Mechanical study of 3-D-printed insect and arachnid bioinspired hair acoustic sensor for separate frequency band acquisition. Samuele Martinelli (Electron. and Elec. Eng., Univ. of Strathclyde, 99 George St., Glasgow G1 1RD, United Kingdom, samuele.martinelli@strath.ac.uk), Andrew Reid (Electron. and Elec. Eng., Univ. of Strathclyde, Glasgow, United Kingdom), and James Windmill (Univ. of Strathclyde, Glasgow, United Kingdom)

Insects and arachnids have mechanoreceptors and similar sensilla which allow a wide variety of sensing mechanisms, from air flow to olfactory stimuli. These structures, often hair-like, vary in shape and size based on their function. The trichoid sensilla is a hair-like sensing mechanism that reacts to low-frequency near-field sound. This project successfully 3-D-printed structures inspired by both insects and arachnids. Of particular inspiration is the flat hair structure of the adult *Buthus occitanus* scorpion, which allows for a larger area of air particles to excite the sensor. The multi-material sensor design mimics trichoid sensilla, with the rigid hair transferring excitation energy to the softer basal area responsible for transduction. Using a Laser Doppler Vibrometer (LDV) it has been possible to show a change in frequency response due to a change in hair structure. Essentially showing a different frequency sensitivity based on hair length, thickness, and shape. The 3-D-printing technique used allows the production of an array of sensors in shorter times. It proves therefore useful for printing sensing arrays that would allow separate frequency band acquisition. Early development to convert mechanical displacement into electric signals has begun. [This work is funded by the Defence Science and Technology Laboratory (DSTL), United Kingdom.]

3:20

2pEA7. Phase characteristic of a Mach–Zehnder interferometric acoustic sensor. Michael S. McBeth (Res. and Appl. Sci., Naval Information Warfare Ctr. Atlantic, NASA Langley Res. Ctr., Hampton, VA 23681, michael.s.mcbeth@navy.mil)

A push–pull Mach–Zehnder interferometric acoustic sensor is composed of two parallel fiber optic line sensor elements separated by one-half acoustic wavelength. The maximum signal response or optical path difference occurs for a plane acoustic wave with an acoustic wave angle of $\pm 90^\circ$ since one line sensor will be in a sound pressure peak while the other is in a sound pressure trough. The minimum signal occurs with an acoustic wave angle of 0° or 180° since both line sensors experience the same sound pressure variations. A phase characteristic of the sensor is obtained by evaluating the optical path difference equation at acoustic wave angles from 0° to 360°. At each acoustic wave angle, the phase of the peak optical path difference is plotted against the acoustic wave angle. The phase characteristic indicates from which side of the line sensors the acoustic wave arrives. Using the magnitude response alone it is not possible to determine which side of the line sensors from where the acoustic wave arrives.

3:40

2pEA8. A cantilever beam-based optomechanical inertial sensor in silicon nitride photonics platform. Yuqi Meng (Chandra Family Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 2501 Speedway, Austin, TX 78712, yuqimeng@utexas.edu), Xiaoyu Niu, Ehsan Vatankhah, Zihuan Liu, and Neal A. Hall (Chandra Family Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Integrated photonics has emerged as a popular platform for performing a wide range of measurements with high sensitivity. Here, an optical accelerometer in a silicon nitride photonics platform was designed, and the fabricated device's performance was measured. The device consists of serpentine waveguide loops on the top of a cantilever beam. Light is first split into a reference and sensing arm. When the cantilever is subjected to acceleration, a strain field is generated within the sensing arm that leads to a proportional optical phase change. Light from the reference and sensing arm is combined, and the output has an intensity modulated by the incoming acceleration. These intensity variations are detected at a fiber-coupled photodetector. Low-frequency phase drift from environmental noise has been suppressed by a negative feedback controller connected to an optical fiber stretcher. Because of the high mechanical compliance of the beam, a low noise floor equal to 10.0 ng/$\sqrt{}$Hz is targeted. The silicon cantilever is thinned to approximately 50 μm, and a bulk silicon proof approximately 500 μm thick resides at the tip. These features are realized with backside etching processes.

4:00

2pEA9. Spider resonance frequency assessment using normal-incidence, transmission loss measurements. Spencer T. Neu (Dept. of Phys. & Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, stneu73@student.byu.edu) and Brian E. Anderson (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT)

The spider in a loudspeaker driver ideally only behaves as a spring, providing a restoring force to bring the loudspeaker diaphragm back to its rest position. In concert with the surround, it also helps ensure that the motion of the diaphragm is unidirectional. Because of the finite area and mass of the spider, an independent resonance of the spider is possible, which serves to decrease the energy radiated from the loudspeaker at the spider's resonance frequency. The goal of this work is to develop a new technique to identify the spider's resonance frequency before the loudspeaker driver is ever installed into a box. A plane-wave tube method to assess the parameters through measurement of the sound transmission through the loudspeaker was developed previously [Leishman and Anderson, J. Acoust. Soc. Am. 134(1) (2013)] and will be utilized in this work. Here, the two-microphone transfer function technique and the two-load method are used to handle non-anechoic terminations to the tube. The assessment of this resonance frequency from transmission loss measurements will be compared to the ability to assess it from electrical impedance measurements.

4:20

2pEA10. Resonance effects of paralleled-pored protective mesh on tweeter frequency response. Reh-Lin Chen (Feng-Chia Univ., No. 100, Wenhua Rd., Xitun Dist., Taichung 407102, Taiwan, rlchen@fcu.edu.tw) and Ting-Yu Wang (Meiloon Ind. Co. Ltd., Taoyuan City, Taiwan)

Tweeter grills are typically designed to protect drivers while maintaining high acoustic transparency, often using a combination of high-porosity stiff structures and soft fabrics to minimize sound attenuation. In contrast, this study investigates a 3-D-printed parallel-pored mesh with sufficient thickness and relatively low porosity to explore resonance effects, primarily Helmholtz resonance. The inertance originates from each pore neck, while compliance arises from the volume between the mesh and the tweeter surface. Numerical simulations using *Comsol* were performed across various geometries, with experimental validation conducted using *Klippel* and

SoundCheck systems. Both approaches demonstrated satisfactory agreement, showing that frequency response can be selectively tuned and enhanced at high frequencies, followed by a sharper roll-off. On-axis measurements were carried out, and *DeltaEC* was used for rapid resonance frequency estimation, though deviations at high frequencies indicate the limitations of one-dimensional assumptions. This study illustrates the potential of tailored 3-D-printed mesh structures to offer additional control over acoustic performance in tweeter applications.

TUESDAY AFTERNOON, 20 MAY 2025 GALERIE 6, 2:00 P.M. TO 4:40 P.M.

Session 2pMUa

Musical Acoustics: Musical Instruments in Jazz II

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, School of Architecture, 110 8th Street, Troy, NY 12180

Murray Campbell, Cochair
School of Physics and Astronomy, University of Edinburgh, James Clerk Maxwell Building, Peter Guthrie Tait Road, Edinburgh, EH9 3FD, United Kingdom

E. K. Ellington Scott, Cochair
Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Invited Papers

2:00

2pMUa1. How trumpet-like is female broadway belting? Ana F. Zuim (MPAP, New York Univ., New York, NY) and Ingo R. Titze (Utah Ctr. for Vocology, Univ. of Utah, 240 S 1500 E, Rm. 206, Salt Lake City, UT 84112, ingo.titze@utah.edu)

Belting is a vocal technique used in musical theater and pop singing. In contrast to classical female singing, it de-emphasizes energy in the fundamental frequency and emphasizes energy in the second to fourth harmonics. The spectrum often resembles that of a trumpet on notes around C5 (523 Hz). This study aimed to investigate spectral trends in five female belters using three different vowels on two slightly different pitches. The trumpet-like spectra are not produced by tuning resonances to harmonics, but rather by distributing supraglottal inertive reactance across the 500–3000 Hz range with a short (17 cm) mini-trumpet airway shape. Vowels produced are highly modified from normal speaking vowels. The results provide helpful information to both singers and vocal pedagogues in aiding belting strategies.

2:20

2pMUa2. An active mute for the trumpet. Miranda Jackson (Music Res., CIRMMT, McGill Univ., Schulich School of Music, Montreal, QC H3A 1E3, Canada, miranda.jackson@mail.mcgill.ca) and Gary Scavone (Music Res., CIRMMT, McGill Univ., Montreal, QC, Canada)

A design of a prototype active mute for the trumpet will be presented. The mute, which has the basic shape of an ordinary mute, contains microphones, an audio driver, and a processor, to modify the impedance of the instrument by feeding back a modified sound signal back through the instrument from the bell. The feedback signal is produced through signal processing techniques, made possible by the speed of the processor, rather than through electronic filters present in previous similar devices intended for the trombone. The intention of the mute is to replicate the behavior of various types of trumpet mutes, with a cancellation mode to allow for silent practice without significantly blocking the airflow. The presentation will include information regarding the basic construction of the mute and the choice of components. The software used to drive the function of the mute will also be outlined. Results of preliminary simulations, as well as of acoustical and playing tests will be discussed, and the feasibility of this device as a mainstream product, as well as possible future enhancements and functionality, will be explored.

2:40

2pMUa3. How cornets and trumpets differ. Arnold Myers (Royal Conservatoire of Scotland, 100 Renfrew St., Glasgow G2 3DB, United Kingdom, a.myers@ed.ac.uk) and Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, Edinburgh, United Kingdom)

The lead brass instrument in many early New Orleans jazz bands was the cornet, but by the 1930s most jazz ensembles featured the trumpet. Louis Armstrong conspicuously made the transition from cornet to trumpet while at the height of his career. Although in general musicians regard the two instruments as being recognizably different, the acoustical behavior of cornets and trumpets is more difficult to distinguish. Most of the features of the two instruments, such as mouthpiece rim and cup, tuning-slide and valve section bores and bell flare, do not differ systematically; only the mouthpiece backbore and instrument leadpipe are necessarily different. This paper draws on input impedance measurements and the analysis of spectra obtained from playing tests recorded under controlled conditions to propose instrument-dependent characteristics of cornet and trumpet sounds.

3:00–3:20 Break

3:20

2pMUa4. A database of anechoic modern-jazz recordings for auralization. E. K. Ellington Scott (Rensselaer Polytechnic Inst., Troy, NY), Lukas Aspoeck (IHTA, RWTH Aachen Univ., Aachen, NRW, Germany), Herbie Klinger (LANI GIRO Recordings, Marl, NRW, Germany), Michael Vorlaender (IHTA, RWTH Aachen Univ., Aachen, Germany), Jillian Willis (Fine and Performing Arts, Iona Univ., New Rochelle, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Anechoic music databases serve as essential tools for the auralization of acoustic environments through the use of measured or simulated impulse responses. Utilizing free-field recordings ensures the absence of early and late reflections, thereby preventing any undesired artifacts that might compromise the authenticity and integrity of the simulated auditory experience. Early public anechoic databases focused on European Classical music initially recorded as stereo and later evolved as individual instrument tracks and multichannel instrument tracks recorded with spherical microphone arrays. While a dataset exists for early Jazz music, analogous databases for modern and contemporary Jazz remain scarce. To bridge this gap, a set of jazz sample tunes was recorded for various instrument configurations from solos to quintets at the hemi-anechoic chamber at the RWTH Aachen University, Institute of Technical Acoustics. The repertoire encompassed Jazz standards, including 12-bar blues form, 32-bar form, and rhythm changes, as well as free improvisation. The recorded ensembles included combinations of voice, soprano saxophone, guitar, bass, and drum set, in trio, quartet, and quintet formations. Unlike conventional anechoic recordings, which typically involve isolated performances accompanied by a virtual conductor or backtrack, these recordings capture simultaneous performances, allowing for real-time communication and improvisation among the musicians.

Contributed Papers

3:40

2pMUa5. A horn model for the study of boundary layer separation at the open end of brass intruments. Carlos Málaga (Física, Facultad de Ciencias, Universidad Nacional Autónoma de México, Av. Universidad 3000, Mexico City 04510, Mexico, cmi.ciencias@ciencias.unam.mx), Aaron Lozano (Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Mexico City, Mexico), Reyna Ramírez (Física, Facultad de Ciencias, Universidad Nacional Autónoma de México, Mexico City, Mexico), Federico Hernández, and Pablo L. Rendón (Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, México City, Mexico)

We present a family of curves that resemble horn profiles of some brass instruments, like trumpets, and use them to experimentally study boundary layer separation and vortex formation in such geometries. The curves are solutions of an ordinary differential equation. Scaling and balance of inertial and centripetal effects provide the basis to derive the equation, and suggest a dimensionless parameter that determines separation. Using particle image velocimetry techniques, vortex formation inside the horns can be measured, allowing for the obtention of the values of the dimensionless parameter from the mentioned balance. [Authors acknowledge partial support from UNAM-PAPIIT-IN117823. A. Lozano acknowledges support from the CONAHCYT scholarship program.]

4:00

2pMUa6. Characterization of acoustic pressure field in a saxophone mouthpiece using fast pressure-sensitive paint. Enis Ukshini (Univ. Antwerp, Groenenborgerlaan 171, Antwerp 2020, Belgium, enis.ukshini@uantwerpen.be) and Joris Dirckx (Univ. Antwerp, Antwerp, Belgium)

Characterizing the acoustic pressure distribution and the air flow in saxophone mouthpieces is valuable input for parameter-driven design improvement. Such improvements can enhance sound quality, playability and instrument performance. In this study, fast pressure-sensitive paint (PSP) is used to experimentally measure the time-varying acoustic pressure field inside a saxophone mouthpiece while the reed vibrates. Simultaneously, the motion of the reed is captured to analyze the phase relationship between the reed's vibration and the acoustic pressure field. The experiments are conducted using a 3-D-printed transparent mouthpiece to allow clear visualization of the pressure distribution. The spatial distribution of acoustic pressure fields inside the mouthpiece will be shown as a function of time. Challenges remain in further improving the sensitivity of the paint to detect finer pressure variations, and to measure superimposed static pressure variations caused by air flow.

4:20

2pMUa7. Experimental study of nonlinear oscillations of a double-reed system. Olinka Ramírez, Federico Hernández (Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Mexico City, Mexico), and Pablo L. Rendón (Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, México City, Mexico, pablo.rendon@icat.unam.mx)

The nonlinear nature of double-reed systems used to induce oscillations in wind instruments such as the oboe and the bassoon has been the

subject of some study, but is still not fully characterized. We have used high-speed photography in a purpose-built chamber to visualize both the transitory regime leading up to a quasi-stationary oscillation and the oscillation of the reeds themselves. In order to explore the relationship between the pressure drop across the reed and the flow through the reed we use tracing particles to analyze the direction of the flow as well as the flow velocity. The amplitude of the opening between the reeds is also examined as a function of pressure. [The authors acknowledge financial support from DGAPA-UNAM and in particular from grant PAPIIT IN117823.]

TUESDAY AFTERNOON, 20 MAY 2025 GALERIE 5, 1:35 P.M. TO 5:00 P.M.

Session 2pNS

Noise, Physical Acoustics and Computational Acoustics: From Boom to Zoom: Department of Defense and Noise II

James M. Potter, Cochair
Department of the Air Force, Department of Defense, 1260 Air Force Way, Arlington, VA 20330

Erica Rohr, Cochair

Kent L. Gee, Cochair
Department of Physics and Astronomy, Brigham Young University, N281 ESC, Provo, UT 84602

Chair's Introduction—1:35

Invited Papers

1:40

2pNS1. The development of Department of Defense aircraft reference noise measurement approaches. J. M. Downing (Blue Ridge Res. and Consulting. LLC, 29 N Market St., Ste. 700, Asheville, NC 28801, micah.downing@blueridgeresearch.com) and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

The key components for Department of Defense aircraft noise impact analyses are the accurate characterization of the noise source and its propagation through the atmosphere. The development of military aircraft noise data collection procedures started in the 1970s and was shaped by the integrated modeling assumptions at that time. The objective of these original data collection procedures was to obtain integrated reference noise levels. The procedures were refined through the 1990s as instrumentation improved. However, at the end of the 1990s, the next generation of military fighter aircraft had new nozzle geometries, vectored thrust capabilities, and more powerful engines. These changes have driven the development of simulation-based acoustical modeling capabilities. The expanded modeling approaches require more detailed characterizations of the aircraft noise source. This presentation focuses on the evolution of data collection procedures to obtain the improved characterization of noise generated by fixed-wing military aircraft, which is mostly documented in ANSI/ASA S12.75 standard. This presentation provides an update to the current data collection protocol based on recent measurements and serves as a discussion for further development in aircraft source noise characterization procedures.

2:00

2pNS2. T-7A Red Hawk beddown noise comparison. Joseph J. Czech (Harris Miller Miller & Hanson Inc., 300 S. Harbor Blvd, Ste. 516, Anaheim, CA 92805-3717, jczech@hmmh.com), Daniel T. Botto (Harris Miller Miller & Hanson Inc., Anaheim, CA), and Paul J. Krusell (Harris Miller Miller & Hanson Inc., Burlington, MA)

The U.S. Department of Defense (DOD) is acquiring the Boeing-Saab T-7A Red Hawk aircraft. The T-7A is a new advanced transonic single-engine jet trainer proposed to replace the aging 1960s-era supersonic twin-engine Northrop T-38 Talon. The DOD is proposing to beddown hundreds of the T-7A across at least five U.S. bases. Drawing from analyses conducted in support of base-specific Environmental Impact Statements spurred by the National Environmental Policy Act, this paper compares the predicted single-event and cumulative noise exposures that may be introduced by the T-7A at three southern Air Force Bases (AFB): Joint Base San Antonio-Randolph, Columbus AFB, and Laughlin AFB. Consistent with DOD guidance, cumulative noise exposure comparisons are presented in terms of acreage and population within specific Day-Night Average Sound Level contours and supplemental noise metrics. This paper also describes key challenges in modeling the T-7A.

2:20

2pNS3. Comparing the predictive capabilities of subjective crackle metrics using leave-one-out cross validation. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov), Maike F. Holthuijzen (Sandia National Labs., Albuquerque, NM), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Metrics that accurately predict the subjective impacts of jet or rocket noise are important to efforts to mitigate the community impacts of such noise. Leave-one-out cross-validation is used to quantitatively compare the ability of linear, logarithmic, logistic, and combined models to predict the perceived crackle associated with unassimilated data points based on measured subject crackle rating data from a prior jury study using measured high-performance aircraft waveforms. The model inputs include one physical metric (the skewness of the derivative of the waveforms) and three sound quality metrics (the standard deviation of the instantaneous sharpness, the standard deviation of the instantaneous loudness, and the skewness of the product of instantaneous loudness and sharpness). Our best combined model was able to predict the values of unassimilated data with a roughly 2-crepit root-mean-square error (on a 50-point category scaling rating task) using a combination of the skewness of the derivative of the pressure waveform and the standard deviation of the instantaneous sharpness as independent variables. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

Contributed Papers

2:40

2pNS4. Acoustical analysis of historical launch vehicle noise from Vandenberg Space Force Base. Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmoats@byu.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), John P. LaBonte (ManTech SRS Technologies, Inc., Lompoc, CA), and Lawrence F. Wolski (ManTech SRS Technologies, Inc., San Diego, CA)

Vandenberg Space Force Base has been launching rockets for different purposes since 1958. Many of these vehicles are retired, making it impossible to collect new launch noise data. However, ManTech SRS Technologies, Inc. has collected recordings from launch vehicles at VSFB for over two decades, including many of these historical vehicles. These include the Titan II, Athena, Delta II, Taurus, TaurusLite, and others. Recordings of these vehicles are useful in investigating the evolution of launch vehicle noise and noise characteristics for different vehicle types. Of the measurements made by ManTech, generally, one location was recorded at each launch, all between 1 and 10 km away from the pad, but some launches feature multiple recording locations. Each recording has been used to calculate the maximum overall sound pressure level, peak frequency, and other metrics. Trajectory estimates potentially allow for sound power calculations for some vehicles, which can be compared to both the available literature and more recent measurements and analysis (e.g., the Falcon 9). [Work supported by USSF through USACE.]

3:00–3:20 Break

3:20

2pNS5. Noise measurements from Starship Flight 5 and the first Super Heavy booster catch. Makayle S. Kellison (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, makayle@byu.edu), Noah L. Pulsipher, Kent L. Gee, Logan T. Mathews, Mark C. Anderson, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

During SpaceX's Starship Super Heavy fifth test launch and first booster catch, far-field noise measurements were taken ranging from 9.7–35.5 km. This paper summarizes the findings of a recent Letter [Gee *et al.*, JASA Exp. Let. 4, 113601 (2024)] and includes an additional spectral analysis. The measurement campaign included eight locations in communities around Starbase, Texas. Four significant acoustical events are noted from the waveforms, including maximum liftoff noise, booster flyback sonic boom, landing maneuver noise, and hot-staging ring sonic boom. Metrics, such as Z and A-weighted maximum levels, sound exposure levels, peak overpressure, and perceived level, were calculated for launch and flyback boom events. Comparing the calculated metrics to predictions made in Environmental Assessment documents reveals discrepancies, particularly for the A-weighted sound exposure level. Finally, Starship launch noise is compared to Space Launch System (SLS) and Falcon 9 at 10 and 20 km using the above metrics and spectra. For these distances, the far-field noise levels produced during Starship Flight 5 are approximately 4–6 times that of SLS and at least 10 times that of Falcon 9. This paper is built upon by N. L. Pulsipher *et al*. in a companion paper describing Flight 6 results.

3:40

2pNS6. Analysis of Starship Super Heavy's acoustics during Flight 6. Noah L. Pulsipher (Phys. and Astronomy, Brigham Young Univ., N 283 Carl F. Eyring Sci. Ctr., Provo, UT 84604, npuls@byu.edu), Makayle S. Kellison (Phys., Rollins College, Provo, UT), Kent L. Gee, Grant W. Hart, Logan T. Mathews, and Mark C. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

SpaceX's Starship Super Heavy, the most powerful rocket ever launched, conducted its sixth test flight on November 19, 2024 from Boca Chica, Texas. Acoustical data were collected from 21 stations ranging from 1 to 35.5 km from the launch pad as a significant update to the eight stations in Flight 5, described in a companion paper by M. S. Kellison *et al*. and published in a recent letter [Gee *et al*., JASA Exp. Let. 4, 113601 (2024)]. In this paper, key results from the Flight 6 analysis are described. First, weather patterns differ from Flight 5 to 6, resulting in reduced sound levels beyond 10 km. Second, one Starship launch is acoustically equivalent to, on average, 2.2 Space Launch System launches and 10.8 Falcon 9 launches. This represents an update from a Flight 5 comparison. Finally, Flights 5 and 6 maximum sound pressure levels are curve-fit and projected onto a Florida map, to estimate sound levels around SpaceX's LC-39A tower at Kennedy Space Center. These findings, which have been submitted to JASA Express Letters, provide additional insights into the Starship noise levels and contribute to a growing understanding of the acoustical environment created by next-generation heavy-lift rockets.

4:00

2pNS7. Tracking a Starship launch using two intensity probes. Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., N357 ESC, Provo, UT 84602, grant_hart@byu.edu), Mark C. Anderson, Kent L. Gee, Makayle S. Kellison, Logan T. Mathews, and Noah Pulsipher (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

On November 19, 2024, Space-X launched the sixth test flight of their Starship rocket. Measurements were made at 21 different locations around the launch pad, including two 3-m radius intensity probes located roughly 2 km north and 2 km south of the pad. There was also an array of measurement stations near the coast ranging from 3 km south of the pad to 27 km north. Using a broadband time-correlation technique, the direction of the sound source can be determined. Having two intensity probes allows the measurement of the trajectory of the rocket as it lifts off. It also allows the determination of the position of the sound source relative to the rocket,

assuming the trajectory is already known. Methods of correcting and compensating for the fact that the intensity probes were on uneven ground will be discussed. Because of a communication problem with the tower, the super-heavy booster did not return to the launch site but was instead diverted to the Gulf of Mexico. There were two transient events that occurred after the launch which were associated with the booster return. The characterization and localization of those events using the intensity probes and other measurement sites will also be discussed.

4:20–5:00 Panel Discussion

TUESDAY AFTERNOON, 20 MAY 2025 BALCONY J, 1:00 P.M. TO 5:20 P.M.

Session 2pPAa

Physical Acoustics, Education in Acoustics and Engineering Acoustics: Celebrating Steven L. Garrett's Fifty Years of Contributions in Acoustics II

David A. Brown, Cochair
ECE, Univ. Massachusetts Dartmouth, 151 Martine Street, Suite 123, Fall River, MA 027230000

Robert W. Smith, Cochair
PSU/ARL, P.O. Box 30, State College, PA 16804

Invited Papers

1:00

2pPAa1. Steven Garrett's contributions as a scientific advisor in the field of acoustophoresis. Bart Lipkens (Pharmaceutical and Administrative Sci., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, blipkens@wne.edu)

Less known to the Acoustics Community is Steven Garrett's contribution as scientific advisor to FloDesign Sonics, a startup company co-founded by the author as a spinout from Western New England University in 2010. The company developed multiple acousto-fluidic platforms for cell processing applications in biopharmacy and cell therapy manufacturing. The platforms were based on piezoelectrical transducer-driven macroscale ultrasonic standing waves, which generate acoustic radiation forces on particles in suspensions. Through an interplay of acoustic radiation forces, fluid drag forces, and gravitational forces, unique cell processing operations were enabled. These processes included cell separation, cell concentration, cell washing, and affinity cell selection at process scale flow rates. As Chief Technology Officer of FloDesign Sonics, the author set up a Scientific Advisory Board tasked to appraise, critique, and support the technology and innovation of the company. Dr. Garrett joined the Scientific Advisory Board from its inception in 2015. Steve was an active and diligent participant. Through our mutual love for physics, acoustics, innovation, and good food, the technical team and Steve developed a warm relationship. His post-meeting reports were detailed with insights that furthered the understanding and development of the technology. This talk reviews Steve's contributions as a Scientific Advisor.

1:20

2pPAa2. Thermoacoustics, sonic gas analyzers, and more of Steven Garrett's research at Penn State University. Robert Smith (Acoust., PSU, State College, PA, bobsmith.atpsu@gmail.com)

Steven Garrett joined The Graduate Program in Acoustics (GPA) at the Pennsylvania State University in 1995, as the United Technologies Professor of Acoustics, and Senior Scientist with the Applied Research Laboratory, and retired from Penn State in 2016. Drawn to many areas of applied acoustics over his 20 years, this talk will (of necessity, briefly!) summarize efforts in four broad research areas that occurred while at Penn State: (1) thermoacoustic refrigeration, continuing and expanding on efforts that had begun at the Naval Post Graduate School; (2) sonic gas analysis which in part arose from the need to track resonances in thermoacoustic refrigerators; (3) nuclear-powered thermoacoustic sensors; and (4) parametric stabilization and excitation efforts which were outgrowths of engineering questions that began with thermoacoustic refrigeration. In addition to many research contributions to Acoustics archived in JASA, these research efforts provide valuable education and training for many acoustical scientists and engineers.

1:40

2pPAa3. Avoiding the wrong result to seven-digit precision. Randall Ali (Inst. of Sound Recording, Music and Media, Univ. of Surrey, Stag Hill, University Campus, Guildford GU2 7XH, United Kingdom, r.ali@surrey.ac.uk)

Professor Steven Garrett's distinctive outlook on acoustics and undaunted pedagogic approach has left a long-lasting impression on students of the Graduate Program in Acoustics at The Pennsylvania State University. From the many adages that would echo throughout the classroom and laboratory such as "a computer could provide the wrong result with seven-digit precision," it was evident that Steve's intent was always on fundamental understanding. While we are fortunate to have this point of view captured and propagated via his recent text aptly titled "Understanding Acoustics," many readers will have to settle for reimagining Steve's dynamic character as he taught the material in person. The in-class "live performances" consisted of a mixture of perilous demonstrations, passionate lectures, illuminating homework problems, and of course, to end it all—an exam-burning party. In this talk, we reflect on the later years of Prof. Garrett's career at Penn State and the contributions he has made to acoustics. Firstly, we will highlight the work of his final research project at Penn State on a thermoacoustic temperature sensor. Following this, we reminisce about his teaching and supervision from the student perspective, and the transformative impact it has had on a future generation of acousticians.

Contributed Papers

2:00

2pPAa4. System noise reduction for chillers. Jin Liu (Sound & Vib., Carrier Global Corp., 11450 Meath Dr., Fairfax, VA 22030, lewjin@yahoo.com)

Transmission loss (TL) is perhaps the most useful metric in the design of noise reduction silencers. However, in chiller systems, the use of TL simply to reduce outlet pulsating pressures is not sufficient for system noise reduction. This is especially true when reflection waves and standing waves build up and increase compressor and pipe noises, which can dominate chiller noise sources. In this study: the transmission loss (TL), the reflection coefficient (RC), and the acoustic specific impedance (Z) are applied for the inlet, outlet, and silencer noise reductions. The viscous Lautrec number (the ratio of viscous penetration depth to the hydraulic radius) is critical in determining the best viscous material choice. For oscillating flow, when the Lautrec number is less than 1, the velocity gradient and shear gradient depend only on the pore geometry. Applying this method, a chiller achieved a reduction in system sound power of 25 dB. The compressor shell's vibrations were reduced to 1/20 and the discharge pulsating pressures to 1/100. Fluid–structure interaction and boundary element method (BEM) were used to model radiation noise, fluid, and structure. The modeling results closely matched experimental data.

2:20

2pPAa5. Collaborating with Steven Garrett to produce an open-source textbook. John LoVerde (Paul S. Veneklasen Res. Foundation, Cypress, CA, johnloverde@gmail.com) and David W. Dong (Paul S. Veneklasen Res. Foundation, Cypress, CA)

The Paul S. Veneklasen Research Foundation (PSVRF), a non-profit dedicated to advancing acoustical research, has had close ties with Steven Garrett through personal relationships with its Board of Directors. The PSVRF provided some funding to several of his projects, notably a wood-burning stove with the thermoacoustic generator. The PSVRF was pleased to contribute to Steven's textbook Understanding Acoustics, and to help make the second edition open source and free to download. The textbook is not only a testament to the strength of the physics foundation of Steven's acoustical work, but to his beliefs in education, access, and sharing his passion for acoustics.

2:40–3:00 Break

Invited Paper

3:00

2pPAa6. Thermoacoustics at the LAUM: The Garrett touch. Pierrick Lotton (LAUM, Av O Messaien, Le Mans, France, pierrick.lotton@univ-lemans.fr), Guillaume Penelet, and Gaëlle Poignand (LAUM, Le Mans, France)

We first met Steve Garrett in December 1994 at the 128th ASA meeting in Austin, Texas. At the time, our team was considering whether to start research into thermo-acoustics at the Laboratoire d'Acoustique de l'Université du Mans (LAUM), France. Steve, with his usual enthusiasm, encouraged us and steered us toward the fundamental publications and numerical tools available then. Since that time, Steve's work in thermoacoustics has largely inspired our own research and the LAUM has found its place in the thermoacoustics community. Our paths have crossed regularly over the years, for example at the 155th ASA meeting in Paris in 2008. In 2016, we had the opportunity to invite Steve to the laboratory for a few months. This first visit was followed by many others and Steve has been an official research fellow at LAUM for several years. Apart from the fact that Steve fell in love with the city of Le Mans, his many stays were the opportunity for more direct and very fruitful collaboration, in particular on the characterization of the limit cycles of thermoacoustic engines and on the dynamics of an electrodynamically driven heated wire. The outcomes of this joint work will be presented.

Contributed Papers

3:20

2pPAa7. Effects of orientation on the temperature distributions inside a thermoacoustic refrigerator's stack. Islam Ramadan (Pprime, Univ. of Poitiers, 6 rue marcel doré, B17, Poitiers 86000, France, islam.ramadan@univ-poitiers.fr), Ernesto Emmanuel Leyva Herrera, and Helene Bailliet (Pprime, Univ. of Poitiers, Poitiers, France)

Non-uniform temperature distributions have been observed in the core of thermoacoustic refrigerators that have large aspect ratios (i.e., transversal dimension/length). In order to investigate the origin of such non-uniformities, a simple thermoacoustic refrigerator (TAR) is developed. The TAR contains a 3-D-printed parallel plate stack equipped with nine thermocouples distributed in both axial and transverse directions. The stack has an aspect ratio of 1.47 and is placed inside a rectangular acoustic resonator at the position (near the closed extremity) where the thermoacoustic effect is maximized (i.e., highest temperature difference across the stack). A loudspeaker is mounted on the other extremity of the resonator. The system is driven at its resonance frequency. The temperature distribution is measured at different acoustic pressure amplitudes for several orientations with respect to gravity. These measurements allow for the study of the effect of natural convection on the temperature distribution. The results show that both orientation and acoustic level influence both axial and transversal temperature distributions within the stack.

3:40

2pPAa8. Acoustic streaming in standing waves at high acoustic levels: A joint experimental, theoretical and numerical analysis. Diana Baltean-Carlès (Institut Jean-le-Rond @'Alembert, Paris Sorbonne Université, Faculté des Sci. et Ingénierie, Paris, France, diana-georgiana.baltean-carles@sorbonne-universite.fr), Virginie Daru (DynFLuid Lab, ENSAM, Paris, France), Catherine Weisman (Institut Jean-le-Rond @'Alembert, Paris, France), and Helene Bailliet (Univ. of Poitiers, Poitiers, France)

We started working on acoustic streaming because it is a source of loss of performance in thermoacoustic devices. It was through this 10-year research work that the Parisian women encountered Steven Garrett and enjoyed all our exchanges. We investigated the physical phenomena responsible for the deformation of Rayleigh streaming patterns in a standing-wave guide at high acoustic levels, focusing on the appearance of reverse streaming cells near the acoustic velocity antinodes observed in both experiments and numerical simulations. Thermal effects were first put aside by considering isentropic acoustic propagation, showing that inertia alone modifies streaming cells at a high acoustic level, but does not create any extra cells since the acoustic field is not modified by the streaming field. When the interaction between streaming and acoustic fields is allowed, small modifications in the acoustic velocity generate extra streaming cells, indicating non-linear interactions. Finally, when taking into account heat transfer within the fluid and the solid walls, transverse mean temperature gradients are generated at high acoustic levels, possibly inducing reverse streaming cells. An analytical criterion for the transition in streaming patterns was established based on the intrinsic coupling between thermal effects and acoustic streaming, and then validated with numerical and experimental results.

4:00

2pPAa9. Experimental and numerical investigation of temperature distributions in a compact thermoacoustic cooler. Catherine Weisman (Institut Jean-le-Rond @'Alembert, Paris Sorbonne Université, Faculté des Sci. et Ingénierie, Paris, France, catherine.weisman@sorbonne-universite.fr), Martin Fontbonne (Pprime, Univ. of Poitiers, Poitiers, France), Yann Fraigneau (LISN, Orsay, France), Islam Ramadan (Pprime, Univ. of Poitiers, Poitiers, France), Diana Baltean-Carlès (Institut Jean-le-Rond @'Alembert, Paris, France), and Helene Bailliet (Pprime, Univ. of Poitiers, Poitiers, France)

This paper presents a measured temperature distribution inside a compact and axisymmetric thermoacoustic refrigerator's core and compares it with numerical calculations. The design of the device benefited from insightful discussions with Steven Garrett and David Gardner, particularly during the design phase. To achieve compactness, the system utilizes two electroacoustic sources around its thermoacoustic core, of which the diameter is 164 mm and the length is 39 mm, capable of supporting a temperature difference of about 70 °C at a working frequency of 47 Hz. Fifteen thermocouples were used to probe the core, revealing significant transverse temperature gradients not predicted by the design tool DeltaEC. To account for these discrepancies, a 2-D numerical model incorporating gravity effects was developed under the low Mach number approximation, and simulations were conducted for a configuration slightly different from the experimental prototype. Calculations were performed for various relative orientations between the device axis and the gravity force to explore the impact of natural convection on the temperature distribution. A hanging system was employed to perform measurements under similar relative orientations, enabling a comparison between experimental data and simulations.

Invited Paper

4:20

2pPAa10. Open mic: A man with one watch knows the time; a man with two is never sure. David A. Brown (ECE/CIE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsdb@gmail.com) and Robert Smith (ARL, Penn State Univ., State College, PA)

This talk is an open invitation to attendees, friends, colleagues and former students to come up and share experiences and stories. Past students are particularly encouraged to share connections and career connections in Acoustics. There will be an informal signup sheet or queue if needed at the meeting. There will also be additional time allocated at the end of the session to share stories and experiences from UCLA, NAVPG-School, Penn State and retirement periods.

Session 2pPAb

Physical Acoustics, Biomedical Acoustics and Engineering Acoustics: Acoustic Manipulations of Objects: Theories and Applications

Joao Ealo, Cochair
School of Mechanical Engineering, Universidad del Valle, Ciudad Universitaria Meléndez, Building E49—Of.2014, Cali 760032, Colombia

Likun Zhang, Cochair
National Center for Physical Acoustics and Department of Physics and Astronomy, University of Mississippi, 145 Hill Dr., University, MS 38655

Chair's Introduction—12:55

Invited Paper

1:00

2pPAb1. Enhancing jet breakup through the manipulation of the fluid interface using fluid-transmitted acoustic waves. Kha Nguyen (Univ. of California San Diego, La Jolla, CA) and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, 9500 Gilman Dr. MC0411, MADLab SME344K, La Jolla, CA 92093, jfriend@ucsd.edu)

Droplet production from fluid jets is useful in many applications, from 3-D printing to fuel combustion in aircraft engines. Here, we incorporate a 7 MHz device producing ultrasound transmitted in the fluid jet to aid its controlled breakup. We demonstrate the ability to significantly reduce the distance to break up using a mechanism not reported before that is substantially different than traditional, low-frequency oscillation-driven instability mechanisms. Instead, here we find the nonuniform acoustic pressure distribution present along the fluid interface produced by the transmitted ultrasound is sufficiently powerful to drive jet breakup. We demonstrate this phenomenon at flow rates and with fluids particularly relevant to fuel injection in aircraft engines.

Contributed Paper

1:20

2pPAb2. High-frequency single-beam acoustic tweezers: radiation pressure versus drag forces. Sarah Vincent (Institut Jean le Rond d'Alembert, Sorbonne Univ., Paris, France), Régis Marchiano (Institut Jean le Rond d'Alembert, Sorbonne Univ., 4, Pl. Jussieu, Paris 75252, France, regis.marchiano@sorbonne-universite.fr), and Jean-Louis Thomas (Institut des Nanosciences de Paris, Sorbonne Univ., Paris, France)

Single-beam acoustic tweezers use focused ultrasound waves to manipulate objects in three dimensions. However, their efficiency is limited by weak axial restoring forces and drag forces generated by acoustic streaming, especially at high frequencies. Indeed, acoustic streaming, a fluid motion induced by ultrasound waves, counteracts the radiation pressure forces responsible for trapping. This study investigates the feasibility of high-frequency acoustic tweezers by analyzing radiation pressure and drag forces in regimes of strong focusing. Using a streaming model and numerical simulations, we analyze the evolution of streaming as a function of acoustic pressure amplitude, beam focusing, and frequency. In particular, we demonstrate that, unlike plane waves where streaming grows quadratically with frequency, for strongly focused beams, streaming becomes less dependent on frequency and eventually stabilizes. This counterintuitive behavior can be explained by the reduction in the size of the acoustic source, which is proportional to the wavelength. A detailed understanding of the evolution of radiation pressure and drag forces enables the identification of trapping regimes to ensure stable trapping, even at high frequencies. These advances pave the way for more reliable devices for the contactless manipulation of micrometric objects, with potential applications in biomedical research and microtechnology.

Invited Papers

1:40

2pPAb3. Stiffness model and object manipulation in near-field acoustic levitation. Yaoke Wang (Mech. Eng., Northwestern Univ., 2145 Sheridan Rd., Evanston, IL 60208, yaokewang2020@u.northwestern.edu) and Ping Guo (Mech. Eng., Northwestern Univ., Evanston, IL)

Near-field acoustic levitation (NFAL) is an emerging technique offering innovative solutions for non-contact manipulation of objects, with applications ranging from precision engineering to autonomous systems. Despite its potential, the theoretical framework of NFAL remains incomplete, particularly concerning its coupled dynamics and nonlinear behaviors. We propose a stiffness model of NFAL that describes the squeezed air film behaviors with effective stiffness coefficients that scale with the air film thickness. The model simplifies the PDE solution into explicit expressions to effectively describe the dynamic behaviors of NFAL systems. Furthermore, by accounting for the real-time vibration of the object, the model demonstrates that the stiffness-mass system resonates at a specific frequency, marked by a rapid phase shift between the two interacting surfaces. This framework provides explanations for several anomalous behaviors previously observed in NFAL systems. Additionally, leveraging insights from this stiffness-mass model, we explore NFAL applications, including the development of a self-running NFAL actuator, an NFAL motor, and a wireless floating robot.

2:00

2pPAb4. Generation of jumping of planar objects and breath-like jet flow generated by high-intensity sound pressure formed near vibrating surface. Kohei Aono (Technol. & Eng. Div., Seidensha Electronics Co., Ltd, 2-2-17 Nishinippori, Arakawa-ku, Tokyo-to 1160013, Japan, ko_aono@sedeco.co.jp) and Manabu Aoyagi (Muroran Inst. of Technol., Muroran Shi, Hokkaido, Japan)

Interesting nonlinear phenomena caused by the high-intensity sound pressure formed near the vibrating surface have been revealed. As novel object manipulation by nonlinear acoustics, the jumping phenomenon of a levitated object and the air jet flow phenomenon similar to breathing are presented. Near-field acoustic levitation (NFAL) occurs when a planar object is placed on a flat vibrating surface. However, for NFAL above the vibrating surface with a recess, a phenomenon was observed in which the levitated object jumped up when the vibration amplitude increased. The jumping phenomenon was understood to be caused by the sound pressure resonance in the cavity between the levitated object and the vibrating surface by increasing the levitation distance. A jet flow phenomenon similar to breathing was observed from a small hole when the object with the hole was placed close to the vibrating surface. Particle image velocimetry confirmed that the air did not flow through the air gap and the hole. The flow velocity of the jet changed to the object's thickness, the vibration amplitude, and the air gap between the object and the vibrating surface. The mechanism, applications to manipulation, and current research on these phenomena are described.

Contributed Paper

2:20

2pPAb5. Reversal of mid-air acoustic radiation forces: From low- to high-pressure trapping. Yusuke Koroyasu (Tsukuba Univ., Kasuga1-2, Tsukuba, Ibaraki 305-0821, Japan, koroyu@digitalnature.slis.tsukuba.ac.jp), Takayuki Hoshi (Pixie Dust Technologies, Inc., Tokyo, Japan), Yoichi Ochiai, and Tatsuki Fushimi (Faculty of Library, Information and Media Sci., Univ. of Tsukuba, Tsukuba, Japan)

Acoustic levitation in mid-air has conventionally relied on confining particles in low-pressure regions, such as standing wave nodes or twin traps. However, analyses of horizontal acoustic radiation forces at larger distances from the sound source reveal a reversal phenomenon wherein particles are instead attracted to high-pressure regions within a plane parallel to the transducer array. Through numerical simulations, we show that this effect appears not only in focused or Bessel beams but also when the phase signatures are applied. Although numerical calculations suggest that focused beams could enable mid-air levitation, stable experimental realizations have proven challenging. Here, we demonstrate for the first time that a single-sided configuration using a Bessel beam can stably levitate and translate particles in high-pressure regions, even at distances exceeding those achievable by twin traps. We further investigate whether our Bessel beam approach aligns with numerical optimal conditions for maximizing trapping stiffness. These findings underscore the potential of harnessing high-pressure regions to enhance acoustic manipulation in air, opening new avenues for single-sided levitation strategies.

Contributed Papers

2:40

2pPAb6. Robustness of stable trapping equilibria in conical shape acoustic tweezers. Jhoan Acevedo-Espinosa (School of Civil Eng. and Geomatics and Ctr. for Bioinformatics and Photonics—CiBioFi, Universidad del Valle, Cali, Valle del Cauca, Colombia), Joao Ealo (School of Mech. Eng. and Ctr. for Bioinformatics and Photonics—CiBioFi, Universidad del Valle, Ciudad Universitaria Meléndez, Bldg. E49—of.2014, Cali, Valle del Cauca 760032, Colombia, joao.ealo@correounivalle.edu.co), David Collazos-Burbano (Univ. of Sao Paulo, Ribeirao Preto, Brazil), Jhon Pazos-Ospina (School of Mech. Eng. and Ctr. for Bioinformatics and Photonics—CiBioFi, Universidad del Valle, Cali, Valle del Cauca, Colombia), and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

Single-axis acoustic tweezers employing multi-element arrays have become a low-cost and easily implemented tool for containerless manipulation in air across various applications. Planar and/or spherical transducer geometries, or a combination thereof, have been primarily used to structure the stationary or progressive acoustic field generating the intended trapping. Typically, the primary objective is to attain the highest possible sound pressure level with minimal infrastructure while maintaining the object under study at a few stable equilibrium points. However, in numerous applications, it is imperative to simultaneously maximize both the number of stable equilibrium points and the trapping forces without compromising longitudinal and transverse trapping capabilities, or increasing the infrastructure or excitation level. In this work, we assess the potential offered by acoustic sources with a conical geometry. This approach leverages their higher beam focus depth, which surpasses that provided by spherical radiators. Simulation results, validated through experiments, are presented to quantify the effect of misalignment, both angular and parallel. Furthermore, the robustness of stable equilibria in this type of trapping device when axial and transverse disturbances are applied is discussed.

3:00–3:20 Break

3:20

2pPAb7. Metamaterial-assisted acoustic manipulation of large objects in water. Dajun Zhang (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Rm. 3533, Madison, WI 53706, dajunjiayou@gmail.com) and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

Metamaterials are engineered structures that exhibit unprecedented wave-structure interactions. Recently, acoustic metamaterials were applied to reshape acoustic radiation force and assist remote manipulation of large objects (i.e., objects greater than at least twice the wavelength) in air. However, metamaterial-assisted acoustic manipulation has not been demonstrated in water before. A key challenge is the limited availability of material types and fabrication methods to create underwater metamaterials with sufficient acoustic impedance contrast with water and high-resolution structure patterns. To address that, we developed a new type of underwater metamaterial based on soft lithography of a metal-resin composite. Using the new metamaterial, we demonstrated the manipulation of objects greater than 20 times the acoustic wavelength in water. We designed metamaterial patches that enable various manipulation functions such as "pushing," "following," and "rotating," and integrated multiple functions into a single patch based on surface pattern superimposition and frequency multiplexing. Based on these fundamental functions, we further demonstrated more complex multi-object, multi-path, non-invasive, and 3-D underwater object manipulation. Our metamaterial-assisted remote underwater acoustic manipulation presents future opportunities for advancing underwater robot actuation, vehicle transportation, manufacturing, as well as drug delivery and minimally invasive surgery.

3:40

2pPAb8. Non-contact actuation of elastic lattices using Bjerknes forces. Laurin Sartori (Inst. for Molecular Systems Eng., Heidelberg Univ., Im Neuenheimer Feld 225, Heidelberg 69120, Germany, laurin.sartori@stud.uni-heidelberg.de), Peer Fischer, and Athanasios Athanassiadis (Inst. for Molecular Systems Eng., Heidelberg Univ., Heidelberg, Germany)

Noncontact actuation of small elastic structures is a challenging problem given the poor scaling of electronic actuators. Sound offers a convenient and steerable power source for actuation, but the direct conversion for actuation via primary acoustic forces is typically inefficient. We recently demonstrated that the secondary scattering forces between bubbles can be dramatically amplified by geometric patterning and used for precise actuation in a fluidic environment. Here, we build on this idea and introduce a dynamic metamaterial composed of trapped gas bubbles in a fluid. We show carefully designed structures that achieve wirelessly switchable mechanical motion via secondary acoustic radiation forces. We introduce a theoretical model of the system to describe the equilibrium configurations of the structures, and support this model with experimental results. Building on our model, we introduce design principles to achieve different actuation behaviors. Our results highlight the potential of such acousto-elastic actuators for applications in soft robotics, programmable matter, and smart materials capable of reconfigurable actuation.

4:00

2pPAb9. Lamb wave-based acoustofluidics. Feiyan Cai (Shenzhen Institutes of Adv. Technol., Chinese Acad. of Sci., 1068 Xueyuan Ave., Shenzhen University Town, Shenzhen 518055, China, fy.cai@siat.ac.cn), Jiaqi Liu, and Hairong Zheng (Shenzhen Institutes of Adv. Technol., Chinese Acad. of Sci., Shenzhen, China)

Both bulk acoustic wave (BAW) and surface acoustic wave (SAW) acoustofluidic technologies have been widely utilized for particle manipulation. However, within fluid, these waves tend to propagate as bulk waves or leaky bulk waves, resulting in undesirable acoustic streaming that causes unstable particle manipulation. Here, we introduce a novel Lamb wave-based acoustofluidic device for precisely trapping and levitating microparticles. The device consists of a piezoceramic plate coated with a one-dimensional periodic array of electrodes on one side, while the other side features a continuous electrode layer. By optimizing the electrode periodicity and plate thickness, we can efficiently excite two resonant modes: the non-leaky zero-order asymmetric (A0) Lamb waves, which are characterized by a highly periodic localized field, and the Wood's anomaly mode, which exhibits a periodic but weaker localized field. We fabricated the device and conducted experiments to manipulate particles suspended in water on the plate's surface, successfully achieving periodic trapping and levitation, as well as controlled movement of trapped or levitated particles by adjusting the phase of the source. The innovative approach to particle manipulation using Lamb waves advances the development of compact acoustofluidic systems with high precision, high throughput, and low energy consumption.

Invited Papers

4:20

2pPAb10. Applications of acoustic tweezers in microparticle manipulation. Zheng Xu (School of Phys., Tongji Univ., No. 1239 Siping Rd., Yangpu District, Shanghai 200092, China, gotoxvzheng@tongji.edu.cn) and Xiaojun Liu (Inst. of Acoust., Nanjing Univ., Nanjing, China)

In this talk, I will present our group's recent research on the application of acoustic tweezers for the trapping and transport of microparticles in liquid media. Our work focuses on phase interference generated by modulated sound sources to achieve particle trapping on surfaces, along lines, or at specific points. I will also elaborate on the generation of non-diffracting acoustic beams through Fourier transform methods or the modulation of reflective waves, which allows the acoustic source to manipulate both the amplitude and phase of the emitted waves. Utilizing these techniques, we successfully generated semi-infinite Bessel beams, Airy beams, and Weber beams, and I will discuss their theoretical implications as well as our experimental observations regarding their effectiveness in particle trapping and transport. Furthermore, I will emphasize our approach to creating tunable single-beam acoustic field configurations, which facilitate more versatile manipulation of particles compared to conventional methods.

4:40

2pPAb11. Radiation pressure deformation of fluid and elastic objects and applications: Overview and selected recent focused-traveling-wave experiments. Philip L. Marston (Phys. & Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu), Sterling M. Smith, Christopher T. Powers (Phys. & Astronomy Dept., Washington State Univ., Pullman, WA), Auberry R. Fortuner (Phys. & Astronomy Dept., Washington State Univ., Keyport, WA), Timothy D. Daniel (Phys. & Astronomy, Washington State Univ., Panama City, FL), Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Ahmad T. Abawi (HLS Res., San Diego, CA)

Low-frequency (LF) shape oscillations of levitated drops in water having amplitudes proportional to the square of the applied ultrasonic pressure were demonstrated by Marston and Apfel [J. Acoust. Soc. Am. 67, 27–37 (1980)]. The LF-resonant drop response and scaling were consistent with the spatially distributed modulated radiation pressure (MRP) of the applied double-sideband-suppressed-carrier ultrasound (DSSCU). The response was measured using light scattering. Related early developments including drop and bubble fission and reduce-gravity applications are noted in [J. Acoust. Soc. Am. 156, 1586–1593 (2024)]. Subsequent experiments concern resonant responses of elastic objects in DSSCU beams in water having short focal lengths H of about 15 cm. Recently, the measurements of LF-sound radiated by the elastic objects were extended to beams with H of 1 and 1.5 m. The radiated LF-acoustic pressure is proportional to the square of the DSSCU-source drive voltage. In some situations, the LF-mode shapes have been inferred by scanning the focal location along the elastic object. Specular reflection of the ultrasound contributes significantly to the MRP [J. Acoust. Soc. Am. 149, 3042–3051 (2021)] when the elastic objects are large in comparison to the ultrasonic wavelength. [Research supported by the US Office of Naval Research.]

TUESDAY NOON, 20 MAY 2025 SALON F/G, 12:55 P.M. TO 4:40 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Auditory Cognition in Interactive Virtual Environments

Virginia Best, Cochair
Speech, Language and Hearing Sciences, Boston University, 635 Commonwealth Avenue, Boston, MA 02215

Janina Fels, Cochair
RWTH Aachen University

Chair's Introduction—12:55

Invited Papers

1:00

2pPP1. Default spatial audio topology in virtual reality. Tanya Wen, W. Owen Brimijoin, Andrew Francl, Philip W. Robinson (Reality Labs Res. at Meta, Redmond, WA), and Antje Ihlefeld (Reality Labs Res. at Meta, 10301 Willows Rd., Redmond, WA 98052, antjeihlefeld@meta.com)

Toward more immersive sound in virtual reality (VR), three experiments investigated the influence of non-auditory feedback on the perceived spatial topology of sound in a visually constrained VR environment, designed to minimize visual cues. Participants, immersed in a minimalistic visual scene with a simple gray sphere, localized anechoically rendered, world-locked audio from various azimuths and elevations using either generic or personalized head-related transfer functions (HRTFs). The first two experiments included training sessions. In the first experiment, participants received visual feedback by pointing toward the perceived audio source and then seeing the actual direction. The second experiment involved proprioceptive training without visual feedback, where participants oriented their noses toward the sound source. The third experiment tested naive listeners without training. Results showed a consistent perceived sound direction topology but revealed differences in localization accuracy. Visual and proprioceptive feedback led to expansive perceived spatial topology. In contrast, the naive, audio-only condition resulted in the most compressed topology. In the first two experiments, feedback had a greater impact on localization accuracy than personalizing HRTFs. These findings highlight the potential to rapidly recalibrate the perceived spatial audio topology through feedback via dynamic interplay between sensory modalities in spatial cognition.

1:20

2pPP2. Embedding auditory cognitive tasks in realistic spatial contexts. G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org), Brittany T. Williams, and Angela M. AuBuchon (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Human cognition is embodied within a real-world spatial context that supports situational awareness, attention, and spatial memory. To maximize experimental control, lab-based research strips away this context, limiting the external validity of perceptual and cognitive measures. Thus, variation in such measures may strongly reflect individuals' ability to tolerate or supplement this missing context rather than the cognitive skills themselves. Virtual environments can approximate spatial aspects of the real world while affording a high degree of experimental and stimulus control, thus "bringing the real world into the lab" [Stecker, Hear J. 72(6), 20–23 (2019)] and restoring (some) contextual dependencies of real-world cognition. Here, we report on using virtual reality to embed tests of auditory attention and working memory within semi-realistic spatial/perceptual contexts. In this approach, the complexity of spatial scenes (e.g., multiple talkers) and of tasks can be varied to better match real-world settings and better assess individuals' use of context when deploying cognitive skills. [Work supported by US NIH R01-DC016643.]

1:40

2pPP3. The impact of coverbal visual cues on speech intelligibility and cognitive load in virtual reality environments. Cosima A. Ermert (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Kopernikusstr. 17, Aachen 52074, Germany, cosima.ermert@akustik.rwth-aachen.de), Andrea Bönsch, Torsten W. Kuhlen (Visual Computing Inst., RWTH Aachen Univ., Aachen, Germany), and Janina Fels (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Aachen, Germany)

In natural communication settings, auditory information, including spatial cues and spectro-temporal variations, is typically accompanied by corresponding coverbal visual cues, specifically information about the talker's location and lip movements. These visual cues can enhance speech intelligibility in noisy environments compared to audio-only conditions but may also increase cognitive load as listeners process information from multiple modalities simultaneously. While numerous studies have examined the effects of isolated visual cues on auditory processing, often using faces displayed on computer screens, the interplay and relevance of these cues in natural scenarios with fully rendered embodied conversational agents remain largely underexplored. This study aims to identify which coverbal visual cues enhance speech intelligibility and which contribute to an increased cognitive load in a virtual reality (VR) environment. Specifically, we evaluate two types of visual cues that directly correspond to information conveyed in the auditory signal: spatial information and lip movements. Using a conversational setting in VR, we assess speech intelligibility and cognitive load with the Oldenburger Sentence Test (OLSA) and verbal response times under varying levels of coverbal information. By investigating these cues in a more natural VR setting, the results contribute to a deeper understanding of multimodal speech processing. [This work was funded by the German Research Foundation (DFG): SPP2236—444724862.]

2:00

2pPP4. Virtual reality potential as a platform to measure listening effort. Kristina DeRoy Milvae (Dept. of Communicative Disord. and Sci., Univ. at Buffalo, 122 Cary Hall, South Campus, Buffalo, NY 14214, klmilvae@buffalo.edu), Ian Phillips (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Mythili Thamilchelvam, Shifali Chambers, Uzaira Sethi (Dept. of Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY), Jacob Lefler, Stefanie E. Kuchinsky, and Douglas Brungart (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Virtual reality (VR) is usually used to simulate complex visual scenes with a head-mounted display (HMD). However, VR can also be used to obtain better control over the visual environment and to monitor ocular dynamics during listening experiments. In this study, a low-cost HMD was used to track eye gaze and pupil diameter during a dichotic listening task in a simple virtual environment that simulated a computer display. The pupillometry results from the VR environment were compared to those obtained on the same task with a desktop monitor research-grade eye tracker. Participants recalled digits in the right ear and ignored interfering digits in the left ear. It was hypothesized that greater pupil dilation would be observed for dichotic compared to monotic listening with both HMD and research-grade eye trackers. Preliminary data support this hypothesis. The HMD was also evaluated for its potential to collect eye-tracking data in tablet-based clinical speech-in-noise tasks not originally designed for pupillometry. Results suggest that the task-evoked pupil response can be measured with the HMD on clinical tasks and a VR-based system has potential as a low-cost clinical tool to collect objective measures of listening effort.

2:20

2pPP5. Auditory spatial attention after dyadic conversation in a cocktail party situation. Lubos Hladek (Acoust. Res. Inst., Austrian Acad. of Sci., Dominikanerbastei 16, Vienna 1010, Austria, lubos.hladek@oeaw.ac.at), Piotr Majdak, and Robert Baumgartner (Acoust. Res. Inst., Austrian Acad. of Sci., Vienna, Austria)

Is auditory spatial attention susceptible to short-term fatigue from effortful communication in noise? We hypothesize that people lose spatial attention after effortful communication in noisy environments. In the current experiment, normal-hearing participants had a free 30-min face-to-face conversation with a fellow participant in the laboratory. Headphones, microphones, and motion tracking provided world-stable virtual acoustic space, while preserving the direct sound. Before and after the conversation, participants conducted a test of auditory spatial attention made of a syllable streaming task. In the task, participants heard two streams of syllables from the left and the right. Possible syllables were: "ba," "da," and "ga" spoken by a male and a female talker. The task was to repeat the syllables from the target stream denoted by a prior auditory spatial cue. Data were analyzed as a change in spatial auditory attention from the pre- to post-test. The experimental condition with the conversation was compared to the control conditions with no conversation, or no noise. The goal of the study is to understand the effect of listening effort and short-term cognitive fatigue on attention control and provide ecologically valid tools for the assessment of cognitive function within acoustic scenes.

2:40–3:00 Break

3:00

2pPP6. Space Party Rescue: An auditory game of finding your friend at a cocktail party. William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, wbologna@towson.edu), Esteban Sebastian Lelo de Larrea-Mancera (Psych., Northeastern Univ., Boston, MA), Kelly Avery (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and Aaron R. Seitz (Psych., Northeastern Univ., Boston, MA)

Virtual reality (VR) technology has enabled researchers to investigate real-world communication behaviors within immersive simulated environments. Twenty adults aged 19–37 years with normal hearing completed a tablet-based spatial release from a masking game (Pipes Puzzle) and a VR game (Space Party Rescue). In Space Party Rescue, participants navigated a virtual party on a spaceship to find specific talkers based on the topic of their monologue. The number of seconds required to locate each talker was recorded as primary data. The average time to find each talker in Space Party Rescue was moderately correlated with the target-to-masker ratio threshold with colocated maskers, measured with Pipes Puzzle ($r = 0.459$). This result indicates some shared variance between the two games in assessing speech recognition at a positive target-to-masker ratio and reflects a common behavioral strategy in Space Party Rescue; participants navigated to positions with favorable signal-to-noise ratio to locate the targets, rather than scanning the scene from a fixed position using spatial cues. Participants reported greater enjoyment and perceived success in Space Party Rescue compared to Pipes Puzzle, which may reflect the immersive nature of the VR game, and the greater autonomy afforded to participants in Space Party Rescue.

3:20

2pPP7. Influence of acoustic and non-acoustic environmental and individual factors on speech perception in real-world complex listening environments. Erik Jorgensen (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, SCH 307, 250 Hawkins Dr., Iowa City, IA 52246, erik-j-jorgensen@uiowa.edu), Hendrik Kayser (Univ. of Oldenburg, Oldenburg, Germany), Theresa Jansen (Horzentrum Oldenburg, Oldenburg, Germany), and Volker Hohmann (Univ. of Oldenburg, Oldenburg, Germany)

Virtual acoustic environments can capture the acoustic properties of real-world environments with remarkable realism. However, listening performance in real-world complex environments could depend on non-acoustic environmental factors that are not as easily reproduced in the lab (environment type, signal location, listening activity, visual cues, listening activity, noise sources) as well as individual perceptual factors (hearing ability, working memory, attention). In this preliminary study, we investigated the contributions of acoustic and non-acoustic environmental factors and individual factors to speech perception in real-world complex listening environments using a hearing aid research platform, the Portable Hearing Lab (PHL), and ecological momentary assessment. Participants completed assessments using the PHL in complex listening environments they encountered in their life over the course of a week. Using all-subset analyses, the results suggest that hearing ability and signal-to-noise ratio are the most important contributors to speech perception in complex environments. However, listening activity, signal location, and environment type also contributed significantly. Surprisingly, we did not find significant effects from visual cues or cognitive abilities. The results can be used to inform designs of ecological experiments using virtual acoustic scenes by highlighting the importance of including scene type, signal location, and task as factors in the experiment design.

3:40

2pPP8. Pros and cons of using virtual reality in clinical research for hearing device evaluation and fitting. Maartje Hendrikse (Otorhinolaryngology and Head and Neck Surgery, Erasmus MC Univ. Medical Ctr. Rotterdam, Dr. Molewaterplein 40, Rotterdam 3015 GD, Netherlands, m.hendrikse@erasmusmc.nl), Gertjan Dingemanse, Jantien Vroegop, and André Goedegebure (Otorhinolaryngology and Head and Neck Surgery, Erasmus MC Univ. Medical Ctr. Rotterdam, Rotterdam, Netherlands)

In recent studies, virtual reality (VR) was explored as a tool for the fitting and evaluation of hearing devices. The first study evaluated a novel VR-based method for fine-tuning hearing aids and cochlear implants. This approach allowed patients to trial various device settings within realistic VR environments under audiologist supervision. By tailoring settings to individual preferences in everyday scenarios, the need for repeated adjustments during follow-up appointments was reduced, enhancing efficiency and personalization. The second study investigated the effectiveness of an automatic scene classifier in children with cochlear implants. The classifier activated a directional microphone mode in specific environments to improve the signal-to-noise ratio when speech originated from the front. However, this mode may hinder children's ability to understand speech if they cannot orient toward the speaker. To study this, a novel listening test was developed in a VR classroom. The test assessed speech intelligibility by requiring repetition of digits presented from the front and evaluated spatial awareness by distinguishing between animal names from different directions. This presentation will critically examine the advantages and limitations of employing VR in clinical research, drawing insights from these two projects to highlight its potential in audiological applications.

4:00

2pPP9. Assessing speech comprehension ability and hearing aid benefit using realistic audio-visual conversation materials. Jorg M. Buchholz (Macquarie Univ., Sydney, New South Wales 2109, Australia, jorg.buchholz@mq.edu.au), Xinyu Guo, Lisa Maggs, Ronny Ibrahim, Kelly Miles (Macquarie Univ., Sydney, New South Wales, Australia), Stefan Raufer, and R. Peter Derleth (Sonova AG, Stäfa, Zürich, Switzerland)

There is a lack of sensitive assessments that can be used to reliably evaluate and optimize the real-world benefit of hearing devices in noisy, multi-talker environments. We have therefore developed a real-time speech comprehension test that is based on the word-monitoring paradigm and utilizes short audio-visual recordings of spontaneous two-talker conversations on different topics (e.g., "seasons"). The participant's task is to press a hand-held button each time they hear a target word related to the topic (e.g., "summer"). Their performance is measured by the accuracy in identifying target words (i.e., hit rate) and associated response times. The conversation recordings were presented in a realistic café and dinner party environment using a 41-channel loudspeaker array and a cylindrical video projector system. Additionally, the participant's head- and eye movements were recorded using Tobii eye-tracking glasses and a Vicon motion-tracking system. We tested 18 adults with moderate to moderate-severe hearing loss who were fitted with research hearing aids that provided different speech enhancement strategies, including an adaptive bilateral beamformer and a DNN-based algorithm. Results revealed that with increasing task difficulty (i.e., environment noise level) the hit rate decreased and response time increased. Overall, the DNN-based algorithm resulted in the highest hit rates and shortest response times.

4:20

2pPP10. The neural underpinnings of attention and distraction in virtually real environments. Orel Levy (Brain Res. Ctr., Bar Ilan Univ., Ramat Gan, Israel) and Elana Zion Golumbic (Brain Res. Ctr., Bar Ilan Univ., Bldg. 901, Ramat Gan 5290002, Israel, elana.zion-golumbic@biu.ac.il)

The ability to maintain focused attention toward a particular task or speaker and avoid distraction by irrelevant background events is crucial for many aspects of real-life behavior, including learning, memory, social communication, decision-making, and self-control. However, empirical lab-based research into the cognitive and neural mechanisms underlying the constructs of 'attention' and 'distraction' has primarily focused on highly artificial paradigms, stimuli and tasks that are a far cry from the challenges of attention in real-life environments. Unfortunately, there is a growing realization that insights gained from these type of studies do not generalize well or explain behavior in real-life settings. To bridge this lab-to-real-life gap, in this talk, I will present data collected using a novel VR-based experimental platform, designed for studying neural, ocular, and physiological manifestations of selective attention and distraction, under ecologically realistic conditions that simulate those we need to deal with on a daily basis. Using two common-day scenarios, a Virtual Café and Virtual Classroom, we show how neural processing of task-relevant speech is affected by background stimuli and noise, and how the sensitivity to irrelevant stimuli varies across individuals. We also discuss implications for theories of attention and possible clinical implications for ADHD research.

TUESDAY NOON, 20 MAY 2025 BALCONY M, 12:55 P.M. TO 5:00 P.M.

Session 2pSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials and Phononic Crystals II

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Alexey Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, Bethesda, MD

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward Street, Ann Arbor, MI 48109

Behrooz Yousefzadeh, Cochair
Concordia University, Montreal, 1455 De Maisonneuve Blvd. W., Room EV-4.139, Montreal, H3G 1M8, Canada

Hussein Nassar, Cochair

Chair's Introduction—12:55

Invited Paper

1:00

2pSA1. Foldable acoustic metamaterial for underwater broadband low-frequency noise mitigation. Junfei Li (Mech. Eng., Purdue Univ., West Lafayette, IN, junfeili@purdue.edu), Yijie Zhang, and Shaocheng Wu (Mech. Eng., Purdue Univ., West Lafayette, IN)

Monopile offshore wind foundations are typically installed using hydraulic impact hammers, generating intense underwater noise that can propagate over 50 km from the installation site. This noise poses significant risks to marine life, potentially causing auditory injury and behavioral disturbances. Existing noise mitigation strategies, such as bubble screens, are hindered by high costs, energy demands, and limited scalability, while resonance-based metamaterials are constrained by narrow operational bandwidths. Developing

broadband, efficient underwater noise mitigation solutions that are cost-effective, scalable, and easy to manufacture remains a critical challenge. In this talk, we present a novel acoustic metamaterial design that delivers ultra-broadband noise mitigation at low frequencies with a compact, deep subwavelength structure. The design is modular and fully foldable, enabling mass production and simplifying transportation for large-scale offshore applications.

Contributed Papers

1:20

2pSA2. Elastic bits in granular metamaterials: A quantum-inspired framework for robust topological computing. Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Ste. #2100, Detroit, MI 48201, hk2799@wayne.edu), Afridi Hasan (Comput. Sci. and Eng., Univ. of Arizona, Tucson, AZ), M Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI), Pierre A. Deymier, and Keith Runge (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ)

This study introduces the concept of elastic bits in a driven granular metamaterial. These elastic bits function similarly to quantum bits (qubits), but they can maintain more stable superpositions by forming topological braiding structures with robust information stored in their phase. We have recently developed models using classical nonlinear granular metamaterials to simulate quantum-inspired computing. Using a two-coupled granular system, we analyze nonlinear vibrations and eigenstates through molecular dynamics simulations and Fourier analysis to map displacement fields into Hilbert spaces. These models demonstrate states resembling a multipartite two-level quantum system, where each subsystem represents an elastic bit analogous to time-dependent coherent superpositions of elastic bits. The study demonstrates the creation of two and three elastic bits, represented as tensor products of eigenstates in higher-dimensional Hilbert spaces. By manipulating these elastic bits—similar to quantum braiding operations—we can adjust the phase of the wave function. This allows for various types of braiding that enable robust quantum-inspired computations. Notably, "time" serves as the direction of braiding in this setup. [Funding: NSF Grants 2204382 and 2242925.]

1:40

2pSA3. Multimodal resonant metamaterials for broadband sound transmission loss improvement. Edwin Reynders (Dept. of Civil Eng., Structural Mech. Section, KU Leuven, Kasteelpark Arenberg 40, Leuven 3001, Belgium, edwin.reynders@kuleuven.be), Daniele Giannini (Dept. of Civil Eng., Structural Mech. Section, KU Leuven, Leuven, Belgium), and Mattias Schevenels (Dept. of Architecture, Architectural Eng. Res. Group, KU Leuven, Leuven, Belgium)

Locally resonant metamaterials (LRMs) are a promising approach for enhancing the vibroacoustic attenuation of partitions. LRMs originate when adding small resonators on the microscale (sub-wavelength scale) to a host structure for targeting exceptional material properties on the macro-scale. While conventional LRMs exploit single-mode translational resonators resulting in narrowband attenuation, the potential for broader attenuation offered by resonators with multiple translational and rotational modes has been recently demonstrated through numerical simulation. However, their effectiveness has not yet been experimentally demonstrated, and the optimized design of multi-modal resonators for broadband attenuation is highly nontrivial. This work focuses on both aspects. First, a multimodal rotational LRM for a cross-laminated timber host structure and medium-density fiberboard resonators is designed and manufactured. A subsequent transmission loss test reveals the effective suppression of the broad coincidence transmission loss dip of the orthotropic host panel through two resonance peaks. Next, a methodology for optimized resonator design is developed. The optimization objective is to maximize the broadband diffuse transmission loss while constraining mass. The topology optimization problem is solved by gradient-based mathematical programming, for which the necessary (adjoint) sensitivities of objectives and constraints are derived. The efficacy of this approach is demonstrated through detailed numerical simulation.

2:00

2pSA4. Design and experimental demonstration of fabric phononic crystals. Michael Y. Wang (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., EH 3533, Madison, WI 53706, michael.wang@wisc.edu) and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

Fabric materials have been extensively studied and utilized in various applications. Acoustic metamaterials are engineered structures that possess unique material properties and wave control capabilities not found in naturally occurring materials. Although there has been a profusion of research on both fabrics and acoustic metamaterials, little attention has been given to combining these two areas. This research introduces fabric phononic crystals, an innovative class of acoustic metamaterials formed through periodic patterns in woven-based fabrics with tailored mechanical properties. The fabric phononic crystals are designed to exhibit application-specific band structures for acoustic wave control. The mechanical properties of the woven fabrics are experimentally characterized. The out-of-plane vibration modes of the fabric phononic crystals and their frequency responses are simulated using a homogenized plate model and experimentally validated. Bandgaps at designated frequency ranges are demonstrated. This study bridges the gap between metamaterial research and its real-world implementations in textiles, highlighting the potential of fabric phononic crystals as a lightweight, flexible, and scalable solution for applications such as vibration protection and wearable sensing.

2:20

2pSA5. Various band structure excitations in twisted bilayer phononic graphene with layer-resolved measurement. Chenzhe Wang (Mech. Eng., Univ. of Michigan, G.G. Brown Bldg., 2350 Hayward St., Rm. 2651, Ann Arbor, MI 48105, czwang@umich.edu), Jiawei Ruan (UC Berkeley, Berkeley, CA), Steven R. Craig (Naval Undersea Warfare Ctr., Div. Newport, Newport, RI), Zhenglu Li (Univ. of Southern California, Berkeley, CA), Yan Deng (Mech. Eng., The Univ. of Michigan, Ann Arbor, MI), Steven G. Louie (UC Berkeley, Berkeley, CA), and Chengzhi Shi (Univ. of Michigan, Ann Arbor, MI)

Twistronics, an exciting upsurging field, has been generated by studying the twisted moiré bilayer graphene, where many phases of fundamental and practical interests are observed. Flat bands are formed in twisted bilayer graphene through intralayer and interlayer coupling at magic twisting angles. However, the coupling between the two layers with subtle interlayer interactions has not been directly observed experimentally. This is limited in reality due to the lack of non-invasive probes between layers of graphene, which is in the order of nanometers. Here, we introduce a phononic solution for direct measurement. PTBG is an analog to the electronic system. Due to its larger structural sizes, direct measurements of the excitations from each layer are possible, enabling *in situ* layer resolution of the special flat-band excitations. The layer-resolved phononic measurement, together with theoretical modeling, directly demonstrates that the interlayer coupling effect in various angled PTBG, among them, the magical angled twisted case is most significant. On the other hand, in large-angle twisted PTBG, the low-energy states are only mildly affected by the moiré effect and can be approximated by those from two decoupled Dirac cones from the individual phononic layers. This work opens new phononic approaches to investigate moiré physics.

2:40–3:00 Break

3:00

2pSA6. Evaluating the relationship between print speed and acoustic performance in additively manufactured parts. Joshua A. Bakheet (Appl. Res. Labs., The Univ. of Texas at Austin, 8150 Whitehead Pl., La Mesa, CA 91942, jbakheet5440@sdsu.edu), Brooklynn petty, Nathan P. Geib, Samuel P. Wallen, and Christina Naify (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Additive manufacturing (AM) is often used as a form of rapid prototyping, allowing for quick fabrication of structures for a variety of applications. Most AM processes, however, are known to have poor dimensional tolerancing when compared to manufacturing methods such as molding or machining. In order to understand to what extent an AM process really is rapid, it is desirable to understand how the speed of the fabrication process correlates to the desired performance of the final structure. This study examines how additive manufacturing parameters, including print speed, layer height, and acceleration, affect the acoustic properties of microperforated 3-D printed absorbers. Systematic variations in print conditions produced significant changes in hole geometry fidelity, such as eccentricity and area, which directly influenced the panels' sound absorption characteristics. Acoustic testing was conducted in an air-acoustic impedance tube in accordance with ASTM E2611 standards. When combined with simplified predictive models, these tests revealed shifts in resonance frequencies and absorption coefficients. Although faster print settings reduced fabrication time by over 80%, geometric precision and acoustic performance were often compromised. Potential solutions include adjusting geometries or refining printing strategies to maintain desirable acoustic properties while increasing throughput.

3:20

2pSA7. Internal damping measurements of beams with an acoustic black hole taper. Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, peterkj@byu.edu), Joshua T. Mills, and Micah Shepherd (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Aluminum beams which taper to a thin central point according to a power law exhibit acoustic black hole (ABH) behavior. As the beam becomes very thin, approaching a non-zero critical thickness, the wave speed decreases, trapping flexural waves to a localized region of high displacement. Typically, damping material is applied to this region for noise control applications. However, recent measurements have shown that increased modal loss factors are achieved even without additional damping. To investigate this, we present loss factor measurements for tapered beams in the absence of dissipative effects, including that of acoustic radiation into the surrounding medium, to determine what proportion of the increased damping is caused by acoustic losses. This is done by exciting the beam inside a vacuum chamber and measuring its flexure using laser vibrometry, then comparing it measurements performed outside of the vacuum chamber. Using beams with different ABH parameters, loss factor measurements are shown to be highly dependent on the critical thickness and greater than those of a non-tapered beam. These results will be discussed in the context of previous work on structures with ABH tapers.

3:40

2pSA8. Fast-speed ultrasonic imaging using a single sensor by combining disordered medium and artificial intelligence. Wei Wang (Inst. of Acoust., Nanjing Univ., No. 22, Hankou Rd., Nanjing, Jiangsu 210093, China, weiwang@smail.nju.edu.cn), Jie Hu (Dept. of Information Sci. and Technol., Nanjing Forest Univ., Nanjing, Jiangsu, China), Jingjing Liu, Bin Liang, and Jian-chun Cheng (Inst. of Acoust., Nanjing Univ., Nanjing, Jiangsu, China)

Imaging using ultrasound is a frontier technique in modern science owing to its high penetration and biocompatibility, with pivotal significance in many applications ranging from crack detection to biological tissue imaging. However, existing technologies fail to support both single-sensor imaging and real-time imaging concurrently because imaging requires data from multiple points in the ultrasonic field, inevitably leading to an increase in either time or fabrication costs. To overcome the above fundamental limitations, we propose and experimentally demonstrate a mechanism for real-time ultrasonic single-sensor imaging based on the coalescence of disordered medium and artificial intelligence (AI), where the disordered medium can encode the information required for imaging into the frequency spectra, while a neural network can decode the signal captured by the single sensor and reconstruct the original object's image. A disordered medium at a cost much less than $1 and a single fixed probe are proved sufficient to yield high imaging quality.

4:00

2pSA9. Tuning the sound-field of an electrodynamic transducer enclosed by an acoustic metamaterial lens with horn-like geometry. Gregory M. Hernandez (Elec. and Comput. Eng., Duke Univ., 101 Sci. Dr. Rm. 3417, FCIEMAS Bldg., Durham, NC 27708, gregory.hernandez@duke.edu), Junfei Li (Mech. Eng., Purdue Univ., West Lafayette, IN), Xiuyuan Peng, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

The radiation pattern of an acoustic source is naturally directional at high frequencies as the emitted wavelength becomes comparable to the size of the oscillating surface. Directional sound-fields are unfavorable in circumstances where sound source quality is necessary for communicating important and urgent messages to a large population, measuring the acoustic impulse response of a room, or listening to a concert at a venue. Additionally, due to this beaming, a loudspeaker's bandwidth is strictly limited. This introduces several transducers varying in size and reproductive capability to fully cover the bandwidth. Acoustic lenses have been a solution to altering the sound-field of acoustic sources, but their complex, narrowband, and bulky designs have not found an impactful application since their inception. However, with the expanding efficiency of 3-D printing and the introduction of acoustic metamaterials, compact, broadband, cost-effective, and complex acoustic lenses can be engineered. Discussed here is an acoustic lens that utilizes twisting horn-like labyrinthine metamaterials to alter the directional sound-field emitted from a loudspeaker. Experimental validation, including 2-D and 3-D simulation of the source and lens, is presented to motivate the utility of an acoustic metamaterial lens in producing an omnidirectional sound-field from a directional source.

4:20

2pSA10. The automated inverse design of pentamode metamaterials using deep reinforcement learning and generative modeling. Cheng Qiu (Mech. Eng., San José State Univ., 1 Washington Sq, San Jose, CA 94520, cheng.qiu@sjsu.edu) and Feruza Amirkulova (Mech. Eng., San José State Univ., San Jose, CA)

This study explores the automation and optimization of pentamode metamaterial (PM) unit cell design, mimicking water's acoustic properties while ensuring its manufacturability. A hybrid framework combining COMSOL Multiphysics simulations with machine learning, generative modeling, and reinforcement learning (RL) was developed. The workflow efficiently generates diverse geometric configurations, conducts automated simulations, and analyzes the physical properties of PM unit cells. A Conditional Variational Autoencoder (CVAE), embedded within the RL framework, predicted the relationships between geometric parameters and outputs like bulk modulus, shear modulus, density, and impedance. Proximal Policy Optimization (PPO) agents were trained over dynamic episodes using custom reward functions to achieve targeted properties, such as maximizing the bulk-to-shear modulus (B/G) ratio. The reward functions prioritized alignment with predefined physical targets while penalizing deviations from manufacturability and structural feasibility. By dynamically adapting learning rates and leveraging the predictive capabilities of the CVAE, the framework efficiently explored the design space and optimized PM unit cells for targeted properties. Results demonstrated the efficacy of the data-driven framework, achieving a substantial improvement in the B/G ratio of 170, highlighting its potential in efficiently modeling manufacturable PM structures and devices thereby enabling fast adaptation of PM in engineering applications and stealth technologies.

4:40

2pSA11. Super resolution focusing of sound among a Helmholtz resonator array: Nonlinear aspects. Mark R. Carlisle (Dept. of Phys. & Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, cmarkr@student.byu.edu) and Brian E. Anderson (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) is a technique used to focus wave energy to a selected location. High energy TR focusing has applications in biomedical ultrasound and nondestructive evaluation of cracks or defects in solids. These applications can benefit from the ability to narrow the spatial extent of the focused sound energy. Two-dimensional Helmholtz resonator arrays placed in the near field of TR focusing have been shown to produce a sub-diffraction limited spatial extent of the focused energy (when compared to the free-space wavelength). There is an apparent amplitude dependence on this focus and this presentation will discuss these nonlinear aspects. These observations were made by analyzing the experimental results of TR focusing on an array of soda cans at different sound levels. These nonlinear effects occur at much lower sound levels than is typical for nonlinear waveform steepening.

TUESDAY AFTERNOON, 20 MAY 2025 STUDIO FOYER, 1:20 P.M. TO 4:20 P.M.

Session 2pSC

Speech Communication: Speech Perception Poster Session I

Pertti Palo, Chair
Speech, Language and Hearing Sciences, Indiana University, 2631 East Discovery Parkway, Bloomington, IN 47408

All posters will be on display from 1:20 p.m. to 4:20 p.m. Authors of odd-numbered papers will be at their posters from 1:20 p.m. to 2:50 p.m. and authors of even-numbered papers will be at their posters from 2:50 p.m. to 4:20 p.m.

Contributed Papers

2pSC1. Perception of noise-vocoded Mandarin Chinese lexical tones by native and L2 listeners. Ning Zheng (Dept. of Linguist, Purdue Univ., 3384 Peppermill Dr., Apt 1B, West Lafayette, IN 47906, zheng874@purdue.edu)

The production, acoustics, and perception of Mandarin lexical tones have been studied extensively. Mandarin Chinese listeners rely heavily on the perception of lexical tones to distinguish word meanings and decode linguistic information. Mandarin Chinese has four lexical tones: high-level tone 1, high-rising tone 2, low-fall-rise tone 3, and high-falling tone 4. Fundamental frequency (F0) is the acoustic correlate of pitch in tones, but other acoustic cues can also contribute to tonal information. Noise-vocoded speech is thought to simulate the speech perceived by hearing-impaired listeners through a cochlea implant. When the speech passed through a noise vocoder with a specified channel number, the spectral information was replaced by random noise in the amplitude envelope across several frequency bands. F0 contour is absent in such degraded speech signals of noise-vocoded speech. The study aims to investigate normal-hearing listeners' ability to identify noise-vocoded tones with a limited number of acoustic cues. The results show that native listeners are able to distinguish the four tones, but the level tone is the most difficult to identify. L2 listeners show a different picture in tone identification. An effect was observed in the tone identification across vocoded bands and tone categories. Besides, there is a certain order of perceptual difficulty and a pattern of mislabeled tendency.

2pSC2. Examining the effect of noise exposure in group fitness classes on auditory processing ability. Siana Lai (CUNY, 365 5th Ave., New York, NY 10016, slai1@gradcenter.cuny.edu) and Clare DeGennaro (CUNY, New York, NY)

This study aims to determine the possible effects of repeated temporary threshold shifts from group fitness classes on phoneme recognition in noise ability. Past research has shown that cochlear synaptopathy resulting from repeated exposure to loud environments can result in recovery of hearing thresholds but auditory processing may remain affected. High levels of noise exposure in group fitness classes may expose young people to an increased risk of permanent auditory processing damage. To our knowledge, there have not been any studies linking noise exposure in group fitness classes and speech in noise processing. Participants were asked to complete a sound recognition in noise test with ratios of noise ranging from −10 SNR to 20 SNR. Consonant-Vowel-Consonant (CVC) English words varying in initial consonant and medial vowel were presented in noise. Participants were then asked to complete a survey probing the level and frequency of noise exposure, hearing health background and persistent difficulties with hearing. Based on previous research we hypothesize participants who have experienced repeated temporary threshold shifts from noise exposure in group fitness classes would have lower accuracy on speech in noise phoneme perception. Data collection is underway with a recruitment goal of 50 participants, our findings will contribute to the literature by examining the link between auditory processing and noise exposure.

2pSC3. The effect of noise types on working memory and speech-in-noise recognition. Xinming Zhou (Univ. of Texas at Austin, 2504A Whitis Ave., (A1100), Austin, TX 78712, xinming.zhou@utexas.edu) and Chang Liu (Univ. of Texas at Austin, Austin, TX)

The Ease of Language Understanding (ELU) model suggests that working memory is more engaged when speech perception is affected by signal distortion, especially in individuals with hearing impairments. While research by Oberfeld *et al.* (2024) found no significant relationship between working memory performance in noisy environments and speech-in-noise (SiN) understanding, the impact of noise types on the relationship between working memory and speech perception tasks is yet explored. This study aims to investigate how different types of background noise, specifically long-term speech-shaped noise (LTSSN) and multi-talker babble, influence the relationship between working memory and SiN recognition among monolingual adults. English-native monolingual adults with normal hearing will be assessed with their memory performance through digit recall tasks and speech recognition tasks in both quiet and noise across two types of noise and various signal-to-noise ratios (SNRs). Overall, significant correlations between working memory performance in noise and SiN recognition are expected with multi-talker babble demonstrating a stronger relationship than LTSSN. Findings from this research will enhance our understanding of how various types of noise affect cognitive processing during speech recognition and clarify the role of working memory in adapting to challenging listening environments.

2pSC4. Effects of fundamental frequency and vocal tract resonance on sentence recognition in noise. Jing Yang (Dept. Commun. Sci. and Disord., Univ. of Wisconsin-Milwaukee, 2400 E Hartford Ave., Enderis 873, Milwaukee, WI 53201, jyang888@uwm.com), Xianhui Wang (Univ. of California, Irvine, Irvine, CA), Victoria Costa, and Li Xu (CSD, Ohio Univ., Athens, OH)

This study examined the effects of change in the talker's sex-related acoustic properties [fundamental frequency (F0) and vocal tract resonance (VTR)] on speech recognition in noise. The stimuli were HINT sentences with the original male talker's F0 and VTR being manipulated (doubling F0 and/or scaling up VTR by a factor of 1.2) into four conditions: low F0 low VTR ($L_{F0}L_{VTR}$, the original recordings), low F0 high VTR ($L_{F0}H_{VTR}$), high F0 high VTR ($H_{F0}H_{VTR}$), and high F0 low VTR ($H_{F0}L_{VTR}$). Randomly selected sentences from each condition were presented to 193 adults for a gender rating task on a 7-point scale. Then, all sentences were mixed with speech-shaped noise at signal-to-noise ratios of −10, −5, 0, and +5 dB, and presented to 42 normal-hearing adults for recognition. The two conditions with matched F0 and VTR ($H_{F0}H_{VTR}$ and $L_{F0}L_{VTR}$) were perceived as male or female voices and showed no significant differences in recognition accuracy and estimated speech reception thresholds. However, the mismatched conditions $H_{F0}L_{VTR}$ and $L_{F0}H_{VTR}$ showed reduced recognition performance and significantly higher SRTs than the matched conditions. In general, voices with matched F0 and VTR yield equivalent speech recognition in noise, whereas voices with mismatched F0 and VTR may reduce intelligibility in noise.

2pSC5. Gamification: Virtual medical scenarios to explore the impact of hospital noise on comprehension and memory. Sarah E. White (Lingustics, The Univ. of Chicago, 6024 Nicholas Glen, Columbus, OH 43213, sarsar10green@gmail.com), Dakota Whisler (Lingustics, Univ. of Oregon, Eugene, OR), Alex Holly (Univ. of Rochester, Chicago, IL), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska–Lincoln, Omaha, NE), and Melissa Baese-Berk (Linguist, Univ. of Chicago, Chicago, IL)

Hospital noise levels often exceed World Health Organization recommendations, raising concerns about their impact on spoken communication. Bent *et al.* (2022) examined this issue by presenting medical sentences to participants in quiet, speech-shaped, and hospital noise conditions. Their findings demonstrated that both types of noise reduced speech intelligibility, underscoring the challenges noise presents to effective communication in clinical settings. While this research highlights noise's detrimental effect on intelligibility, additional studies suggest that noise can also increase cognitive load potentially impairing comprehension and subsequent memory. It is essential to explore how noise influences cognitive processes to better understand its effects on healthcare outcomes. Our study extends this research by investigating the impact of hospital noise on comprehension and memory. Participants assume the role of a patient and complete four virtual medical scenarios; two in quiet and two in hospital noise. After each scenario, participants complete questions designed to assess their ability to recall, integrate, and extend information from a given scenario. In addition to scenario-based tasks, participants undergo assessments of hearing, memory, and auditory comprehension. These include pure-tone audiometry, the Wechsler Memory Test, and the NIH Toolbox Picture Vocabulary Test. Data collection is ongoing at the University of Chicago and Indiana University Bloomington, involving young (18+) and older adults (65+).

2pSC6. A comparative study of the influence of voice type on speech-in-noise tests and questionnaires among non-native English speakers. Won So (Commun. Sci. and Disord., Grand Valley State Univ., 1628 W 6th St. Apt B, Apt C107, Austin, TX 78703, wonso1983@gmail.com), Michael Degennaro (Commun. Sci. and Disord., Grand Valley State Univ., Grand Rapids, MI), and Sungmin Lee (Speech-Lang. Pathol. and Aural Rehabilitation, Tong Myong Univ., Busan, Korea)

Non-native English speakers often struggle in noisy environments due to their language background, limited ability to identify contextual clues, and the presence of competing auditory stimuli. These challenges resemble those faced by individuals with hearing impairments; however, the characteristics of their listening experiences differ. This study focuses on the voice preference between male and female speakers in noisy environments. The research investigates how non-native English speakers perceive voice types in competing auditory conditions and measures their listening performance using a speech-in-noise test. Preliminary findings suggest that participants prefer female voices in noisy environments, likely due to the higher pitch, which is easier to detect. A customized questionnaire, adapted from the Hearing Handicap Inventory for Adults, aligns with these findings and highlights the preference for female voices. The speech-in-noise test results indicate equivalent performance for male and female voices at higher signal-to-noise ratios. However, at lower signal-to-noise ratios (+5 and 0 dB), participants showed better performance with female voices. These results suggest that vocal pitch plays a critical role in listening outcomes under challenging auditory conditions. The implications of these behavioral findings and questionnaire results will be further discussed in detail.

2pSC7. Effects of stimulus repetition on L2-accented English sentence recognition thresholds and psychometric function slopes. Tiana Cowan (Childhood Deafness, Lang. and Learning, Boys Town National Res. Hospital, 401 N 46 St., 4107, Omaha, NE 68132, Tiana.cowan@boystown.org), Lori Leibold (Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Emily Buss (Dept. of Otolaryngology/Head and Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Characterizing second language (L2)-accented sentence recognition in noise using traditional methods requires large corpora of stimuli, which are not always available. Repeating sentences at ascending target-to-masker ratios (TMRs), instead of using novel sentences for each trial, reduces the number of sentences required for a given experiment. For native-accented speech, repeating sentences tends to elevate sentence recognition thresholds (SRTs) and flatten psychometric function slopes compared to using novel stimuli for each trial; poorer performance for repeated stimuli may be due to error perseveration, whereby incorrect recognition at low TMRs is retained as the TMR increases. The present study extends that work with native-accented speech to evaluate how using repeated versus novel sentences influences L2-accented speech recognition. Repeating L2-accented stimuli could be more prone to error preservation due to unfamiliar pronunciation or prosody. Twenty-two English monolingual adults with normal hearing were tested with sentences in three accent conditions: Midland (native), Korean, and Spanish. The masker was a speech-shaped noise. Tasks were blocked by the test method (novel or repeated stimuli), counterbalanced across participants, with accent conditions randomized within blocks. The effects of stimulus repetition and L2 accent on SRTs and slopes will be discussed.

2pSC8. Locating fricative categories in a synthetic acoustic space: Spectral moments measures of a perceptual English fricative system. Kenneth J. de Jong (Dept. of Linguist, Indiana Univ., Ballantine Hall, Bloomington, IN 47405, kdejong@indiana.edu) and Yuka Tashiro (Linguist, Indiana Univ., Bloomington, IN)

There are several approaches to measuring spectral differences between fricatives, among which spectral moments analysis [Forrest *et al*., JASA 84, 115–123 (1988)] is the most common. To perceptually evaluate measurement systems, stimuli are constructed with four Gaussian noise components whose base frequency is systematically varied. Then, to vary the spectral shape, the amplitude of each component is independently varied to yield a five-dimensional space of synthetic fricatives. The stimuli were reduced to four dimensions and presented to listeners in a self-directed search protocol, in which participants located examples of /f, θ, s, ʃ, h/. Spectral moments were calculated from the stimuli, showing a complex relationship between the synthesis dimensions and spectral moments. This paper presents high-probability locations of these fricatives, regressed against the synthetic parameters and their associated spectral moments. Preliminary results indicate that acoustic measures of the chosen stimuli conform to differences found in previous fricative production studies, but also that the spectral moments measures do not outperform the synthetic parameters, and hence do not seem to be adding insight into which acoustic differences are guiding the listeners.

2pSC9. How twang vocal timbre impacts intelligibility in noise. Tzu Pei Tsai (Speech, Lang. and Hearing Sci., Indiana Univ., 2451 E 10th St., Apt 1014, Bloomington, IN 47408, tsaitz@iu.edu) and Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Twang vocal timbre is thought to be an efficient voice strategy that minimizes vocal effort while maximizing acoustic output. Therefore, the use of twang timbre may enhance communication success in noisy environments while preserving the voice. However, the intelligibility of twang speech has not been evaluated. This study investigated how vocal timbres (neutral and twang) influence intelligibility in common traffic (train- and plane-shaped) noises. Sixty-two participants identified words within a 50-word English sentence matrix. Sentences with twang and neutral timbres were synthesized from eight speakers (four female and four male) with deep-learning timbre conversion that controlled sound level, pitch, prosody, and accent. Stimuli were masked with white noise shaped with spectral contours from ambient train and plane noise at −18 and −24 dB SNRs. Intelligibility, determined by word identification accuracy, was significantly higher in twang speech than in neutral speech for both female and male speakers. The twang advantage might arise from higher perceived loudness, higher formant frequencies from F1 to F3, the clustering of F3 and F4, and/or the less breathy or hoarse voice quality in twang compared to neutral timbre. The result suggests that vocal timbre adjustment is a valid strategy for talkers to improve intelligibility in noise.

2pSC10. Individual differences in lexical recruitment across contrasts. Weiyi Zhai (Linguist, McGill Univ., 1085 Dr. Penfield Ave., Montreal, QC H3A1A7, Canada, weiyi.zhai@mail.mcgill.ca) and Meghan Clayards (Linguist, McGill Univ., Montreal, QC, Canada)

Lexical recruitment in perception can vary across individuals and tasks (Giovannone and Theodore, 2023). In this study, we explore whether it also varies across different acoustic contrasts. Using a Ganong Task for phoneme categorization, we examined how individuals rely on lexical information across four contrasts: /ɛ/—/æ/ (vary in F1—F2), /r/—/l/ (F3), /s/—/ʃ/ (fricative spectrum), and stops /d/—/t/ (VOT). Among 84 monolingual English listeners, at the individual level, lexical bias correlated positively across all contrasts, except for /ɛ/-/æ/ and /s/-/ʃ/, suggesting individuals' lexical bias is largely stable across contrasts, with some stimuli-specific effects. Additionally, individuals' reliance on the acoustic information (continuum step) showed positive correlations across these contrasts as well. Interestingly, we found positive correlations between the use of step and lexical bias for /s/-/ʃ/ and /r/-/l/, while negative correlations for /d/-/t/ and /ɛ/-/æ/. We posit that the strength of the lexical and acoustic effects may influence the direction of this correlation. Further analysis is needed to explore this relationship and its implications.

2pSC11. Speaker-independent vocal tract shapes that optimally elicit vowel recognition. Adam C. Lammert (Mathematics & Comput. Sci., College of the Holy Cross, 1 College St., Swords 331, Worcester, MA 01610, alammert@holycross.edu), Wesley Smith, Jane Gargano, Sarah Cornacchia (Mathematics & Comput. Sci., College of the Holy Cross, Worcester, MA), and Benjamin Parrell (Univ. of Wisconsin–Madison, Madison, WI)

When speakers shape their vocal tracts in specific ways, it elicits the recognition of certain phonemes in listeners. But, which vocal tract shapes, independent of any particular speaker, best explain the phoneme recognition behavior of listeners? Prior studies have associated recognition with simplified shaping actions—e.g., isolated, narrow constrictions. In this study, we use the method of Reverse Correlation (RC) to reconstruct unconstrained vocal tract area functions that best explain the phoneme recognition behavior. Here, we ask listeners to render yes–no recognition responses over a series of ambiguous stimuli derived from random area functions, which are then used to estimate an area function via RC regression analysis to optimally predict the listener responses. Using RC, optimal area functions were reconstructed for vowels /i/, /a/, and /u/ from native English speakers with normal hearing (4 collected, 30 planned). Current results suggest that these area functions accurately predicted listener responses (67%–90%, mean 78%). Furthermore, reconstructed area functions reveal specific localized constrictions of the vocal tract that contribute consistently to explaining responses across listeners, but which were different from constrictions exhibited by speakers producing those same vowels. Results hold implications for understanding the complex relationship between perception and production of speech.

2pSC12. Telepractice and effort during interactive conversations: Using EmbracePlus health monitor to study normal-hearing children. Maria V. Kondaurova (Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, maria.kondaurova@louisville.edu), Ruchik Mishra (Elec. & Comput. Eng., J.B. School of Eng., Univ. of Lousville, Louisville, KY), Loui Chang (Psychol. & Brain Sci., Univ. of Louisville, Louisville, KY), Alan F. Smith (Otolaryngology-Head/Neck Surgery & Communicative Disord., Speech-Lang. Pathol. Program, Univ. of Louisville, Louisville, KY), Irina Kondaurova, and Qi Zheng (Bioinformatics & Biostatistics, School of Public Health, Univ. of Louisville, Louisville, KY)

Wearable sensors are increasingly being used to evaluate physiological biomarkers associated with perceived effort in adverse listening conditions. The current study examined the feasibility of using EmbracePlus (E4), a wearable health monitor, to assess effort in normal-hearing children who interacted with a clinical provider during telepractice. Ten children (mean age = 10.4 years, age range 8–12 years)participated in four (two in-person, two tele) weekly visits, order counterbalanced. At each visit, the clinician asked the children to hold a standardized conversation using a Diapix picture task and provide subjective ratings of perceived effort. Measures of electrodermal activity and blood volume pulse amplitude were collected from the EmpracePlus wristband worn by each child. Children self-reported a significant increase in perceived effort during tele- compared to in-person sessions. Their effort, however, decreased over time. Based on preliminary data, it is predicted that physiological measures of effort will be consistent with child subjective ratings, although individual differences may affect the results. The study demonstrates the feasibility of using EmbracePlus to collect physiological biomarkers associated with perceived effort in normal-hearing children during remote compared to in-person communication. This may impact the telepractice delivery of speech-language intervention in the pediatric population with and without hearing loss.

2pSC13. How do classroom activities influence noise levels and student's listening effort? Julia Seitz (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52062, Germany, julia.seitz@akustik.rwth-aachen.de) and Janina Fels (Inst. for Hearing Technol. and Acoust. (IHTA), RWTH Aachen Univ., Aachen, Germany)

This study investigated the acoustic environment in primary classrooms specifically for different activities. Lessons were categorized into teacher-student interaction, silent work, group work, and classroom breaks. Noise measurements were conducted in seven German primary schools (4 first grades, 3 fourth grades) with 78 students, analyzing sound pressure level (SPL), A-weighted SPL, loudness, and sharpness across these activities. In addition, children's listening effort was assessed using a questionnaire. Results showed significant differences in A-weighted SPL across activities. Loudness was found to be the most sensitive parameter, with notable differences between activities and grades. The subjective listening effort did not vary significantly between activities, suggesting the questionnaire might not have been suitable for the age group. Results emphasized the importance of using perceptually meaningful parameters like psychoacoustic loudness for assessing classroom noise. [This research was funded by the European Union's Horizon 2020 Research and Innovation Program under grant agreement No. 874724 (Equal-Life)]

2pSC14. Effects of low-pass filtering and visual cue availability on English consonant-in-noise recognition for late bilinguals and monolinguals. Sofia Hall (Dept. of Psychol. Sci., Case Western Reserve Univ., Dept. of Commun. Sci., 11635 Euclid Ave., Cleveland, OH 44106, sch92@case.edu), Tiana Cowan (Ctr. for Childhood Deafness, Lang. and Learning, Boys Town National Res. Hospital, Omaha, NE), Emily Buss (Dept. of Otolaryngology/Head and Neck Surgery, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Lori Leibold, and Kaylah Lalonde (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Late bilingual listeners are typically more susceptible to noise when listening to speech in their second language than monolinguals, particularly with auditory-only input. Limited research has examined how auditory-only and visual cue availability influences second-language speech-in-noise recognition. This project examines whether low-pass filtering of auditory cues results in greater performance decrements for late Spanish/English bilinguals compared to English monolinguals, given their differing susceptibility to background noise, and how visual cue availability moderates this effect. Additionally, we investigate the extent to which late bilinguals and monolinguals benefit from audiovisual cue availability at different points on the psychometric function during English consonant recognition in noise. To address these questions, we focus on the effect of face masks, which both low-pass filter speech and occlude visual cues, offering an ecologically valid approach to examine these effects. We assess English consonant recognition in open syllables, such as "key," "she," and "me," in speech-shaped noise presented at four signal-to-noise ratios. Participants are tested under two acoustic filtering conditions (all-pass filter and low-pass filter) and two modality conditions (auditory-only and audiovisual). Findings and their implications will be discussed.

2pSC15. Creating Navajo synthetic speech with deep learning. Niah Nieuwenhuis (Commun. Sci. and Disord., Northern Arizona Univ., 208 E Pine Knoll Dr., Flagstaff, AZ 86011, nnn37@nau.edu) and Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ)

With new developments in machine learning and deep neural networks, speech synthesis technology has made it possible for almost anyone to have a voice, including those with severe speech impairments. However, there exists a mismatch between the technology's capabilities and the availability of naturalistic and personalized voices in a wide variety of languages, especially indigenous ones. To close this gap, this project seeks to create a natural-sounding speech synthesizer for an adult Navajo speaker with limited training data. A pilot speaker was recorded in a sound-attenuated booth. After removing the false starts, fillers, and other non-speech sounds, the speaker produced just over 2 h of data. The speech synthesis software Tacotron 2 and Waveglow were utilized to produce high-quality synthesized speech. To determine the naturalness of the synthetic speech, listeners participated in a mean opinion score survey and offered qualitative feedback. Our results highlight the feasibility of creating synthetic speech for other low-resource languages. We discuss the viability of exploring more cost-effective means of personalized speech synthesis, as opposed to existing methods for high-resource languages.

2pSC16. Nonlinearities of the auditory pathway constrain the cross-linguistic structure of vowel systems. Daniel D. Pyskaty (Departments of Brain and Cognit. Sci., Linguist, Univ. of Rochester, Rochester, NY 14627, dpyskaty@u.rochester.edu), Daniel R. Guest (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY), Douglas M. Schwarz (Dept. of Neurosci., Univ. of Rochester, Rochester, NY), Joyce M. McDonough (Dept. of Linguist, Univ. of Rochester, Rochester, NY), and Laurel H. Carney (Departments of Biomedical Eng., Neurosci., Univ. of Rochester, Rochester, NY)

Vowel contrasts tend to build in similar ways cross-linguistically, dispersing themselves within an acoustic space defined by the first two formants, F1 (height) and F2 (backness). While vowel dispersion models successfully predict vowel contrasts in the F1 dimension, they fail to predict that far fewer vowels occur along F2. Work since the 1970s has defaulted to the auditory system to explain this asymmetry. We hypothesize that in the ascending subcortical auditory pathway, effects of nonlinearities on the representation of formants constrain which vowels may occur. We constructed ecologically valid vowel systems using synthetic vowels with added speech-shaped noise. We simulated neural responses to the synthetic vowels using a physiological model and used a linear support vector machine to classify the responses into vowel categories. Preliminary results show that F1 has a more robust regional coding scheme in noise than F2's simpler representation. F1 supports more contrasts in comparison to F2, which is more vulnerable to noise. These results suggest that auditory peripheral tuning and nonlinearities, such as saturation, compression, and suppression, constrain the structure of vowel systems and the encoding of vowel spectra. [Work supported by NIDCD-R01-010813.]

2pSC17. Acoustic characteristics associated with speech naturalness and listening effort of dysarthric speech. Elizabeth Krajewski (Commun. Sci. and Disord., Penn State Univ., 308 Ford Bldg., University Park, PA 16802, ezr16@psu.edu), Jimin Lee, Anne J. Olmstead, and Navin Viswanathan (Commun. Sci. and Disord., Penn State Univ., University Park, PA)

The aim of the current study was to determine the acoustic characteristics that contribute to speech naturalness and listening effort of dysarthric speech. Listeners heard words produced by speakers with dysarthria and rated either their speech naturalness or listening effort. They also identified the productions, which provided response time as a behavioral measure of listening effort. We examined the associations between the perceptual domains and the following vowel metrics for each production: formants 1 and 2 (F1 and F2), vowel duration, mean fundamental frequency (F0), and cepstral peak prominence (CPPS). Preliminary analyses were conducted using linear mixed-effects models to determine the effects of each acoustic variable on speech naturalness ratings, perceived listening effort, and response time. Listeners and speakers were included as random intercepts. The results indicated that only vowel duration was significantly associated with speech naturalness ratings. In contrast, vowel duration, F1, F2, and CPPS were significantly associated with both perceived listening effort and response time. Only mean F0 was not associated with listening effort. These results enhance our understanding of how dysarthria affects speech production and perception by using multiple productions from each speaker, allowing us to investigate within-speaker acoustic characteristics and their impact on perception.

2pSC18. Assessing the effect of pupillometric fatigue on task-evoked responses in a listening effort paradigm. Brady M. Chisholm (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, chish071@umn.edu) and Juraj Mesik (Psych., Univ. of Minnesota, Minneapolis, MN)

Measurement of task-evoked pupillary responses (TEPR) is widely used for quantifying the cognitive status of participants (e.g., listening effort; LE) in hearing sciences. A key methodological challenge of pupillometry is that TEPR weakens over the course of the experimental session, raising questions of whether such fatigue influences participant response to experimental manipulations in later trials. To address this issue, we measured TEPR in a

sentence recognition task where participants listened to, and repeated back, sentences in the presence of a distracting same-sex talker. To manipulate LE, each of the 200 trials was presented at one of 4 signal-to-noise ratios (SNR; -10 to $+5$ dB), varying pseudo-randomly on a trial-by-trial basis. Linear mixed-effects modeling of TEPR was used to assess whether chronological points in the experiment interacted with the effect of SNR. Preliminary analyses of peak TEPR suggest that despite substantial accrual of pupillometric fatigue over the experimental session, fatigue effects did not significantly interact with SNR, indicating that effects of LE manipulations on TEPR remain comparable across different levels of pupillometric fatigue. These results indicate TEPR may be a suitable measure of LE even in relatively long-lasting (e.g., $\sim$2 h) paradigms. [Research supported by NIH grant R21DC020788 and UMN's UROP program.]

2pSC19. An eye-tracking study of the interaction of text and facial cues in speech comprehension. Kayla Aikins (Univ. of Washington, 1664 N. Virginia St., Reno, NV 89557, kaikins@med.unr.edu), Ella De Falco, Richard A. Wright (Linguist, Univ. of Washington, Seattle, WA), and Gavriel D. Kohlberg (Otolarynology—Head and Neck Surgery, Univ. of Washington, Seattle, WA)

Effective speech comprehension in noisy environments depends on integrating auditory and visual cues. This study examined how signal-to-noise ratio (SNR), text accuracy, and facial cues influence fixation behaviors during audiovisual speech comprehension. Twenty-four participants completed trials combining three SNR levels (-4, -5, and -7 dB), with three text accuracy conditions (accurate, two words missing, and three words missing), and two video conditions (facial cues present or absent). The result was the log-transformed fixation duration ratio of the text dwell time to face dwell time. The result was submitted to a linear mixed-effects model which included fixed effects for signal-to-noise ratio, text accuracy, and video display, and random effects for participants and sentences. Text accuracy ($\chi^2 = 225.35$, $p < 0.001$) and video presence ($\chi^2 = 397.37$, $p < 0.001$) significantly influenced fixation behaviors. Poor text accuracy reduced text dwell time, particularly with three missing words (Estimate $= -3.753$, $p < 0.001$). Video presence decreased text fixation, favoring facial cues (Estimate $= -4.559$, $p < 0.001$). Under a poor signal-to-noise ratio, low text accuracy amplified reliance on facial cues ($\chi^2 = 14.50$, $p = 0.0007$). Therefore, poor text accuracy diminishes its utility, prompting compensatory reliance on facial cues for speech comprehension.

2pSC20. Relationship between self-perceived and formal assessment of speech intelligibility, and association to noise exposure history. Leny Vinceslas (Comput. Sci., Univ. College London, 169 Euston Rd., London NW1 2AE, United Kingdom, l.vinceslas@ucl.ac.uk), Vit Drga (Comput. Sci., Univ. College London, London, United Kingdom), Jesko Verhey (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Magdeburg, Sachsen-Anhalt, Germany), and Ifat Yasin (Comput. Sci., Univ. College London, London, United Kingdom)

Over a lifetime, the effects of noise exposure can be additive, contributing to impaired performance on speech-in-noise assessments and raised hearing thresholds. However, in some cases, such as the early stages of cochlear synaptopathy, the effects of noise exposure may not be so evident in formal auditory assessments. In such cases, perceived speech difficulties in noise may not always correlate with the formal evaluation of speech difficulties. A structured interview approach to eliciting noise exposure history may provide a more reliable estimate of an individual's difficulties in perceiving speech in noise. The current study investigated the relationship between individuals' self-reported difficulties with speech perception in noise, performance on a formal speech-in-noise test (Matrix test), and past noise exposure using a structured interview approach. The Matrix test data was fitted with psychometric functions denoting speech intelligibility in noise. Intelligibility parameters (threshold, and slope) were compared to self-reported evaluation of speech listening in noise and noise-exposure history. Results indicate a complex relationship between self-reported difficulties with speech perception in noise, recall of historical noise exposure, and speech in noise intelligibility parameters. The findings are discussed with reference to procedures for estimating historical noise exposure, adaptive speech testing, and key speech intelligibility parameters.

2pSC21. The impact of effortful listening on comprehension of suprathreshold broadcast dialogue during continuous, naturalistic listening. William D. Curry (Acoust. Res. Ctr., Univ. of Salford, 43 Crescent, Salford, Greater Manchester M5 4WT, United Kingdom, w.d.curry@edu.salford.ac.uk), Ben G. Shirley, and Trevor J. Cox (Acoust. Res. Ctr., Univ. of Salford, Salford, Greater Manchester, United Kingdom)

Broadcast audio accessibility has primarily focused on speech intelligibility, with word recognition metrics among the most common means of assessment. Listening in adverse conditions for an extended period can impact language processing and memory encoding, even when word recognition is perfect. Recent studies have included listening effort (LE) as a metric in accessibility assessment, however, the impact of LE on comprehension of spoken content is not known. The present study presented participants with excerpts of continuous speech in the presence of background noise, at speech-to-noise ratios (SNRs) representative of common broadcast practices and guidelines. Participants reported LE on scales adapted from the NASA-TLX, and completed a recognition task designed to probe for three levels of representation (surface form, propositional, and event model). Electroencephalogram (EEG) recordings were taken, and temporal response functions (TRFs) were estimated to measure the effects of acoustic challenge on cortical tracking of acoustic and linguistic features of the stimulus. Findings show that SNRs common in broadcasts introduce significant LE, and that increased demand impedes the construction of mental representations needed for comprehension and information retention. Further, results are presented from a TRF component analysis, relating response data to different stages of acoustic and linguistic processing. Participating investigators Martin Walsh & Ted Laverty served as advisors for the work. [The work was supported by Xperi Inc.]

2pSC22. Patterns of phoneme discrimination difficulty for adults with Down syndrome. Anne J. Olmstead (Commun. Sci. and Disord., The Penn State Univ., 404D Ford Bldg., University Park, PA 16802, ajo150@psu.edu), Jimin Lee, Carolynn Buckley, Emma Ferry, Maegan Mapes, Leslie Purcell, and Krista Wilkinson (Commun. Sci. and Disord., The Penn State Univ., University Park, PA)

Individuals with Down syndrome (DS) often experience disruption to speech and language processing. However, there is a paucity of research examining the specific characteristics of speech production and perception for this population. As a result, especially for adults with DS, very little is known about even fundamental characteristics of speech perception. Past research (Keller-Bell and Fox, 2007) has suggested that children with DS may have difficulty discriminating certain speech sounds. The current study examines discrimination of phoneme minimal pairs by adults with DS as it relates to hearing status and speech production. Adults with DS performed an AX discrimination task with stimuli differing on one, two, or three phonetic features and vowel minimal pairs. Accuracies in the AX task suggest that minimal pair discrimination varies by phonetic features as well as by participant. In particular, consonant place distinctions may be particularly difficult for individuals with DS but voicing and manner distinctions pose less difficulty. Vowel discrimination varies by participant with some having significant difficulty with vowel discrimination and others being at the ceiling. These results are considered in the context of speech production and hearing status.

2pSC23. Implementing a "Cue-based Features" approach to the phonetics/phonology interface. Alexandra Pfiffner (Linguist, UC Berkeley, Berkeley, CA, apfiffner@berkeley.edu) and Keith Johnson (Linguist, UC Berkeley, Berkeley, CA)

Models of the phonetics/phonology interface have typically dealt with phonetic detail in a quite idealized way, for instance by placing exemplars in a one or two-dimensional idealized phonetic cue space. This paper explores a more phonetically realistic approach to modeling phonological cues building on a "Cue-based Features" account of perceptual cue trading and the interface between phonetic cues and phonological features. We train non-negative matrix factorization over large speech corpora to discover spectrotemporal patterns that are potential acoustic cues for phonological contrast, and then we use linear discriminant analysis to find combinations of cues that can be optimally used to specify the value of a phonological

feature. We present a preliminary study of plosive voicing cues, and the resulting models reveal that the cues trade with each other across exemplars and that they encode information that is temporally spread beyond the segment's acoustic boundaries.

2pSC24. Comprehension and recognition memory for spectrally degraded sentences. Terrin N. Tamati (Otolaryngol., Vanderbilt Univ. Medical Ctr., 1608 Aschinger Blvd, Columbus, OH 43212, terrintamati@gmail.com) and Victoria A. Sevich (Ohio State Univ., Columbus, OH)

Everyday communication requires not only the ability to recognize speech but also the ability to comprehend and remember what has been said. However, when the speech signal is degraded, as for cochlear implants (CIs), additional cognitive effort may be required. Other challenges, such as speaking rate alterations, may further strain the listener's limited cognitive resources, with implications for comprehension and memory. The current study examined the effects of spectral degradation and speaking rate on comprehension and recognition memory in adult CI users and normal-hearing (NH) listeners under CI simulation. Participants completed a sentence verification task in which they classified sentences as true or false, and a sentence recognition memory task (SRMT) in which they classified sentences as new or old (from the SVT). Sentences were altered to have fast, average, and slow speaking rates. Results demonstrated that both CI and NH listeners achieved highly accurate sentence comprehension and recognition memory, emphasizing the robustness of speech processing to spectral degradation. However, slower speaking rates enhanced comprehension and memory, particularly for CI users. Faster speaking rates may exacerbate the cognitive demands imposed by spectral degradation, hindering speech communication. Comprehension and memory tasks may provide valuable insights into CI users' real-world listening challenges.

2pSC25. Neural responses to human versus AI interlocutors: An MEG study. Maggie Clarke (ImageTech Lab, Simon Fraser Univ., Burnaby, BC, Canada, mdclarke@sfu.ca), Jetic Gu (School of Computing Sci., Simon Fraser Univ., Burnaby, BC, Canada), Meagan Durana, Han Zhang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada), Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS), Joan Sereno (Linguist, Univ. of Kansas, Kansas City, KS), and Yue Wang (Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

Human interactions with AI-powered vocal interfaces are widespread. Since AI-generated speech is often more acoustically ambiguous than human speech, a question arises: What cues could humans rely on to more effectively process AI speech? Using magnetoencephalography (MEG), we investigate how native English perceivers' brain responses to AI (synthetic) versus human speech differ as a function of segmental contrast and semantic context. The stimuli include target words involving a fricative contrast, preceded by either a congruous or an incongruous semantic context (e.g., "*A triplet is made of three*." versus "*A triplet is made of free*."). Analyses involve event-related field components associated with phonetic processing (M100) and semantic anomaly detection (M400), focusing on brain responses time-locked to the target word. By comparing the two semantic congruency conditions, we predict that, for human speech, sensitivity to the fricative contrast will trigger sensitivity to the semantic anomaly, indexed by M100 and M400 effects. However, processing of AI speech with ambiguous fricatives may result in more similar responses to both congruency conditions, with a lack of or reduced M100 and M400. This suggests that when faced with acoustic ambiguity in AI speech, perceivers might depend more on contextual information to make sense of the speech.

2pSC26. The impact of noise and dysphonic speech on children's listening comprehension and cognitive effort: Insights from subjective, behavioral and EEG measures. Silvia Murgia (Dept. of Psych., Univ. of Illinois at Urbana-Champaign, 603 E Daniel St., Champaign, IL 61820, smurgia2@illinois.edu), Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

This study examines the impact of background noise and voice quality on sentence comprehension and listening effort in children, using a combination of self-assessment, response times (RT), and electroencephalography (EEG). Inhibitory control (IC) and working memory (WM) were assessed to account for individual differences. Twenty-four children (8–12 years) completed the Test for Reception of Grammar under two signal-to-noise ratios (SNRs; 0 and 6 dB) and with two voice qualities (typical and dysphonic). EEG analysis centered on event-related spectral perturbations in the theta and alpha bands, offering physiological insights into the cognitive demands. The results demonstrated that comprehension significantly decreased in the presence of higher noise levels and dysphonic speech. Perceived effort was higher in noisier and dysphonic conditions, with greater WM linked to increased reported effort. Dysphonic speech also led to longer RT, although children with stronger IC responded faster. Finally, listening to dysphonic speech was associated with increased theta activity and decreased alpha activity, with WM enhancing theta power and IC affecting alpha power, reflecting the cognitive demands of challenging auditory conditions. These findings reveal that noise and dysphonic speech uniquely tax cognitive resources, impairing comprehension and effort while drawing on executive function to adapt to challenging acoustic conditions.

2pSC27. Exploring the impact of individual differences and SLP experience on accent perception. Priya Kearney (Commun. Sci. and Disord., Univ. of Iowa, Iowa city, IA), John B. Muegge (Linguist, Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), and Ethan Kutlu (Linguist, Psychol. and Brain Sci., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com)

Spoken language exhibits considerable variability, with factors such as age and linguistic experience contributing to changes in speech patterns. Despite advancements in speech perception theories, there is a gap in understanding this variability from an applied perspective, particularly in Speech-Language Pathology (SLP). The current approach in SLP tends to treat voices as uniform acoustic signals, overlooking ecological variation in favor of identifying pathological markers. This study aims to explore how SLP training influences speech-language pathologists' sensitivity to ecological speech variability and whether this variability is mistakenly perceived as pathological. We tested 30 SLP trainees, measuring their ability to understand unfamiliar accents and individual differences in speech categorization (through a Visual Analogue Scaling Task). We find that listeners who are less consistent in their ability to categorize the same speech sound are worse at understanding unfamiliar accents. We argue that SLP trainees benefit from more diverse training. Future work aims to test SLP clinicians to examine whether further training also impacts accent perception.

2pSC28. Gradiency in speech categorization and its relation to speech intelligibility. Ethan Kutlu (Linguist, Psychol. and Brain Sci., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com)

Understanding how listeners categorize highly variable speech signals into discrete units is a central question in speech perception research. Traditionally, it was assumed that listeners disregard variation and focus solely on the category itself, a concept known as categorical perception (Liberman *et al.*, 1957). However, when faced with highly variable speech signals, such as unfamiliar accents, ignoring variation can hinder comprehension and increase processing difficulty. In this ongoing online study (expected $n = 120$), we investigate English-speaking adult listeners' speech categorization patterns using a continuous measure—the Visual Analogue Scaling Task—to examine their sensitivity to spoken language variation (Kutlu *et al.*, 2022; Apfelbaum *et al.*, 2022). Participants' exposure to varied accents is quantified through an extensive social network questionnaire. This questionnaire captures the extent and diversity of accents they regularly encounter (Kutlu *et al.*, 2024). Subsequently, participants listen to sentences spoken in a variety of unfamiliar accents and transcribe them. Sentence intelligibility is assessed by scoring the accuracy of these transcriptions. Preliminary results show that sound categorization consistency predicts novel accent perception such that listeners who are more consistent in their sound categorization are more accurate in transcribing novel accents. The implications for exposure to diversity will be discussed.

2pSC29. Examining the consistency of spectral context effects across frequency regions in speech perception. Christian E. Stilp (Speech Pathol. and Audiol., Marquette Univ., 317 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@marquette.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Speech categorization is influenced by spectral contrast effects, or the perceptual magnification of spectral differences between successive sounds. Spectral contrast effects result in the categorization of a target sound being biased away from spectral properties in the preceding acoustic context. Because of the remarkable consistency in the magnitudes of these contrast effects within the same frequency region, we hypothesized that they would also show a stable relationship across different frequency regions. In this study, normal-hearing listeners' phoneme categorization and contrast effects were assessed where phonetic contrasts were driven by changes in low-frequency F1 ("big"-"beg"), mid-frequency F3 ("dot"-"got"), or high-frequency regions (frication spectrum in "sheet"-"seat"). On each trial, listeners heard a precursor sentence that was filtered to emphasize energy in the lower or higher range for one of these frequency regions, followed by a target word that hinged on that filtered frequency region. Spectral contrast effects influenced categorization in each frequency region, as expected. However, effect magnitudes were not reliably correlated with each other across frequency regions. This further clarifies perception-in-context on a broader scale, as using spectral context during speech categorization may be consistent within a single frequency region but not across different frequency regions.

2pSC30. Acoustic variability shapes adaptation to single versus multiple sound sources. Anya E. Shorey, Kate Criner (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), Sung-Joo Lim (Psych., Binghamton Univ., Binghamton, NY), Julia Dobbs, Kristin Fuerstenberg (Speech Pathol. and Audiol., Marquette Univ., Milwaukee, WI), and Christian E. Stilp (Speech Pathol. and Audiol., Marquette Univ., 317 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@marquette.edu)

Many studies have demonstrated superior recognition of words spoken by a single talker compared to multiple talkers. Recently, Shorey *et al.* (2023 *AP&P*) reported parallel results when labeling words from talkers and tones from musical instruments. Here, we further examined the effects of stimulus variability in both speech and music perception. Experiment 1 adopted the paradigm of Stilp and Theodore (2020 *AP&P*) with multiple mixed-talker blocks where voices had Low Variability (similar mean f0s) or High Variability (dissimilar mean f0s); instruments with Low Variability (similar attacks and spectra) or High Variability (dissimilar attacks and spectra) were also presented. Music perception was slower and less accurate at each successive increase in stimulus variability. However, the effects of increased stimulus variability were less pronounced in speech blocks than in music blocks. Experiment 2 used this paradigm to measure vowel identification spoken by multiple talkers with dissimilar mean f0s saying the same two target words (heed and hoed, Low Variability) or eight different target words that all shared the /i/ or /o/ target vowel (High Variability). Speech responses became slower at each successive increase in stimulus variability, paralleling music blocks. Thus, adaptation to sound sources (formerly "talker normalization") displays domain-general responses to stimulus acoustic variability.

2pSC31. Vocal emotion recognition in noise: The role of intonation perception. Benjamin T. Amartey (Commun. Sci. and Disord., Northwestern Univ., Frances Searle Bldg., 2240 Campus Dr., Evanston, IL 60201, benjaminamartey2027@northwestern.edu), Kendra Marks (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Monita Chatterjee (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Previous research has established a relationship between intonation perception (involving relatively simple changes in fundamental frequency) and vocal emotion recognition (involving complex changes in fundamental frequency) in older listeners. However, typical communication environments for older listeners are often acoustically challenging, raising questions about how this relationship holds under noisy conditions and whether objective performance on vocal emotion aligns with self-reported experiences in real-world settings. We hypothesize that strong intonation perception supports accurate vocal emotion recognition, even in the presence of background noise. In two experiments, older listeners completed intonation perception (in quiet) and vocal emotion recognition tasks at $+10$ and $+3$ dB SNRs, with six-talker babble maskers spatially separated (90°, 180°, and 270°) from the target. Listeners also completed an emotion questionnaire (EMO-CHeQ) to examine the relationship between self-reported and objective vocal emotion recognition. Findings suggest that listeners with better intonation perception demonstrated better vocal emotion recognition, regardless of SNR levels. Furthermore, listeners reporting greater difficulty on the EMO-CHeQ had lower accuracy in objective vocal emotion recognition, indicating a relationship between self-reported challenges and measured emotional recognition. These findings suggest that individuals with strong intonation tracking will be better at recognizing vocal emotion, even in background noise.

2pSC32. Exploring effects of speaker and face mask type on speech intelligibility. Paul J. Smith (Dept. of Biobehavioral Sci., Teachers College, Columbia Univ., New York, NY 10027, pjs2194@tc.columbia.edu), Laura Koenig (Haskins Labs, New Haven, NY), Melissa Randazzo (Evidence in Motion, San Antonio, TX), and Ryan Priefer (Magstim EGI, Roseville, MN)

Subsequent to the COVID-19 pandemic, face masks remain a consistent part of many individuals' lives. While research on face masking has revealed effects on speech intelligibility, relatively few studies have examined speaker differences, the effects of masking on speaker behavior, and how these influence intelligibility. We report here on data from four healthy American English-speaking adults who produced low-predictability sentences in four face mask conditions: surgical, cloth, N95, and no mask. Typically hearing listeners ($n = 89$) heard sentences, mixed with multi-talker babble, from one of the speakers and typed the words they heard. Percent words correct was scored by a text-matching algorithm, allowing for morphological errors and homonyms. From all speakers, we obtained: long-term average spectra; first and second formant ranges in words with /i, ɑ, o/; sentence durations and average intensities; and phonetic features of common misperceptions. Error rates were highest for the N95 mask. Speakers demonstrated individual patterns of the above acoustic variables. The speaker with the lowest intelligibility displayed faster speech, lower intensity, and unexpectedly higher F2 ranges in the masked conditions. These results suggest that quantifying speaker differences in intelligibility and acoustics can better elucidate the effects of face masks on communication.

2pSC33. Is speech categorization consistency achieved in part through lexical knowledge? Hyoju Kim (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA, hyoju-kim@uiowa.edu), Ethan Kutlu, Sita Carraturo, John B. Muegge, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Speech categorization is a gateway for downstream language processes. Recent evidence from work with the Visual Analog Scaling (VAS) task (Kapnoula *et al.*, 2017) underscores the importance of categorization consistency (trial-by-trial variability around the mean function) over slope (long-term category structure) as a critical predictor of real-world outcomes such as L1 language ability and L2 language learning (Kim *et al.*, under review; Honda *et al.*, 2024). However, the mechanisms that contribute to categorization consistency are unclear. Given that higher-level factors like the lexicon can stabilize the percept by cleaning up lower-level perceptual noise (Luthra *et al.*, 2021), we examined the relationship between categorization consistency and the lexical status of the percept. We tested adult American English listeners ($n = 48$, data collection ongoing) with the two sets of VAS tasks: one with word continua (e.g., *beach-peach*) and the other with nonword continua (e.g., *beag-peag*). Preliminary results showed that listeners' categorization consistency for the word continuum is significantly higher than for the nonword continuum. However, we did not find a significant difference in slope. This suggests that categorization consistency (but not slope) can be stabilized by listeners' lexical knowledge, supporting "auditory/phonological clean-up" driven by lexical chunks.

2pSC34. Extending the buffer: Real-time cue integration requires encapsulated auditory memory, even at absurdly long durations. Hyoju Kim (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA, hyoju-kim@uiowa.edu), John B. Muegge, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Listeners face a critical challenge in speech perception: acoustic cues to a given speech sound often unfold asynchronously. Traditional work suggests listeners solve this problem by processing each cue immediately and continuously to update higher-level interpretations. However, recent findings suggest that certain speech sounds (e.g., voiceless sibilant fricatives) may be *buffered*—judgment about the identity of the fricative is withheld until the vocoid arrives. It is unclear what triggers the release of this buffer: is it driven by specific incoming information or fixed cue duration? Using the visual-world paradigm, we tested this by extending the durations of /s/ and /ʃ/ to double (∼300 ms) and triple (∼450 ms) their typical lengths (∼150 ms). Consistent with previous findings, listeners withheld processing typical-length fricatives until frication offset. For double-length fricatives, listeners extended the buffer, waiting for the entire fricative duration before committing to a decision. When fricative lengths were tripled, partial commitment emerged during the frication period but final decisions still awaited additional cues. This suggests that buffering is largely based on the arrival of the vocoid, though at extreme durations listeners may access partial information. Thus, listeners flexibly delay processing as they wait to integrate current acoustic information with upcoming cues.

2pSC35. Utilizing extended high frequencies for fricative identification. Viktor Kharlamov (Florida Atlantic Univ., 777 Glades Rd., CU-97, Ste. 280, Boca Raton, FL 33431, vkharlamov@fau.edu), Daniel Brenner (Alameda, CA), and Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., Flagstaff, AZ)

This study investigates whether fricatives in conversational speech can be accurately identified using random forest modeling based solely on extended high frequencies (EHFs), defined as frequency information above 8 kHz. Previous research suggests that EHFs enhance speech intelligibility and clarity, particularly in noisy environments, and contribute to sound localization and speaker identification in conversational contexts. However, their role in distinguishing specific speech sounds remains unclear. To address this, we analyze high-pass filtered recordings of sociolinguistic interviews in Western Canadian English, focusing on whether EHFs provide sufficient acoustic cues to differentiate fricative consonants in a random forest classification model.

2pSC36. Phonetic variability leads to gradient speech perception. Ege Gur (Psychol. and Brain Sci., Univ. of Iowa, 340 Iowa Ave., Iowa City, IA 52242, ege-gur@uiowa.edu) and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Listeners show sensitivity to within-category variation in speech sounds. This *gradient* categorization system helps promote flexibility, deal with ambiguity, and maintain plasticity. Prior work has raised the possibility that statistical learning mechanisms might be involved showing that listeners adopt a more gradient categorization when exposed to greater phonetic variability (Clayards *et al*., 2008; Theodore and Monto, 2019). However, this work relied on a 2AFC task in which a shallower slope could derive from either difference in gradiency or differences in noise in the system. We thus reinvestigated this issue across three experiments. Participants were trained on distributions of voice onset time with either high or low variance. Later, gradiency was assessed with the VAS task, which overcomes these limits. Experiment 1 ($n = 84$) did not support the hypothesis that gradiency is an adaptation to increased variance. Instead, higher variance increased trial-by-trial inconsistency. Experiment 2 ($n = 168$) utilized 28 continua to better generalize across stimulus characteristics and found robust evidence for increased gradiency as a result of variability. Experiment 3 ($n = 85$) introduced a baseline condition to rule out the alternative explanation that listeners were adapting to low variance by becoming *less* gradient, instead of becoming *more* gradient in the face of high variance.

2pSC37. Lexical status influences response variability in older adults' speech perception. Samarium N. Knight (Psychol. and Brain Sci., Univ. of Iowa, 340 Iowa Ave., Rm. G60, Iowa City, IA 52242, samknight@uiowa.edu), Hyoju Kim, Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), and Sarah E. Colby (Linguist, Univ. of Ottawa, Ottawa, ON, Canada)

Speech sound categorization is a building block of downstream language processes, however, speech categorization can itself be influenced by lexical knowledge: Speakers tend to resolve ambiguous speech sounds in favor of real-word rather than non-word interpretations (Ganong,1980). Some studies have shown that older adults exhibit an increased lexical bias (e.g., Mattys and Scharenborg, 2014). However, previous work establishing this link has relied on an experimental paradigm in which participants are forced to make a binary choice, a task that may be overly sensitive to higher-level strategies and can obscure the distinction between responses that are highly variable and highly gradient (Apfelbaum *et al*., 2022). We thus asked if the link between aging and lexical bias persists in a more sensitive paradigm, in which participants indicated the extent to which a stimulus matches one of two categories using a Visual Analogue Scale (VAS). We performed an online study ($n = 60$) in which older and younger adults responded to a nine-step continua between either two words, or a word and a non-word (à la Ganong, 1980). Preliminary results suggest that while older adults are not necessarily more lexically biased generally, those with increased response variability do show a stronger lexical bias.

2pSC38. Neurophysiologic processing of Ling-6 speech sounds. Reethee Antony (Div. of Speech and Lang. Pathol., Binghamton Univ., 10 Gannett Dr., Johnson City, NY 13790, rantony@gradcenter.cuny.edu), Emma Schaedler, and Erica Scheinberg (Speech Lang. Pathol., Misericordia Univ., Dallas, PA)

The Ling-6 speech sounds have been widely in the fields of audiology and speech-language pathology. However, there is little knowledge of the neurophysiologic processing of speech sounds at a cortical level, hence the need for this study. The aim of this study was to examine the neurophysiologic processing of Ling-6 sounds. Sixteen adults were recruited for the study. The stimuli included natural digitized speech stimuli of the Ling-6 phonemes: /a/, /i/, /u/, /s/, /ʃ/, /m/. Electroencephalography was recorded in a sound attenuated booth and a passive paradigm was used. The responses were processed, analyzed, and the brain responses specifically, amplitudes and latencies of P1, N1, and P2 responses were measured. Descriptive statistics and inferential statistics were performed, specifically, analysis of variance (ANOVA) was used. The N1 latencies were shorter for /s/, P2 latencies were longer for /ʃ/ relative to the other sonorants. In terms of amplitudes, /u/ had the lowest P1 amplitudes and /m/ had the highest amplitude. N1 amplitudes were significantly lesser ($p < 0.05$) for vowels relative to fricatives. Results from this research have clinical implications and allow for future research in electrophysiology toward aural rehabilitation.

2pSC39. Articulation and executive function: Exploring the bilingual advantage. Imani Lyte (Kingsborough Community College, 2001 Oriental Blvd, Brooklyn, NY 11235, IMANI.LYTE47@students.kbcc.cuny.edu), Cinty Chang Wu (Speech Commun., Kingsborough Community College, Brooklyn, NY), Anastasiia Myslyk (Touro Univ., Brooklyn, NY), Laura Muscalu (Ithaca College, Ithaca, NY), and Laura Spinu (Communications & Performing Arts, CUNY Kingsborough, Brooklyn, NY)

In a pilot study, we investigated the relationship between articulatory skills and higher-level executive function in monolinguals and bilinguals. Participants completed a tongue-twister task to assess articulatory skills and a Simon Task to measure conflict resolution. While monolinguals exhibited slightly higher articulatory accuracy, bilinguals demonstrated a small advantage in reaction times for the Simon Task, particularly in inhibiting responses, but performance was close to 100% across the board. Contrary to expectations, we found no significant correlation was found between articulatory skill and executive function. Our findings partially replicate previous research on bilingual advantages in executive function while suggesting that the relationship between articulation and executive function may be more

complex than initially hypothesized. Limitations, such as small sample size and potential methodological factors, are discussed. To further explore this and the potential effects of task complexity, we are currently replicating this study with higher numbers of participants and with a more difficult auditory switch task in which participants switch from making decisions regarding auditory stimuli [whether (a) they are presented in English or a different language and (b) they start with a nasal sound or not]. Our findings help us gain more insight into the mechanism underlying the bilingual advantage.

2pSC40. Real-time word recognition in challenging listening conditions is defined by two primary dimensions, each of which exhibit within-subject stability and link to outcomes. John B. Muegge (Univ. of Iowa, G60 Psychol. and Brain Sci. Bldg., Iowa City, IA 52242, john-muegge@uiowa.edu) and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

All listeners must contend with the fact that speech is often heard in sub-optimal, challenging conditions. While much is known about the compensatory auditory mechanisms listeners might use in these contexts, these may not be universal. Spoken word recognition offers an intriguing locus for context-dependent adaptation that could apply to all challenging conditions; indeed, research has found that spoken word recognition changes in these contexts, but what these changes represent and whether they have meaningful effects on outcomes was unknown. Here, we build on recent work by applying a Principal Component Analysis approach to eye-tracking data, demonstrating that word recognition primarily varies along two dimensions across different types of challenging listening conditions (noise and vocoding). Furthermore, we show that individual listeners are fairly consistent in how they configure word recognition across challenge types, and that these configurations are predicted by domain-general factors. Finally, we find that these changes to word recognition offer indirect benefits for performance outcomes, but only for a subset of listeners. Together, this work provides insight into how word recognition operates in and compensates for challenging contexts.

2pSC41. Measuring conversational dynamics to evaluate the effects of speech-perception defecits. Eric M. Johnson (Div. of Commun. Sci. and Disord., West Virginia Univ., 375 Birch St., Morgantown, WV 26506, eric.johnson5@hsc.wvu.edu)

Speech perception is essential for successful communication through spoken language, which often involves both speaking and listening by two or more conversation partners. The speech perception abilities of one conversation partner affect the dynamics of the entire conversation, not just the passive speech-recognition performance of that individual. However, speech perception is often evaluated through single-person listening tasks in controlled settings, such as an audiometric booth. While this type of testing can provide insight into an individual's ability to hear and understand speech passively, it is not representative of typical real-world conversations, and it fails to capture the effects of one individual's speech perception abilities on other conversational participants. This pilot study begins the development and validation of a measure of conversational dynamics for the purpose of evaluating hearing interventions such as hearing aids. Five pairs of conversation partners participated in two experiments. The first involved a series of semi-structured conversation tasks in a controlled laboratory environment. The second involved a naturally occurring conversation in an uncontrolled environment. In both experiments, one conversation partner wore earplugs during half of the experimental trials. Objective and subjective measures of conversational dynamics revealed that the semi-structured laboratory task could serve as an ecologically valid proxy for real-world communication.

2pSC42. The T-complex responses to lexical tone in bilingual children with Mandarin-English and Spanish-English backgrounds. Victoria Billack (Saddle Brook Middle/High School, Saddle Brook, NJ), Rigel Baron (Neurosci., St. John's Univ., Queens, NY), and Yan H. Yu (Neurosci., St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Individuals with various developmental disorders, including auditory processing disorders, often show abnormalities in cortical auditory evoked potentials (CAEPs) at temporal sites. Temporal CAEPs, or the T-complex, have been studied in monolingual and bilingual children with non-tonal languages, but the developmental trajectory of the T-complex in bilingual children with tonal languages, like Mandarin-English, remains largely unexplored. This study aims to examine neurodevelopmental changes in Mandarin lexical tone processing, reflected in the T-complex, in children from Mandarin-English and Spanish-English backgrounds. We also examined how different bilingual experience influences the T-complex responses. We used a multiple oddball paradigm and recorded event-related potentials from bilingual Mandarin-English adults and children aged 5–10 from Mandarin-English and Spanish-English households, employing 65 sensors. Tone 1 (high-level tone) and Tone 2 (low-rising tone) were the deviant/infrequent tones, while Tone 3 (low dipping tone) served as the standard/frequent tones. T-complex was analyzed using the standard tone (Tone 3). Our results show that bilingual language experience and age-related brain maturation influence the T-complex. In children 5–10 years old, both lexical tone exposure and increasing age led to more distinct T-complex formation, suggesting that the developmental trajectory of the T-complex is prolonged.

2pSC43. Gaussian-mixture-model-based spectral cue detection for modeling lexically guided learning. Jeffrey Kwon (Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, jeffkwon@mit.edu), Jeung-Yoon Choi (MIT, Cambridge, MA), and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

The acoustic cue patterns used to realize and distinguish among phonemes (and thus among target words) in a spoken utterance include the spectral characteristics of frication noise, which signal the difference between, e.g., /s/ and / ʃ/ in American English. As part of a larger project developing a system of modules for automatic detection and interpretation of such cues, this study focuses on enhancing an existing Gaussian-mixture-model-based module for detecting spectral bursts/frication (Tubbs *et al.* 2024), which is designed to detect the spectral characteristics of frication noise that distinguish between these two obstruent consonants. The model provides the posterior probability of the acoustic cue (place of articulation of spectral burst/frication) given the relevant speech-related measurements. These detection probabilities are used to decide the identity of the phoneme, given a sequence of speech stimuli containing ambiguous spectral bursts embedded within a lexical context. The distributions for /s/ and /ʃ/ are then updated to reflect these decisions. The transparent nature of the GMM-based framework enables explicit tracking of the mechanism that can be used to model lexically guided learning (Cummings and Theodore 2023).

Session 2pSPa

Signal Processing in Acoustics: Acoustic Array Processing and Sound Field Reconstruction III

Efren Fernandez-Grande, Cochair
Department of Electrical Engineering, Technical University of Denmark (DTU), B.352, Oersteds Plads, Kgs Lyngby, 2800, Denmark

Peter Gerstoft, Cochair
Univ. of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093

Contributed Papers

1:00

2pSPa1. Invariant mode factorizations of a Bayes Factor high active sonar discriminant. Paul J. Gendron (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, pgendron@umassd.edu), Kenneth Bowers, and Abner C. Barros (ECE, Univ. of Massachusetts Dartmouth, Dartmouth, MA)

A Bayes factor active sonar (BFAS) processor for high-frequency broadband inference on vertical arrays in shallow water waveguides is presented. Relevant information regarding the refractive media, rough surface and volume reverberation is incorporated to build the marginal likelihood under the composite null while the alternative hypothesis must account for scattering from an object of uncertain depth. The processor is contrasted with the generalized likelihood ratio test (GLRT), where maximization is a surrogate for proper marginalization. The processor aggregates a set of time-varying quadratic forms over the array observations over beam-delay and we present an invariant mode factorization that is particularly insightful in terms of the eigenrays of the waveguide. We illustrate the approach by considering various refractive waveguides and demonstrate how prior information regarding the waveguide attenuates reverberant and noise subspaces weighing favorably target subspaces, effectively increasing signal-to-noise and reverberation ratios under uncertain target depths. The distribution of the BFAS under each hypothesis, while not admitting a simple closed form, does allow a lower bound through an approximation via moments. Bounds on receiver operating characteristic curves under depth uncertainty show the potential to outperform a direct path arrival detector with known target depth. [This work is supported by ONR.]

1:20

2pSPa2. A computational efficient Bayesian approach for active sonar localization under uncertainty in sound speed profile. Abner C. Barros (Naval Undersea Warfare Ctr., 285 Old Westport Rd., Dartmouth, MA 02747, abarros1@umassd.edu) and Paul J. Gendron (Dept. of Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Active sonar localization is challenging due in part to boundary interactions, relatively small aperture constraints, and the short coherence time associated with mobile bodies and dynamic environments. To address these challenges, a computational Bayesian approach is employed for joint inference of the wavevectors, characterizing the scattered field associated with the angle/Doppler spread arrivals. The resulting posterior is mapped to the posterior of the scattering body's location and speed under an uncertain sound speed profile (SSP) using a second-order variational Bayesian method. The approach employs a multivariate Gaussian model of the SSP with a lower-dimensional subspace representation of SSP uncertainty. The mobile scattering body's range, depth and speed joint posterior is constructed using a computationally efficient solution based on the Laplace approximation and marginalization over uncertainty in sound speed. A case study using SSPs from the Mediterranean Sea is presented to lend credence to the approach. [Work supported by NUWC ILIR and ONR.]

1:40

2pSPa3. Acoustic reflection parameterization based on the Spatial Decomposition Method. Lucas Hocquette (L-Acoust., 13 rue Levacher Cintrat, Marcoussis 91460, France, lucas.hocquette@l-acoustics.com), Philip Coleman, and Frederic Roskam (L-Acoust., London, United Kingdom)

The Spatial Decomposition Method (SDM) has gained interest within the acoustics community as a way to analyze and visualize the spatial sound field observed at a measurement position. Although the full visualization provides rich details of the sound field, it can be difficult to identify specific acoustic features or reflections within the analysis time frames. In this paper, we propose a parameterization of the SDM using a mathematical model of time-localized von Mises–Fisher distributions inspired by the image source model. Reflection directions of arrival, spatial concentration and diffuseness are estimated using an Expectation Maximization algorithm. Results are presented in our application domain of sound reinforcement. We compare parameters for sound reproduction with different acoustic absorption conditions and analyze how an immersive system in a theater excites the natural room acoustics.

2:00

2pSPa4. Enhanced speed of sound measurement in Distributed Acoustic Sensing (DAS) using Slepian multitaper frequency-wavenumber analysis. Peyman Moradi (SoundFalls LLC, 1600 Springwoods Plaza Dr., Apt 616, Spring, TX 77389, peyman.moradi@soundfalls.com)

Distributed Acoustic Sensing (DAS) has emerged as a powerful technology for monitoring wells and pipelines, with Speed of Sound (SoS) being a unique fluid property that can be derived from DAS data. However, accurately estimating the local SoS presents challenges primarily due to the nonstationary behavior of the data in the spatial domain, limited spatial resolution, and spectral leakage when applying FFT along either the time or spatial domains. In this study, we evaluate the performance of spatiotemporal multitaper frequency–wavenumber (FK) analysis for measuring SoS compared to conventional 2-D Fast Fourier Transform (FFT). We also introduce a joint methodology that applies multitapering in the time domain while implementing FFT in the spatial domain. This hybrid technique leverages the spectral concentration and noise robustness of Slepian sequences alongside the frequency resolution and lower variance of FFT. The joint methodology is rigorously validated through comprehensive analyses of synthetic and real-world DAS datasets, demonstrating its effectiveness in reducing the uncertainty of DAS-based SoS measurements.

2:20–2:40 Break

2:40

2pSPa5. Exploring autoproduct fields in three-dimensional acoustic environments: Simulations and experiments. Leslie V. Arciniega (Naval Architecture and Marine Eng., Univ. of Michigan, 521 N Div. St., Apt 1, Ann Arbor, MI 48104-1136, larcin@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Acoustic remote sensing and source localization are fundamental topics in underwater acoustics, with applications ranging from navigation, environmental monitoring, surveillance and reconnaissance. Traditional signal-processing techniques for such tasks typically rely on frequencies within the bandwidth of acoustic recordings. Recent developments, however, have shown that it is possible to extract environmental and source location information at frequencies outside a recorded signal's bandwidth using the frequency-difference and frequency-sum autoproducts [Worthmann and Dowling, J. Acoust. Soc. Am. 141, 4579–4590 (2017)]. Previous acoustic autoproduct studies have involved sound fields that primarily vary in two spatial dimensions. This presentation provides simulation and experimental autoproduct-based source localization results using vertical-array recordings collected in the fully three-dimensional four-ray-path acoustic field arising in Lloyd's mirror environment augmented with a vertical reflecting surface. The experiments were conducted in a 1.07-m-diameter freshwater tank using short Gaussian-enveloped pulses having a nominal bandwidth of 40–110 kHz. Time gating of the recordings was used to remove the tank sidewall and bottom reflections. The importance of acoustic reflection from the added vertical surface is explored by varying its distance from the plane defined by the source and receiving array. Comparisons between simulated and experimental localization results are presented. [Work sponsored by ONR.]

3:00

2pSPa6. Kalman filter based estimation of multiple signal waveforms from acoustic array data. Garth Frazier (NCPA, Univ. of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu)

This work presents a real-time signal processing algorithm that estimates *time-domain* waveforms of multiple plane wave signals on a sample-by-sample basis from data measured by an acoustic array. The algorithm is a modification of one presented at the ASA meeting in Ottawa, Canada, in May 2024 in which estimation is driven explicitly by performing nonlinear least-squares on sliding windows of data. In this new variation a state-variable approach is taken that enables the use of a Kalman filter for estimation, thus avoiding the sliding window approach. The algorithm essentially tracks estimates of the direction-of-arrival (DOA) of the waveforms from which the waveform estimates then can be obtained easily from the measured data. Other features of the approach include no restrictions on signal waveform shape and automatic estimation of the number of signals present. As before B-splines have been chosen for the waveform basis functions because of their partition-of-unity property.

3:20

2pSPa7. Development and evaluation of a cost-effective and flexible 4-microphone linear array for beamforming and sound source localization. William D. Fonseca (Acoust. Eng., Federal Univ. of Santa Maria, Av. Nossa Sra das Dores, 305, ap503B, Santa Maria, Rio Grande do Sul 97050-531, Brazil, will.fonseca@eac.ufsm.br) and Lucas Bogaz (Acoust. Eng., Federal Univ. of Santa Maria, Santa Maria, Rio Grande do Sul, Brazil)

This paper presents the design and analysis of a four-microphone linear array with variable spacing, aiming to offer a cost-effective solution for beamforming and direction of arrival (DoA) estimation. The project spans from the physical assembly of the array to the implementation of signal processing methods for accurately determining the position of sound sources in different environments. Initially, the choice and configuration of components, including microphones and data acquisition interfaces, are detailed, emphasizing the positional adjustment among elements to achieve optimal performance in various scenarios. Next, the results of experimental tests, which evaluated the system's robustness in terms of angular resolution, sensitivity to external noise, and ability to focus on specific directions are discussed. Finally, practical applications in sound source localization are demonstrated, highlighting the versatility and cost-effectiveness of the proposed array for academic research, industrial prototyping, and monitoring and security purposes.

3:40

2pSPa8. Space Launch System time alignment methodology. David Gillespie (ER42, NASA, Marshall Space Flight Ctr., Huntsville, AL 35812, david.m.gillespie@nasa.gov)

The Space Launch System (SLS) has thousands of sensors that are used to collect data across a range of physical environments including acoustics, thermal, and structural vibration. The SLS vehicle has multiple data acquisition units equipped with different types of acquisition cards, which record data at various sample rates depending on the connected sensor. These hardware and processing differences cause a wide range of filter delays that contribute to time misalignment between sensors. The subject of this paper is correcting this misalignment using cross-channel analysis to determine and visualize group delays and phase delays of the vehicle sensors. By grouping the sensors into families based on hardware and processing, analysis of less than 60 sensors yielded filter delay values for over 2000 sensor configurations on Artemis I. Verifying these corrections using time-corrected test data yielded alignment within one sample at the highest sample rate. Comparing the time of arrival of the booster igniter shock and ignition overpressure transients showed that the acoustic wave propagation was linear with vehicle elevation, thus validating the filter delay corrections. Throughout similar testing for Artemis II, the analysis highlighted multiple issues with the simulation setup, leading to improvements in both the simulations and processes.

4:00–4:20 Break

4:20

2pSPa9. Measurement of sonic boom with optimal planar microphone array in far field. Jae-hyoun Ha (6th R&D Inst., Agency for Defense Development, Keunheung-ro 1421-15, Tae-an, Choong-nam 32131, Korea, hahyoun@hanmail.net)

Planar microphone arrays are effectively used to measure and estimate the arrival angle of the sonic boom impinging on the ground. For the arrival angle estimation of sonic boom, the time difference model is more effective than the time and the phase signal model which is commonly used in array signal processing, because sonic boom signals are more likely to be overlapped by its reflection and diffraction. In this paper, performance analysis of arrival angle estimation with respect to its design is studied. The optimal sensor positions are presented in case of prediction of the arrival angle is given through the Cramer-Rao bound.

4:40

2pSPa10. Optimization framework for battery-powered sensor selection for sound source detection. Paritosh Mokhasi (NCPA, Univ. of MS, 145 Hill Dr., University, MS 38677, pmokhasi@olemiss.edu), William G. Frazier, and Wayne Prather (NCPA, Univ. of MS, Oxford, MS)

This study aims to address the multi-objective problem of selecting the optimal sensors to activate from a spatially distributed sensor array for sound source localization. We aim to achieve this with minimal sensor deployment, thereby maximizing the battery life of a device that powers both the sensors and an onboard processing system. The optimization problem has two conflicting objectives: (a) minimize the sensor deployment, which directly translates to lower power consumption and (b) maximize source localization confidence. We present a novel deterministic greedy algorithm for solving the multi-objective optimization problem. A dictionary of optimized sensor locations is first pre-computed and stored for a variety of scenarios for single source locations case. For test cases, we use the dictionary to generate an initial guess for the actual greedy optimization algorithm. We demonstrate the efficiency of the algorithm through various simulated test cases.

5:00

2pSPa11. Anomalous drone detection and localization with a single acoustic vector sensor. Jianghan Hai (Sichuan Univ., Chengdu, China), Dengjian Zhou, and Yue Ivan Wu (Sichuan Univ., No. 24 South Section 1, Yihuan Rd., Chengdu 610065, China, y.i.wu@ieee.org)

Proposed in this work is a method to detect and locate an anomalous drone by passively receiving and processing the drone's noise with a single acoustic vector sensor on the ground. Specific spatio-temporal-frequency features of sound pressure and particle velocity are extracted from the multichannel noise data, formulating the designed input feature vector to train a machine learning model for anomaly detection. The detected anomalous drone is then located by array signal processing with a single sensor. In the field tests, a drone hovers in eight distinct directions with respect to the sensor. To evaluate the proposed algorithm, the artificial anomaly sounds are broadcasted by a loudspeaker carried by the drone. More practically, the noise from a drone with two fractured blades is also captured and processed, which is indistinguishable from the normal drone noise to human hearing. Experimental results show that the proposed algorithm achieves good performance at both anomaly detection (with an F1 score over 90%), and drone localization (with an average angular estimation error below 9 deg).

TUESDAY AFTERNOON, 20 MAY 2025 SALON D, 1:00 P.M. TO 5:00 P.M.

Session 2pSPb

Signal Processing in Acoustics: Signal Processing Potpourri II

Trevor Jerome, Chair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, Bldg. 3 #329, West Bethesda, MD 20817

Contributed Papers

1:00

2pSPb1. The Chebyshev polynomial frequency modulation waveform model. Stephen P. Blackstock (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, spb@arlut.utexas.edu) and Michael R. Haberman (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Polynomial-phase signals (PPS) are a staple of waveform design and analysis in sonar, radar, and communications fields. They also find application in the modeling of bioacoustics emissions, especially those of echolocating animals such as bats and odontocetes. This work presents a novel PPS waveform formulation that exploits some special properties of Chebyshev polynomials, such as orthogonality, recurrence relations, and equivalence to trigonometric functions. The result is the Chebyshev polynomial frequency modulation (CPSFM) family of waveforms, which prove useful in the modeling of bioacoustic signals and the approximation of non-polynomial-phase signals such as hyperbolic chirps. We demonstrate that the CPSFM model admits compact analytic expressions for fundamental continuous-time signal processing functions such as the Fourier transform, the convolution and correlation operations, and the ambiguity function. Derivations for these expressions using CPSFM are presented, along with their application to the analysis of biosonar emissions of Mexican freetail bats.

1:20

2pSPb2. Efficient stimuli-driven global sensitivity analysis of acoustic metamaterials using radial basis function interpolation. Jiahua Zhang (Test Div., Siemens Digital Industry Software, Interleuvenlaan 68, Leuven 3001, Belgium, jiahua.zhang@siemens.com), Jacques Cuenca, Laurent De Ryck (Test Div., Siemens Digital Industry Software, Leuven, Belgium), Lucas Van Belle (Mech. Eng., Flanders Make@ KU Leuven, KU Leuven, Heverlee, Belgium), and Elke Deckers (Mech. Eng., Flanders Make@ KU Leuven, KU Leuven Campus Diepenbeek, Leuven, Belgium)

The design optimization of acoustic metamaterials represents a critical challenge in modern noise control applications. Global sensitivity analysis (GSA) has emerged as a powerful tool for quantifying the correlation between metamaterials' design variables and corresponding sound quality metrics (loudness, sharpness, and speech intelligibility index). However, GSA's high computational intensity limits the practical application of design optimization, especially for sound quality metrics which vary with sound stimulus. This study introduces a computational framework using radial basis function (RBF) interpolation to accelerate the GSA of acoustic metamaterials, with a periodic arrangement of micro-perforated panels (MPPs) serving as an exemplary case study. The methodology employs RBF interpolation to create a surrogate model of metamaterial acoustic response, demonstrating the potential for computational cost reduction while maintaining accuracy. It is validated across three acoustic scenarios that represent common challenges in noise control: stationary signals (broad/narrowband noise), environmental sounds (mechanical noise, traffic), and dynamic signals (speech, music). The study systematically compares sensitivity indices derived from direct numerical simulation and interpolated predictions, assessing the RBF's impact on GSA accuracy. [The European Commission is gratefully acknowledged for supporting the Marie Sklodowska Curie program through the Horizon Europe DN METAVISION project (GA 101072415)]

1:40

2pSPb3. Graph-based localization of impact noise in buildings. Haesang Yang (Artificial Intelligence and Robotics, Sejong Univ., Neungdong-ro 209, Gwangjin-gu, Seoul 05006, Korea, haesang.yang@sejong.ac.kr)

This study focuses on localizing impact noise in buildings, particularly noise that disturbs residents. Traditional data-driven approaches using supervised learning with location labeling often neglect the structural influence of buildings, which act as complex mediums for sound transmission. Previous research revealed reduced localization performance when the same source-to-receiver positions were tested across different floor sections [Appl. Sci. 11, 5399 (2021)]. To overcome this limitation, this study proposes a graph-based localization method that incorporates building structure. The building's components are modeled as graph edges, spatial relationships as nodes, and medium properties as vertex features. The model's learnable weights are optimized to minimize the discrepancy between observed impact noise and predicted responses at receiver nodes. The method is validated using two data splits with identical source-to-receiver positions but different floor sections. Visual analysis highlights distributional differences between splits. Transfer learning tasks demonstrate that the graph-based method outperforms baseline models in knowledge transfer. Additionally, visual demonstrations confirm its superior localization performance.

2:00–2:20 Break

2:20

2pSPb4. Impact of cochlear implant processing on acoustic cues critical for room adaptation. Prajna BK (Speech & Hearing Sci., UIUC, 901 S 6th St., Champaign, IL 61820, prajnab2@illinois.edu) and Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Typical-hearing listeners are generally resilient to distortions in low to moderate reverberant environments, whereas cochlear implant (CI) users are significantly affected. Typical-hearing listeners adapt to a reflective space through short-term learning from prior exposure to the distortion patterns, which enhances signal comprehension. This adaptation relies on specific acoustic cues, such as attenuation of modulation depth in the temporal envelope, spectral coloration introduced by the room, and statistical properties of the reverberation tail, including skewness and kurtosis. It remains unclear how CI processing alters these acoustic cues and whether they are accessible to CI users. To investigate, dry and reverberant speech signals were analyzed before and after simulating key stages of CI signal processing, including dynamic range compression, bandpass filtering, envelope extraction, and current spread. Preliminary findings suggest CI processing causes slightly greater attenuation of the speech envelope's modulation depth compared to reverberation alone relative to the dry signal. Additionally, room-induced spectral coloration was altered by CI processing, and reductions in the kurtosis and skewness of the reverberation tail were observed. These findings may partially explain why CI users struggle in reverberant environments.

2:40

2pSPb5. An optimal direct sequence spread spectrum packet design for underwater acoustic channels with multipath and Doppler effect. Yegwon Hong (Radio Commun. Eng., Korea Maritime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Busan 49112, Korea, cl170625@gmail.com), Hyun-Woo Jeong, JaeHun Lee, and Ji-Won Jung (Radio Commun. Eng., Korea Maritime and Ocean Univ., Busan, Korea)

Underwater acoustic communication channels have various characteristics including attenuation, distortion, multipath, and Doppler frequency shifts that degrade communication reliability. To overcome these challenges, among various communication techniques, direct sequence spread spectrum (DSSS) has been studied recently. DSSS compensates for signal distortion caused by multipath effects and is robust against frequency-selective fading due to its wide frequency spectrum. For compensating DSSS signals, a RAKE receiver was employed which is a matched filter that correlates received the signal with the spreading code on each tap. When configuring spread data format in designing packet structure for various spreading factors and source data rates, it is essential to consider channel state information (CSI) such as coherent time and coherent bandwidth, which are influenced by multipath and Doppler frequency shifts. Through simulations, we propose an optimal DSSS packet design model that satisfies CSI both in the time domain and frequency domain by evaluating performance for these channel conditions. [This research is supported by a KRIT grant funded by the Korean government (DAPA) (KRIT-CT-23-035-01, Multi AUV operation Technology for Mine Detection ('23–'28))]

3:00

2pSPb6. Polar-turbo equalization in orthogonal frequency division multiplexing for underwater acoustic communications. JaeHun Lee (Radio Commun. Eng., Korea Maritime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Busan 49112, Korea, bear9907@g.kmou.ac.kr), Hyun-Woo Jeong, Yegwon Hong, and Ji-Won Jung (Radio Commun. Eng., Korea Maritime and Ocean Univ., Busan, Korea)

Orthogonal frequency division multiplexing (OFDM) is known to be an effective technique to overcome frequency-selective fading caused by multipath propagation and Doppler effect. OFDM divides the transmitted signal into multiple narrowband subcarriers, significantly reducing the negative impact of fading across the frequency spectrum. To improve the reliability and performance of OFDM systems, polar codes, introduced by Erdal Arıkan in 2008, are applied in underwater communications. Another approach is to use the polar-turbo equalization which the equalizer and polar decoder are connected through interleaver and de-interleaver recursively updating each other's information. We propose a model that combines with polar-turbo equalization in OFDM systems. Proposed polar-turbo equalization in OFDM effectively mitigates performance degradation caused by frequency-selective fading and enhances communication reliability in underwater environments. [This research is supported by KRIT grant funded by the Korean government (DAPA) (KRIT-CT-23-035-01, Multi AUV operation Technology for Mine Detection ('23–'28))]

3:20

2pSPb7. Design of new auxiliary function for fully blind spatially regularized independent low-rank matrix analysis. Sota Hirata (The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8654, Japan, sh020112todai@g.ecc.u-tokyo.ac.jp), Norihiro Takamune, Kouei Yamaoka (The Univ. of Tokyo, Bunkyo-ku, Tokyo, Japan), Daichi Kitamura (National Inst. of Technol., Kagawa College, Takamatsu-shi, Kagawa, Japan), Hiroshi Saruwatari (The Univ. of Tokyo, Bunkyo-ku, Japan), Yu Takahashi, and Kazunobu Kondo (Yamaha Corp., Hamamatsu-shi, Shizuoka, Japan)

A representative method for blind source separation (BSS) is independent low-rank matrix analysis (ILRMA). Spatially regularized ILRMA (SR-ILRMA) utilizes prior information about the acoustic transfer system, such as steering vectors (SVs) for each source, as a regularizer in ILRMA. Although it has been reported that SR-ILRMA achieved higher separation performance than ILRMA in an experiment, SR-ILRMA is not a fully blind method; i.e., SVs should be known in advance. In our previous study, we proposed a fully blind SR-ILRMA that simultaneously estimates SVs and other parameters in SR-ILRMA on the basis of the majorization–minimization (MM) algorithm. Since an auxiliary function is not unique in the MM algorithm, it may be possible to design a better auxiliary function that achieves faster convergence. In this paper, we design a new auxiliary function for deriving the update rule of SVs, motivated by the conjecture that an auxiliary function that better approximates the cost function would lead to faster convergence. In a numerical experiment, we confirm that the update rule based on the new auxiliary function achieves faster convergence than that based on the conventional one.

3:40–4:00 Break

3:40

2pSPb8. Analysis of projection-back-free property in fully blind spatially regularized independent low-rank matrix analysis. Sota Hirata (The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8654, Japan, sh020112todai@g.ecc.u-tokyo.ac.jp), Norihiro Takamune, Kouei Yamaoka (The Univ. of Tokyo, Bunkyo-ku, Tokyo, Japan), Daichi Kitamura (National Inst. of Technol., Kagawa College, Takamatsu-shi, Kagawa, Japan), Hiroshi Saruwatari (The Univ. of Tokyo, Bunkyo-ku, Japan), Yu Takahashi, and Kazunobu Kondo (Yamaha Corp., Hamamatsu-shi, Shizuoka, Japan)

Independent low-rank matrix analysis (ILRMA) is a representative method for blind source separation. Unfortunately, in ILRMA, there is scale ambiguity in separated signals for each frequency bin. To adjust the spectral scale of separated signals, a post-processing technique called projection back (PB) is commonly used. However, PB requires the calculation of the inverse of the demixing matrix, which often amplifies numerical errors when the demixing matrix is ill-conditioned. In our previous study, we proposed fully blind spatially regularized ILRMA (FB-SR-ILRMA), which is a blind method to simultaneously estimate the demixing matrices and the steering vectors of each source without knowing sensor positions, as with PB. In this paper, we focus on the potential ability of FB-SR-ILRMA to automatically adjust the scale of separated signals on the basis of distortionless response, which is a common principle in the field of beamforming. Thus, FB-SR-ILRMA can dismiss PB. In a numerical experiment, we investigated the difference in separation performance of FB-SR-ILRMA with and without PB. We confirmed that FB-SR-ILRMA without PB provides separated signals with sufficiently high quality, comparable to the oracle minimum variance distortionless response beamformer. We also confirmed that PB tends to degrade the source-to-interference ratio.

4:20

2pSPb9. Comparison of field programmable gate arrays and digital signal processors for the implementation of active noise control algorithms. Tim Karl (Mechatronics, Helmut-Schmidt-Univ., Hamburg, Germany, karlt@hsu-hh.de) and Sachau Delf (Mechatronics, Helmut-Schmidt-Univ., Hamburg, Germany)

Active noise control (ANC) systems have become increasingly versatile due to the advancing computational capabilities of modern digital signal processors (DSPs) and field programmable gate arrays (FPGAs). The implementation of ANC algorithms requires consideration of not only acoustic performance, but also factors such as resource consumption, thermal behavior, efficiency, and cost-effectiveness. These aspects play a critical role in the design and deployment of ANC applications in various industries. This study examines the implementation of an ANC algorithm on both FPGA and DSP platforms, providing a comparative analysis that highlights the strengths and weaknesses of each ANC system. The inherent architectural differences between FPGAs and DSPs are examined, with a particular focus on processing efficiency, latency, adaptability and scalability. In addition, the study evaluates the power consumption and thermal management of both platforms, which are critical for long-term operation in real-world applications. Based on the experimental results, the analysis provides practical recommendations for selecting the optimal platform for specific ANC use cases.

4:40

2pSPb10. Semi-supervised deep monaural speech enhancement with positive-negative-unlabeled learning. Ryo Ogawa (The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, ogawaryo2531@g.ecc.u-tokyo.ac.jp), Yuki Yonekura, Nobutaka Ito, Norihiro Takamune, Kouei Yamaoka, Yuki Saito, and Hiroshi Saruwatari (The Univ. of Tokyo, Bunkyo-ku, Tokyo, Japan)

Monaural speech enhancement (SE) is a technique for extracting a clean speech signal from a monaural noisy speech signal. Its mainstream approach, supervised learning, uses supervised data, i.e., pairs of clean and noisy speech data. However, this approach has the problem that supervised data are expensive because recording clean speech data requires a quiet environment such as a studio. In this paper, an SE method using a semi-supervised learning method called positive-negative-unlabeled (PNU) learning is proposed. To achieve high SE performance even with limited supervised data, the proposed method leverages unsupervised data, i.e., only noisy speech data. Note that unsupervised data can be easily collected, e.g., from smart speakers or the Web. In our method, a deep neural network predicts a binary mask for SE by classifying time-frequency bins as speech-dominant (positive, P) or noise-dominant (negative, N). It is trained through PNU learning using P and N data from supervised data and unlabeled (U) data from unsupervised data. An experiment confirmed that increasing U data improves the SE performance of the proposed method and enables it to outperform supervised learning.

Session 2pUW

Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Computational Acoustics: Ambient Sound Measurements and Models II

Martin Siderius, Cochair
Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201

S. B. Martin, Cochair
Halifax, JASCO Applied Sciences, 20 Mount Hope Avenue, Dartmouth, B2Y 4S3, Canada

Kay L. Gemba, Cochair
Naval Postgraduate School (NPS), 833 Dyer Road, Bldg. 232, Physics Department, Monterey, CA 93943

Jie Yang, Cochair
Applied Physics Lab, University of Washington, 1015 NE 40th Street, Seattle, WA 98105

Invited Papers

1:00

2pUW1. Detecting and quantifying oceanic rainfall using passive acoustics. Barry B. Ma (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, barry@apl.uw.edu)

Raindrops striking a tin roof create distinctive sounds that allow the human ear to discern the onset, intensity, and cessation of rainfall. Similarly, the ocean surface functions as a natural "tin roof," producing acoustic signals when impacted by raindrops. By deploying hydrophones beneath the water's surface, these unique sound patterns can be detected and analyzed to quantify rainfall events. This study details the process of identifying and interpreting these acoustic signals, focusing on how specific sound characteristics—such as frequency and amplitude—correlate with rainfall rates. It further introduces a quantitative model designed to translate these acoustic signatures into accurate rain rate measurements. When integrated into oceanographic monitoring systems, this technique enhances the ability to observe rainfall in remote ocean regions where traditional methods are difficult to apply. By improving the accuracy and accessibility of rainfall measurements, this innovative approach has significant potential to advance weather and climate predictions. The study also addresses critical topics, including the origin of the concept, the mechanisms for detecting and quantifying rain rates, accumulation, and drop size distributions, as well as a comprehensive comparison of its assumptions, limitations, advantages, and disadvantages relative to other rainfall measurement methods.

1:20

2pUW2. Comparing ocean ambient sound measurements and modeling techniques around Newfoundland's seismic surveys. Pablo Borys (Jasco Appl. Sci., 20 Mt Hope Ave., Halifax, NS B2Y4S3, Canada, pablo.borys@jasco.com), Marie-Noel Matthews (Jasco Appl. Sci., Dartmouth, NS, Canada), S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), Alexander MacGillivray (Jasco Appl. Sci., Victoria, AB, Canada), and Corey Morris (Fisheries and Oceans Canada, St. John's, NF, Canada)

To assess the impact of anthropogenic noise on groundfish off Newfoundland's coast, JASCO Applied Sciences worked with Fisheries and Oceans Canada to model the ocean soundscape during months of high fishing and seismic survey activity from 2015 to 2021. The study area included key fishing grounds on the Grand Banks where commercial groundfish harvests overlap with seismic exploration activities by the offshore oil and gas industry. The project used ARTEMIA, a soundscape mapping model, to simulate underwater sound from fishing and commercial vessels, seismic surveys, and wind. ARTEMIA integrated data from vessel tracking, sound emission specifications, and meteorological conditions to predict marine soundscape contributions. Sound propagation was modeled in octave bands from 16–512 Hz using the parabolic equation method, providing insights at unprecedented scales and fine spatial resolutions. The model's predictions were validated against measured sound pressure levels (SPL) from JASCO's acoustic recorders, showing good agreement with median differences below 4 dB and standard deviations below 1 dB for some years. In this work, we discuss the factors influencing agreement between modeled and measured data, including cases of agreement, challenges in specific scenarios, and discrepancies due to inaccuracies in the ambient model at low and high wind speeds. These insights help refine the understanding of soundscape dynamics and improve the accuracy of soundscape models. [Work supported by Fisheries and Oceans Canada.]

Contributed Papers

1:40

2pUW3. Using seismic sources of opportunity to investigate sound propagation in the Gulf of Mexico. John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu), Sean Wiggins, Bruce Cornuelle, and Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

For more than two decades, a network of acoustic sensors has been monitoring ambient sound across the Gulf of Mexico (GOM), a region with extensive seismic survey activity. These surveys, trackable through the Automatic Information System (AIS), produce low-frequency acoustic signals that offer a unique opportunity to study sound propagation over large distances. The signals are consistently detectable across the sensor network, covering ranges of up to ∼1000 km. Analysis reveals that transmission loss is strongly influenced by bathymetric features between survey locations and receivers. Specifically, downslope conversion and subsequent deepwater propagation reduce propagation loss, while upslope and along-slope propagation lead to greater propagation loss. This study highlights the potential for seismic surveys to be a valuable resource for advancing our understanding of acoustic propagation in marine environments.

2:00

2pUW4. Ambient noise modeling in ODIN. Andrew Holden (Underwater Group, Dstl, Fareham PO17 6AD, United Kingdom, apholden@dstl.gov.uk)

ODIN is an underwater engagement simulation tool that models interacting entities (such as ships and UUVs) in underwater environments where the entities have tactics that define their behavior over time. An ODIN user has great flexibility in defining a wide range of entity and environmental parameters. ODIN can model ambient noise to predict in beam ambient noise. It undertakes these predictions using its in built ray trace model along with the Kuperman Ferla, Ainslie and APLUW surface noise source models. The presentation will show ODIN predictions of omni noise levels and array response to noise in range-dependent environments using the three source level equations. These are compared with some published papers on isotropic ambient noise data. There is also a comparison with the CANARY Ambient Noise Model that was developed in the 1990s.

2:20

2pUW5. Time domain model of underwater sound propagation for offshore wind turbine. Eunkuk Son (Wind Energy Res. Dept., Korea Inst. of Energy Res., 200 Haemajihaean-ro, Gujwa-eup, Jeju 63357, Korea, eunkuk.son@kier.re.kr), Jinjae Lee, and Seungjin Kang (Wind Energy Res. Dept., Korea Inst. of Energy Res., Jeju, Korea)

This paper introduces a numerical method for modeling underwater sound propagation in the time domain, specifically applied to offshore wind turbines. Given that the majority of wind turbines are situated in ocean depths less than 100 m, termed shallow water, the paths of underwater sound propagation become intricate, particularly toward specific receptor positions. Dealing with the complexities of multi-path reflection waves and scattering of the edges, especially in the shallow water regime, incurs significant computational costs in the time domain. To analyze sound propagation in moving inhomogeneous media within the time domain, a numerical technique was implemented. Treating the top and bottom of the ocean as acoustically hard surfaces and considering the horizontal direction as an infinite boundary, the perfectly matched layer is implemented. To address the high computational demands, GPU parallelization was employed, significantly reducing computation time, particularly for large-scale simulations. The validity of the algorithm is further verified through underwater sound propagation experiments. The results demonstrate that the differences in transmission loss between measurements and simulations at the target frequencies remain within 10 dB.

2:40–3:00 Break

Invited Papers

3:00

2pUW6. Modeling source levels of marine shipping. Alexander O. MacGillivray (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z7X8, Canada, alex@jasco.com) and Max Schuster (JASCO ShipConsult, Schwentinental, Germany)

Noise from shipping traffic is often the dominant anthropogenic contributor to the underwater soundscape. Therefore, modeling underwater ambient sound requires accurate methods for estimating the source levels of marine vessels. This paper reviews recent developments in modeling source levels for ships and other marine vessels within the context of underwater ambient sound prediction. Source levels for ships are typically estimated using incomplete descriptions of vessel characteristics from data sources such as the Automatic Identification System (AIS). Large datasets on vessel source levels, such as the ECHO database, have enabled the development of improved models for marine shipping source levels. Source levels are often estimated statistically, based on information about ship class, operating conditions, and design characteristics. While source levels for individual vessels may have high uncertainties, source levels for populations of vessels follow predictable trends. This paper reviews ship noise models used in large-scale soundscape mapping projects, such as JOMOPANS and NAVISON. The different approaches are compared, and limitations and data gaps are discussed.

3:20

2pUW7. NOAA efforts to advance community standards for marine soundscape interpretation and applications. Samara Haver (NOAA Northeast Fisheries Sci. Ctr., NOAA NEFSC, Woods Hole, MA 02543, samara.haver@noaa.gov), Megan F. McKenna (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado Boulder, Pacific Grove, CA), Robert P. Dziak (NOAA PMEL, Newport, OR), Jason Gedamke (NOAA NMFS, Silver Spring, MD), Leila Hatch (NOAA ONMS, Scituate, MA), Jessica McCordic (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), Xavier Mouy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Timothy Rowell (NOAA NMFS, Beaufort, NC), Taiki Sakai, Rebecca Van Hoeck (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), Carrie Wall (Univ. of Colorado, Boulder, CO), and Sofie Van Parijs (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA)

Interest and data for underwater soundscape monitoring is rapidly expanding, necessitating scalable and efficient data handling, processing, and dissemination tools. The NOAA National Marine Fisheries Service (NMFS) passive acoustic monitoring strategic initiative (PAM SI) is one of these efforts, and includes national-level planning for analysis and integration of soundscape metrics. Under the PAM SI, we are building on multi-year progress toward community analysis standards to scale national-level comparable data products from multiple recording technologies and projects. This effort includes refining archival methods, software tools, and baseline soundscape metrics to integrate with species detection and environmental variables. In collaboration with other NOAA line offices, agencies, and academic partners our work will continue to advance PAM data analysis and integration approaches beyond the PAM SI. A coordinated NOAA soundscapes program will leverage this progress and products to provide information about the status and condition of protected species and environments. The NOAA/National Park Service Noise Reference Station network has been sampling throughout U.S. waters for over a decade, providing exemplar datasets for each step in this big data challenge. Current applications for these data include quantifying multi-year regional trends in relation to species presence, anthropogenic factors, and climate patterns.

3:40

2pUW8. Exploring dolphin spatial hearing in ocean ambient soundscape via model head-related transfer functions. Wu-Jung Lee (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), YeonJoon Cheong (Univ. of Washington, Seattle, WA), Matt Schalles (New College of Florida, Pittsburgh, PA), and Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA)

The ocean ambient soundscape is a dynamic mixture of natural and anthropogenic acoustic signals. As fully aquatic mammals, dolphins rely primarily on acoustic cues—from pure listening and echolocation—to navigate, communicate, and forage. Spatial hearing, the ability to localize sound sources, is central to how dolphins interpret and negotiate rich underwater acoustic scenes. Here, we explore dolphins′ spatial hearing capabilities using head-related transfer functions (HRTFs) predicted from finite element modeling using a 3-D volumetric representation of a dolphin head. The HRTF predictions revealed a distinct transition of hearing directionality around 15–20 kHz: Broad, lateralized patterns dominate the lower communication band, while highly directional, frontally oriented patterns appear in higher echolocation frequencies. These features arise from the unique anatomical structures in the dolphin head, including mandibular fat bodies that refract and guide sound toward the ears, and extensive air spaces around the ears and beneath the skull that acoustically isolate each ear from contralateral sources. Results suggest that dolphins can effectively localize environmental sounds and achieve high spatial acuity during echolocation despite long acoustic wavelengths of audible underwater sound. Our study offers fresh perspectives on dolphin acoustic ecology and provides a foundation to assess the potential impacts of anthropogenic sounds in the ocean.

4:00

2pUW9. Predicting the biological component of ocean soundscapes. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Andrew Heaney (Appl. Ocean Sci., Fairfax Station, VA)

The ocean can be a noisy place. Sound travels extremely well, particularly at lower frequencies (below 1 kHz) and sound sources from great distances can contribute to the local soundscape. The soundscape is defined as the spectral and spatial characteristics of the sound field from all sources. Emphasis has been primarily on "noise modeling" for Anti-submarine warfare (ASW) performance and modeling and more recently on sound exposure to marine mammals, for conservation purposes. For these reasons, the primary sources of interest have been wind and ships, seismic exploration, and pile driving. Yet with animals using sound for many things (communication, hunting and possibly local environmental awareness and navigation), there is a rich contribution to the soundscape from most marine animals. In this work, we present a modeling approach to simulate the spectral and spatial contribution of marine mammals into the ocean soundscape. The model includes state-of-the-art knowledge of animal distributions, call behavior and rates (inter-click intervals) and characteristics. Each of these is the result of entire communities of researchers. We present a simulated soundscape (spectrogram and bearing time record) that includes wind, ships, Minke, Fin, and Sei whales in the Gulf of Maine.

Contributed Papers

4:20

2pUW10. Towards a transiting ocean observer network with passive acoustic localization. Erin M. Fischell (Acbotics Res., LLC, 82 Technol. Park Dr., East Falmouth, MA 02536, efischell@acbotics.com), Oscar A. Viquez (Acbotics Res., LLC, Falmouth, MA), Holly Bik (Univ. of Georgia, Athens, GA), Theodore W Callis, Emmanouil M Tentzeris (Georgia Inst. of Technol., Atlanta, GA), Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), and Martin Siderius (Portland State Univ., Portland, OR)

Most underwater soundscape recording systems are limited to a single hydrophone without peripheral, time-synchronized environmental sensing, meaning that local environmental parameters and directional soundscape information are lost. PLUTOS (Passive Localized Underwater Transiting Observing System) is currently under development to provide multi-modal sensing capability relating acoustic, physical, and eDNA sampling in a low-cost underwater drifting or moored package. Based on the AcSense stack with creative commons non-commercial hardware, the objective is flexible multi-drifter sensing of acoustic, environmental, and biological parameters in a package that is accessible to academic users. A novel soundscape-relative localization method provides a pathway to a passive acoustic approach for underwater localization. We present the initial system integration, simulation, as well as field deployment and data analysis of the prototype that included synchronous 8-channel hydrophone array recording with CTD and

pumped eDNA sampler. The system design includes an objective of multiple drifting systems including hydrophone arrays to disambiguate time and space signals in ocean properties in addition to triangulation of sound sources [Work funded by NSF.]

4:40

2pUW11. Estimating sea-surface ambient noise levels at mid-frequency using a compact hydrophone array towed by an autonomous surface vehicle. Davis Rider (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405, drider3@gatech.edu), James S. Martin (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Laurent Grare, Luc Lenain (Scripps Inst. of Oceanogr., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ambient noise monitoring in the ocean can provide insight into physical oceanographic processes such as wind and wave generation. Additionally, accurate environmental parametrization of the noise levels is needed for SONAR systems performance prediction. Typically, passive acoustic monitoring is conducted with stationary buoys that can record for long periods of time or large hydrophone arrays towed by manned vessels, creating a tradeoff between monitoring area and mission duration. Here, an autonomous surface vehicle platform (Liquid Robotics Wave Gliders—WG) instrumented for physical oceanography measurements and also outfitted with a compact four-element tetrahedron-shaped hydrophone array is presented. Multiple WGs were deployed in the vicinity of the Atlantis II seamount to perform ambient acoustic surveys at depths ranging from 10 to 150 m. During these geo-localized surveys, the measured relationship between mid-frequency ambient noise levels and local wind speed was found to be consistent with previous experiments, demonstrating the efficacy of the Wave Glider as an ambient noise monitoring platform. Additionally, the mid-frequency beamforming capabilities of their compact array are leveraged to estimate in-situ the effective sound pressure level of the local sea-surface noise sources and its parametrization on the local wind speed which is needed for ambient soundscape modeling. [Work sponsored by ONR.]

Session 3aAAa

Architectural Acoustics: Architectural Acoustics Potpourri

Molly Smallcomb, Cochair
Grad. Prog. in Acoust., Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

Zane T. Rusk, Cochair
The Pennsylvania State Univ., 104 Engineering Unit A, University Park, PA 16802

Contributed Papers

9:00

3aAAa1. Identifying gaps in acoustic questionnaires for residential care facilities to enhance comprehensive subjective evaluation methods. Saleh Naseer (College of Design, Construction and Planning, School of Arch., Univ. of Florida, 2919 Sw 13th St., 92, Gainesville, FL 32608, saleh.naseer@ufl.edu) and Hassan Azad (College of Design, Construction and Planning, School of Arch., Univ. of Florida, Gainesville, FL)

The quality of the sonic environment in residential care facilities (RCFs) is one of the several factors that significantly influence the quality of life (QL) of older adults. It can affect their communication, sleep, and level of satisfaction with the space, in general. However, existing subjective acoustic assessment techniques often fail to address this issue comprehensively and as it relates to the QL. This study investigates the limitations of current acoustic questionnaires used across various building types when assessing occupants' sonic experiences, with a particular focus on RCFs. By identifying critical gaps in evaluating both positive and negative sounds in the sonic environment, the research aims to contribute to a more effective framework for assessing acoustic quality and experiences in RCFs and their impact on the QL of older adults. A preliminary study will be conducted in RCFs in Gainesville and Trenton, Florida, to further analyze and evaluate the suggested approach to ensure its effectiveness in providing comprehensive acoustic assessments in these settings.

9:20

3aAAa2. Optimizing low-frequency reverberation in a critical listening room. José N. Sarinho Filho, Rodolfo Thomazelli, and Bruno Masiero (Universidade de Campinas, Av. Albert Einstein, 400, Campinas, São Paulo 13083-852, Brazil, masiero@unicamp.br)

To evaluate subtle nuances in different audio reproduction systems or conduct auditory perceptual studies, international standards, such as the ITU-R BS.1116-3, indicate the need for a space with minimal residual noise and very low reverberation in all frequencies of interest. We constructed a Critical Listening Room (CLR) designed to meet these stringent requirements. The first construction phase employed a "box-in-a-box" design, featuring external walls made of cement blocks and internal walls built with a light steel frame and triple-layered gypsum boards, all resting on a floating floor. This design achieved outstanding sound isolation, surpassing the NR10 criterion, with residual noise levels of less than 6 dBA. The second construction phase focuses on installing internal wall treatments to address the room's reverberation time, which remains notably high at lower frequencies (e.g., 7 s at 50 Hz). To mitigate this, mineral wool and membrane absorbers are commonly used. In this study, we evaluate the effectiveness of various absorber configurations in controlling low-frequency reverberation, presenting results from detailed acoustic measurements conducted in the CLR.

9:40

3aAAa3. An attempt to apply the element-free Galerkin method to room acoustics simulation. Jun Sasaoka (Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 860-8555, Japan, 245d9242@st.kumamoto-u.ac.jp) and Keiji Kawai (Kumamoto Univ., Kumamoto-shi, Japan)

The Element-free Galerkin method (EFGM) is one of the meshfree methods that have been developed primarily in the fields of strength of materials and fluid dynamics. Compared to conventional numerical methods, such as FEM, BEM, and FDTDM, EFGM does not divide the analysis area or boundary into elements, thus eliminating the need for tasks and time required for element division. In addition, divergence is less likely to occur in the mid- and high-frequency ranges, and this method can be applied accurately to complex geometries, because the interpolation function at each node is calculated continuously in the analysis area. However, these benefits have not been confirmed because there is no previous study of EFGM for room acoustics simulation. In this study, an attempt was performed to examine the potential of EFGM for room acoustics simulation. The results of the EFGM calculations were compared with the results of benchmark platform organized by Japanese numerical analysis researchers and acoustic measurements in an actual space.

10:00–10:20 Break

10:20

3aAAa4. Numerical wave propagation in complex acoustical environments using hybrid wave-based and geometrical acoustics methods. Solvi Thrastarson (Treble Technologies, Reykjavik, Iceland), Konstantinos Gkanos (Treble Technologies, Kalkofnsvegur, Reykjavik 101, Iceland, konstantinos.gkanos@treble.tech), and Jesper Pedersen (Treble Technologies, Reykjavik, Iceland)

Accurate modeling of sound propagation in realistic environments presents significant challenges due to the wide frequency range of interest and the intricate details of the boundary interactions. Wave-based methods excel at low frequencies but become expensive at higher frequencies. In contrast, geometrical acoustics (GA) methods are highly efficient for high-frequency modeling, yet they may lack accuracy compared to its wave-based counterparts. This study presents a hybrid approach that combines these two methodologies to offer a practical, but highly accurate, solution for sound propagation in complex acoustic environments, such as fully furnished multi-room apartments. By integrating both methods, we achieve high accuracy without the expense of using a wave-based technique across the entire frequency range. The hybrid method is applied to a multitude of multi-room apartment scenarios, showcasing how it effectively handles complex geometries that influence sound propagation. Key to the success of this method is the tuning and implementation of the GA method to ensure it's compatibility with the wave-based method. The method's efficiency and accuracy makes it well-suited for practical applications where computational resources are limited but high-fidelity results are required across a broad range of frequencies making it a cost-efficient and more scalable alternative to measuring real-world scenarios.

10:40

3aAAa5. Theoretical refinement of sound field diffusion indices determined from incidence directivity. Ryo Hagiwara (Architecture, The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, hagiryo23@g.ecc.u-tokyo.ac.jp), Tetsuya Sakuma, and Yosuke Yasuda (Architecture, Kanagawa Univ., Yokohama, Kanagawa, Japan)

This study focuses on two types of sound diffusion indices determined from incidence directivity at a receiving point: directional coefficient and isotropy indicator. The directional coefficient is determined from the directional deviation of sound intensity, where employing a normalization procedure taking account of the directional resolution of a receiver. On the other hand, the existing isotropy indicator is directly given through spherical harmonic expansion of sound intensity without the above normalization procedure. First, to resolve this inconsistency, we incorporate a similar normalization procedure into the definition of the isotropy indicator. Moreover, we revise the definition formulas of both indices in the light of mathematical applicability and physical comprehensibility. As a result, these modifications lead to an explicit correspondence between the two indices, providing a unified interpretation of them. Finally, to observe the behaviors of the two indices, we conduct numerical simulations of steady-state sound fields in a room with different absorption conditions using the fast multipole boundary element method. Numerical results demonstrate the trends of the revised indices, which suggests their validity for evaluating sound field diffuseness.

11:00

3aAAa6. Assessment of physicist Higini Arau-Puchades's career in architectural acoustics: Between scientific vocation and the profession of consultant in a peripheral country. Julián Álvarez Chaia (Universitat Politècnica de Catalunya, Campus Diagonal Sud, Edifici P. Av. Doctor Marañón, 44-50, Barcelona 08028, Spain, julian.alvarez@upc.edu), Ramon Graus (Universitat Politècnica de Catalunya, Barcelona, Spain), and Helena Martín-Nieva (Universidad de La Rioja, Barcelona, Spain)

Spanish physicist Higini Arau-Puchades (Barcelona, 1946) has been an example of professional achievement in a country with a tradition of acoustic studies that was not highly developed. His prestige was established during the years of the Spanish democratic transition, in which the country invested in the construction of a large network of auditoriums and theatres. In the 1980s, during the refurbishment of the Palau de la Música Catalana of Barcelona (1981-89), he grew as a professional in close contact with German consultant Lothar Cremer. Soon afterward, he published his perfected formula for calculating reverberation time (1988) and collaborated with architect Rafael Moneo in the design of the Auditori of Barcelona (1987-99) and the Kursaal of San Sebastián (1990-99). Moneo and Arau also participated in the consultation of 2002 on the Avery Fisher Hall of New York. The acoustics of the reconstructed Gran Teatre del Liceu of Barcelona (1994-99) opened the door to him for the renovation of the Teatro alla Scala of Milan (2004). His latest contributions have been focused on correcting concert halls using a type of diffuser that can increase the acoustic volume. This principle was applied brilliantly to Tonhalle St. Gallen (2010).

11:20

3aAAa7. Exploring conceptual soundscape of Derinkuyu Underground City. Sezin Nas (Interior Architecture and Environ. Design, Istanbul Galata Univ., Bereketzade Mahallesi Okçu Musa Caddesi No.: 1, Istanbul, Beyoglu 34421, Turkey, deryasezinnas@gmail.com), Konca Saher (Interior Architecture and Environ. Design, Kadir Has Univ., Istanbul, Fatih, Turkey), Saadet Aytis (Interior Architecture, Mimar Sinan Fine Arts Univ., Istanbul, Turkey), and Gürkan M. Mihçi (Visual Commun. Design, Herron School of Art + Design, Bloomington, IN)

This research aims to facilitate the experience of the cultural heritage of the ancient Derinkuyu Underground City, one of the largest ancient underground cities in the world, through audio-visual virtual reality (VR) representations of past life in the city. The objective of this study is to define the conceptual framework of the soundscape in the Derinkuyu Underground City based on soundscape standards. The focus is on three spaces: the church used for religious ceremonies, the living area for resting, and the kitchen for food preparation and service. The spatial functions of the church, kitchen, and living spaces were analyzed, sound sources were classified and the acoustic environment characterized, considering daily tasks, defining sound sources, establishing reverberation time values by measurements and simulations and describing absorption properties of surfaces. The results showed the limitations in description of context due to a lack of literature and historical records. The temporal ambiguity, interconnected spaces, and absence of clear boundaries in Derinkuyu Underground City created a timeless environment where the shared soundscape heightened auditory sensitivity, supporting both survival and security needs.

Session 3aAAb

Architectural Acoustics, Noise, Psychological and Physiological Acoustics, and Speech Communication: At the Intersection of Speech and Architecture III

Kenneth Good, Cochair
Architecture Acoust., Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17601

Evelyn Hoglund, Cochair
Speech and Hearing, Ohio State Univ., 104a Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210

Pasquale Bottalico, Cochair
Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820

Invited Papers

9:00

3aAAb1. Speech in cinema. Neil A. Shaw (Menlo Sci. Acoust., Inc., P.O. Box 1610, Topanga, CA 90290, menlo@ieee.org), Brian Vessa (Sony Pictures (retired), Los Angeles, CA), Charles “CJ Flynn (CinemaTestTools, Peille, Alpes Maritimes, France), and Woody Woodhall (Allied Post, Santa Monica, CA)

Talkies—motion pictures with speech synchronized to the picture—made their feature film debut on October 6, 1927, with the premier of *The Jazz Singer* (although it only had limited dialogue sequences, including Jolson’s “…You ain’t heard nothin’ yet!”). In 1929, the year the ASA was founded, the 2nd Academy awards were presented. Since then, soundtracks in motion pictures have progressed from a single monophonic track to immersive formats capable of reproducing 3-D sound fields, with dozens of sound objects, in auditoriums with different dimensions and quantities of loudspeakers. Speech, known as dialogue in cinema, is mixed with music and effects, to produce the movie soundtrack. While the perceived acoustics of the scenes are created using artificial reverberation during mixing, the dialogue itself is recorded in an acoustic space, either on a set or outdoors. Despite the best microphones and tools available to the sound mixer and cinema auditorium technical teams, the dialogue is often hard to understand in the local Bijou. How the dialogue is manipulated (and compromised), from the production recording to final mix is discussed, as is the importance of the local cinema’s audio B-chain and the room acoustics on the dialogue intelligibility.

9:20

3aAAb2. ASHRAE RP 1852 toward a unified metric for speech privacy in high-performance buildings: Speech level variation by office environment and communication type. Rewan Toubar (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., 1515 Saint-Catherine St. W, EV-0S3.412, Montreal, QC H3G 1S6, Canada, rewantoubar@yahoo.com), Joonhee Lee (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., Montreal, QC, Canada), and Roderick Mackenzie (Soft dB, Montreal, QC, Canada)

The loudness of speech is critical in predicting speech intelligibility and privacy in office environments. The surrounding environment can influence speech levels, necessitating accurate measurement in typical work settings to enhance predictions of speech privacy. Standardized speech levels and spectra, as outlined in ASTM or ANSI standards, can aid in predicting speech privacy or intelligibility. However, these data are collected in anechoic chambers with participants following scripted scenarios. This study presents a revised and updated examination of speech levels in two offices in Quebec, Canada, analyzing data from over 70 employees across different room types, communication scenarios, languages, and tasks. In open offices, desks with partitions showed higher speech levels (56 dBA) compared to those without partitions (52 dBA). Meeting rooms showed relatively consistent levels (52-54 dBA) regardless of size. Teleconference group meetings resulted in employees using slightly higher levels (54 dBA) compared to other communication methods within the same rooms (53 dBA). Statistical analysis revealed significant effects of office type, communication method, language, and task on speech levels. Individual variations in speech were more significant than office layout or communication methods. The observed variability in speech levels across different individual speakers and office settings suggests that current standardized methods for assessing speech privacy may need re-evaluation.

Contributed Papers

9:40

3aAAb3. Evaluating STC and other single number ratings of airborne sound insulation by comparison with Zwicker loudness. David W. Dong (Paul S. Veneklasen Res. Foundation, 11623 Talaud St., Cypress, CA 90630, info@veneklasenresearchfoundation.org), John LoVerde, Sunit Girdhar (Paul S. Veneklasen Res. Found., Cypress, CA), and Benjamin M. Shafer (PABCO Gypsum, Tacoma, WA)

One of the primary uses of single number ratings, such as STC (Standard Transmission Class), is to estimate subjective evaluations of airborne noise isolation, which are commonly determined by surveys or listening tests. Such testing showed that STC did not adequately describe the isolation of a partition compared to arithmetic averages, which helped lead to the development of Speech Privacy Class. Due to the challenges in conducting such tests, subjective data are not available for many wall types and sources of interest. It has been shown that calculated loudness can be used to assess single number ratings. This method is applied to a large dataset of laboratory wall tests and used to compare the perceived loudness with common single number ratings, including STC with and without the 8 dB rule. The effectiveness and applicability of single number ratings are discussed, and new ratings are considered.

10:00–10:20 Break

10:20

3aAAb4. Subjective and objective validation of a virtual reality system as a tool for studying speech intelligibility in architectural spaces. Angela Guastamacchia (Energy Dept., Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin 10129, Italy, angela.guastamacchia@polito.it), Riccardo G. Rosso, Giuseppina E. Puglisi (Energy Dept., Politecnico di Torino, Torino, Torino, Italy), Fabrizio Riente (Dept. of Electronics and Telecommunications, Politecnico di Torino, Turin, Italy), Louena Shtrepi (Energy Dept., Politecnico di Torino, Torino, Italy), Franco Pellerey (Dept. of Mathematical Sci., Politecnico di Torino, Turin, Italy), and Arianna Astolfi (Energy Dept., Politecnico di Torino, Torino, Italy)

Speech is essential for communication, but poor acoustics can disrupt it. Studying how indoor architectural features affect speech perception is key for boosting communication. Virtual Reality (VR) systems allow the auralization of realistic auditory scenarios, enabling and easing controlled studies of Speech Intelligibility (SI) in different architectural spaces. This study builds on previous research presenting and objectively validating a cost-effective VR setup—16 speakers synced with a VR headset—and evaluates its adequacy for perceived SI assessments. Thirteen subjects underwent SI tests in a reverberant lecture room and its VR reproduction, within five auditory scenarios varying in target speech spatial configurations and masking noise type. Speech Reception Threshold results indicate alignment between real and virtual settings for all scenarios. Subjective ratings further confirm a good coherence between real and virtual auditory experience, supporting the system validity.

10:40

3aAAb5. Integrating acoustic and non-acoustic sensor data in a working office environment. Anat Grant (DLR Group, 700 South Flower, 22nd Fl., Los Angeles, CA 90017, agrant@dlrgroup.com), Steph Ahrens (DLR Group, Omaha, NE), Paul McDonald (Sonitus Systems, Dublin, Ireland), Eoin King (Galway Sound Lab, Univ. of Galway, Galway, Ireland), and David Manley (DLR Group, Omaha, NE)

Green building standards like LEED and WELL promote sustainable design and occupant wellbeing, incorporating acoustic guidelines to ensure healthy workplace environments. A growing area of interest for acoustic research is soundscape studies, which examine human perception of sound in context. Soundscape studies combine acoustic and non-acoustic factors—such as perceived control and sense of engagement—and visual stimuli, including room dimensions, window ratio, worker density, and biophilia. This study will be conducted in a functioning office in Nebraska and integrates acoustic and non-acoustic data to evaluate indoor soundscapes. Continuous acoustic monitoring data, personality assessments, user surveys, and sentiment analysis from audio and photo diaries will be combined for a mixed-methods approach. The study seeks to visualize disruptions within the soundscape and examine the real and perceived impact on task performance and workspace preferences. By integrating diverse data sources, the study aims to provide a holistic understanding of how interior environments influence health, satisfaction, and workplace effectiveness. The objectives are to empower occupants to personalize their environments based on individual needs and work tasks, while guiding enterprises to prioritize strategic design improvements. This presentation outlines the study framework, data collection methods, and potential implications for enhancing workplace acoustics and overall performance.

11:00

3aAAb6. Recommendations of sound absorption areas for everyday architectural spaces. Tetsuya Sakuma (The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, sakuma@arch1.t.u-tokyo.ac.jp)

In the Architectural Institute of Japan, an acoustic standard and its design guideline of sound absorption are currently being established to ensure basic sound environment for a variety of everyday rooms, referring to DIN 18041. In the new standard, reasonable recommendations of the amounts of absorption will be incorporated with the indicator of equivalent absorption area per floor area. This indicator is a dimensionless quantity and corresponds to the sum of absorption coefficients of floor and ceiling if not taking the absorption of walls into account. Everyday rooms are subdivided into four types according to their absorption needs, which are determined by the loudness of generated noise and the need of quietness or speech transmission. Considering the balance of noise reduction and reverberation suppression, the recommended values of the indicator are formulated for the four room types as a function of average ceiling height. Specifically, as the ceiling height increases, noise reduction is strengthened while allowing the increase in reverberation to some extent. In addition, the recommended values are set to increase by 25% for the four room types, which corresponds to a 20% decrease in reverberation time and a 1 dB reduction of noise level.

11:20

3aAAb7. Acoustic influence of multi-resonator screens on speech signals in a virtual open-plan office. Giulia Fratoni (Dept. of Industrial Eng., Univ. of Bologna, Viale del Risorgimento 2, Bologna, BO 40136, Italy, giulia.fratoni2@unibo.it) and Dario D'Orazio (Dept. of Industrial Eng., Univ. of Bologna, Bologna, Italy)

Intelligible speech is one of the most distracting noises in workplaces. Sound-absorbing partitions attenuate speech signals, enhancing the workers' comfort and privacy. However, isolating their acoustic contribution from other sound-absorbing environmental variables, such as ceiling or wall treatments, is challenging. This study employs a wave-based room acoustic modeling approach to investigate the acoustic role of specific desk screens in a virtual open-plan office. A multi-resonator unit cell capable of attenuating the primary formants of voice signals was optimized through analytical models, finite-element simulations, and experimental data from 3-D-printed samples. The authors evaluated the acoustic performance of potential desk panels—constructed from unit cells' iteration—within the full-scale digital model of the office calibrated on in-field measurements. The study quantifies the proposed panels' resulting speech attenuation compared to traditional (glass and porous) screens.

11:40

3aAAb8. Mitigation of anthropic and mechanical noise for enhancing acoustic comfort in open-plan offices: A multi-objective decision method for treatment strategies based on ISO 22955 standard. Costanza Vittoria Fiorini (DIAEE - Dept. of Astronautical, Elec. and Energy Eng., "Sapienza" University of Rome, Sapienza Univ. of Rome, Via Eudossiana 18, Rome, Rome 06184, Italy, costanzavittoria.fiorini@uniroma1.it), Tarsitano Anna, and Andrea Vallati (DIAEE - Dept. of Astronautical, Elec. and Energy Eng., "Sapienza" University of Rome, Sapienza Univ. of Rome, Rome, Italy)

The research aims to assess acoustic comfort in office spaces, focusing on mutual disturbances between occupants and the contribution of HVAC systems for thermal comfort. A case study was conducted in an open office at Sapienza University of Rome. Acoustic descriptors, such as T60, STI, Das, and D2s, were evaluated through modeling and *in situ* measurements, considering various sound-absorbing materials like suspended ceilings, wall panels, and separation screens. The impact of HVAC power on these descriptors was also assessed. A validated model was used to create a training database for a neural network predicting acoustic descriptors based on room volume, absorption area, HVAC power, and occupancy. Post-intervention data were used to develop a predictive model to identify the optimal corrective solutions, aiming for ISO 22955 and UNI 11532-2 acoustic comfort levels while minimizing sound-absorbing material area. The model aids early-stage design by simplifying complex calculations.

WEDNESDAY MORNING, 21 MAY 2025 GALERIE 4, 9:00 A.M. TO 11:40 A.M.

Session 3aAB

Animal Bioacoustics: Arthropod Biotremology and Bioacoustics

Sebastian Oberst, Cochair
School of Mech. and Mechatronic Eng.; Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Lord St. 32-34, Sydney 2019, Australia

Joseph Lai, Cochair
School of Eng. and Technol., Univ. of New South Wales, School of Eng. and Technol., UNSW Canberra, Canberra 2600, Australia

Invited Papers

9:00

3aAB1. Vibroacoustic signals produced by flower visitors and their role in plant–insect interactions. Francesca Barbero (Life Sci. and Systems Biology, Univ. of Turin, Via Accademia Albertina 13, Torino 10123, Italy, francesca.barbero@unito.it), Simona Alberti, Lorenzo Bianco, Luca P. Casacci (Life Sci. and Systems Biology, Univ. of Turin, Torino, Italy), Jone Echeverria, Tomás J. Matus (Inst. for Integrative Systems Biology, Universitat de València-CSIC, Valencia, Spain), Abhishek R. Mohapatra (School of Mech. and Mechatronic Eng.; Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Sydney, New South Wales, Australia), David Navarro-Payá (Inst. for Integrative Systems Biology, Universitat de València-CSIC, Valencia, Spain), Can Nerse (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Sydney, New South Wales, Australia), Gastón Pizzio (Inst. for Integrative Systems Biology, Universitat de València-CSIC, Valencia, Spain), Ivan Sili (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Sydney, New South Wales, Australia), Maria R. Tucci (Life Sci. and Syst. Biol., Univ. of Turin, Torino, Italy), and Sebastian Oberst (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Sydney, New South Wales, Australia)

Flower visitors, including pollinators, produce characteristic sounds through flapping wing movements during flight. Recent research underscores the value of studying these acoustic signals to develop non-invasive, efficient tools for monitoring pollinator communities. Additionally, these sounds may provide key information to flowering plants, potentially influencing their resource allocation to attract pollinators, thus impacting their fitness. In this study, we investigated the acoustic properties of airborne sounds generated by recording different flying visitors to *Antirrhinum* flowers in the field. The audio recordings were annotated according to the observed flying behaviors and analyzed using nonlinear time-series analysis. We also conducted playback experiments to evaluate how plants respond to the buzzing sounds of insects. Our results reveal that distinct flying behaviors, such as hovering, landing, and takeoff, produce unique acoustic signatures. Furthermore, plants exhibit reactions to the vibroacoustic stimuli from pollinators, suggesting potentially adaptive responses. These findings provide valuable insight for developing passive acoustic monitoring tools for flying insects and may inspire further research in the field of plant–pollinator interactions. [Work supported by Research Grant RGP0003/2022 from HFSP https://doi.org/10.52044/HFSP.RGP00032022.pc.gr.153605.]

9:20

3aAB2. Deep learning technology to disrupt vibrational mating communications of invasive citrus psyllids, *Diaphorina citri* (Liviidae), aggregating in Florida citrus groves. Richard W. Mankin (Ctr. for Medical, Agricultural, and Veterinary Entomology, USDA ARS, 1700 SW 23rd Dr., Gainesville, FL 32608-1069, richard.mankin@usda.gov), Sruthi Sentil (Eng., Cornell Univ., Ithaca, NY), Zhihui Tian (Eng., Univ. of Florida, Gainesville, FL), and Seth McNeill (Eng., Embry Riddle Aeronautical Univ., Prescott, AZ)

Diaphorina citri Kuwayama (Hemiptera: Liviidae) is an invasive insect pest that causes Huanglongbing (HLB), a disease of devastating impact in citrus production worldwide. The mating process involves vibrational communications on citrus tree branches that can be artificially disrupted, enabling population reduction and better understanding of the mating behavior of pest insect psyllids in general. However, male *D. citri* vibrational signals cannot be distinguished from those of females by standard statistical methods, and it has been difficult to understand and disrupt mating aggregations in trees with high populations. We have developed deep learning methods to distinguish the vibrational signals of male psyllid signals from those of females and are developing disruptive devices that reduce citrus-tree pest populations. This is a report on initial development and testing of the system.

Contributed Paper

9:40

3aAB3. Applying biotremology to agricultural pest management: Mating disruption of blue-green sharpshooters, *Graphocephala atropunctata*. Jessica R. Briggs (United State Dept. of Agriculture, 9611 S Riverbend Ave., Parlier, CA 93648, jessica.briggs@usda.gov), Crystal Espindola, and Rodrigo Krugner (United State Dept. of Agriculture, Parlier, CA)

The blue-green sharpshooter (BGSS), *Graphocephala atropunctata*, poses a significant threat to California grape growers as a vector of the bacterium *Xylella fastidiosa*, which causes Pierce's disease in grapevines. Current population control methods have become less effective, prompting the need to explore new strategies, such as behavioral mate disruption. During courtship, male and female BGSS communicate through vibrational signals to assess and locate potential mates. The male initiates the interaction, and if the female is receptive, she responds with a distinct signal, leading to a coordinated duet between them. This study aimed to assess the feasibility of disrupting the duetting behavior via playing back female vibrational signals. Controlled behavioral trials were conducted, shortly after the male began signaling, a researcher would manually broadcast a female signal via a mini shaker. In the control group, no signals were broadcast. Results indicate broadcasting a female signal immediately following a male's signal reduced the likelihood of successful mating in BGSS to 19%, compared to 54% in the silent control treatment ($p = 0.014$). Our findings suggest that using vibrational signals for targeted mating disruption could be an environmentally friendly alternative for controlling arthropod pest populations.

Invited Papers

10:00

3aAB4. Non-invasive habitat evaluation through vibroscape monitoring. Rok Šturm (Dept. of Organisms and Ecosystem Res., National Inst. of Biology, Vecna pot 111, Ljubljana 1000, Slovenia, rok.sturm@nib.si), Juan José López Díez, Jernej Polajnar (Dept. of Organisms and Ecosystem Res., National Inst. of Biology, Ljubljana, Slovenia), Matija Marolt (Faculty of Comput. and Information Sci., Univ. of Ljubljana, Ljubljana, Slovenia), and Meta Virant-Doberlet (Dept. of Organisms and Ecosystem Res., National Inst. of Biology, Ljubljana, Slovenia)

The structure and dynamics of the biological components of the vibroscape were analysed at different temporal scales, ranging from diel variations to seasonal changes. Substrate vibrations were recorded from plant stems using laser vibrometers or accelerometers. Initial analyses of vibroscape recordings were annotated manually, with vibrational events manually assigned to vibrational signal types (VST) based on their temporal and spectral characteristics. Later, we began using automatic detection and classification of vibrational signals based on deep audio embeddings and transfer learning to increase the speed of analyses. Our models show approximately 75% precision in detecting the most common signal types observed in our recordings. The highest abundance of VSTs was observed during the day at the beginning of July (middle of summer). The overlap of these signals in the time and frequency domains was significantly smaller than would occur just by chance. This reveals a partitioning of the vibrational communication channel used by many arthropods in the meadow. We propose that extracting information from vibroscape recordings could be an effective non-invasive habitat evaluation approach, encompassing a higher percentage of living beings than acoustic monitoring alone, thereby introducing the field of ecotremology.

10:20–10:40 Break

Contributed Papers

10:40

3aAB5. Snapping shrimp snaps and passive acoustic monitoring of coral reefs. Hannah Gower (Health and Life Sci., Univ. of Exeter, Stocker Rd., Exeter, Devon EX4 4PY, United Kingdom, H.Gower2@exeter.ac.uk)

Increased monitoring of coral reef ecosystems is essential to establish effective management and conservation strategies, yet traditional survey methods are typically both cost and time intensive, often requiring specialised equipment and highly trained individuals. Passive acoustic monitoring (PAM) has recently emerged as a cost-effective alternative, capable of collecting long-term continuous datasets while requiring relatively low manual effort. This study aims to develop PAM methodology on coral reefs, focusing on sounds from snapping shrimp (Order: Caridae; Family: Alpheidae), typically the most ubiquitous sound producers on reefs. These reef-associated crustaceans are closely tied to and influenced by local environmental factors suggesting their sound production could act as an indicator of reef health. Sound recordings were taken using low-cost HydroMoth acoustic loggers at artificial reef sites around Lizard Island, Australia, in both 2023

and 2024, before and after the fifth mass coral bleaching event. Changes in reef condition associated with the disturbance event will be compared to changes in sounds detected, investigating the relationship between the two. Identifying snapping shrimp as an indicator species would streamline acoustic data processing and allow for rapid large-scale classification of reef condition, accelerating essential conservation and management decisions as reefs face an ever-growing number of threats.

11:00

3aAB6. Development of a micro-exciter to mimic insect vibration signals and the effect of the substrate on the playback response. Sebastian Oberst (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Lord St. 32-34, Sydney, New South Wales 2019, Australia, sebastian.oberst@uts.edu.au), Farzad Tofigh (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Sydney, New South Wales, Australia), Joseph Lai (School of Eng. and Technol., Univ. of New South Wales, Canberra, Australian Capital Territory, Australia), Mark Mankowski (US Dept. of Agriculture, Starkville, MS), Rachel Arango, and Grant Kirker (US Dept. of Agriculture, Madison, WI)

In bioassays that involve the playback of recorded insect vibration signals using conventional commercial shakers, two issues need to be addressed. First, common shaker systems do not have the capability of producing insect excitations that are in the micro or milli Newton range. Second, recorded insect signals are the response of the substrate excited by the vibrations produced by the insect, but the influence of the substrate is rarely considered by the biotremology community. Here, we, therefore, present the development of a micro-excitation system across three stages: (1) an initial Arduino-based with a piezoelectric beam element; (2) a Raspberry Pi-controlled piezoelectric buzzer with fast-acting amplification circuit, compared to commercial stacked piezo actuators; and (3) an in-house developed microactuator consisting of a microcontroller and a piezoelectric actuator (MiAC-S). The evolution of the amplification circuit and its integration into MiAC-S will be described. The MiAC-S will be validated. Finally, we demonstrate the effect of the substrate on the playback of an impulse response signal compared to using a direct impulse as an excitation signal. The results indicate that the vibration response should not be used for playback of insect signals on substrates as it produces results inconsistent with theoretical expectations, thereby increasing the uncertainty and reducing the control quality of bioassays.

11:20

3aAB7. Statistical analyses of ant and termite walking response signals. Sebastian Oberst (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Lord St. 32-34, Sydney, New South Wales 2019, Australia, sebastian.oberst@uts.edu.au), Joseph Lai (School of Eng. and Technol., Univ. of New South Wales, Canberra, Australian Capital Territory, Australia), and Theo Evans (Univ. of Western Australia, Perth, Western Australia, Australia)

Eusocial insects have attracted significant research interest due to their unique physiology and ability to detect and respond to various signals, enabling efficient and often highly specialised communication within their colonies. Ants and termites have been locked into a predator-prey arms race for millions of years. Recent research reveals that termites can detect predatory ants by sensing their footsteps. It is, therefore, of great interest to determine the characteristics and differences of the vibrations induced by the walking of ants and termites. Here, we conduct a statistical analysis of six ants and ten termite species' video motion and footstep response vibrations. The motion analysis reveals that although termites are as heavy and active as ants, they move less erratically and quickly fall into a pattern following each other. The median speed relative to the body length shows a moderate negative correlation in ants, but a weak correlation in termites. By isolating single-step vibration response using nonlinear filtering and an enveloping function, the average amplitude and decay times are determined. Compared with termites, ants are found to excite more harmonics and exhibit more power in their vibrations which correlates with their average body weight, the average response amplitudes, and decay times but this is not observed for termites.

Session 3aAO

Acoustical Oceanography: Bioacoustic Attenuation Spectroscopy

Orest Diachok, Cochair
Poseidon Sound, 3272 Fox Mill Rd., Oakton, VA 22124

Christopher Feuillade, Cochair
Inst. of Phys., Pontifical Catholic University of Chile, Vicuna Mackenna 4860, Santiago 7820436, Chile

Chair's Introduction—9:55

Invited Papers

10:00

3aAO1. The effect of attenuation from fish on long-range active and passive acoustic sensing in the ocean. Daniel Duane (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, daniel.m.duane.civ@us.navy.mil) and Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Attenuation from fish can reduce the intensity of acoustic signals and significantly decrease detection range for active and passive sensing in the ocean. Formulations for predicting attenuation from fish depend on the accurate characterization of population density and spatial distribution of fish groups along long-range propagation paths, which is difficult to achieve using conventional survey methods. Here, Ocean Acoustic Waveguide Remote Sensing (OAWRS) is used to instantaneously image massive Norwegian herring shoals that stretch for thousands of square kilometers and simultaneously measure attenuation from these shoals within the active OAWRS transmissions, as well as attenuation to ship-radiated tonals. Reductions in signal intensity are predicted using a normal-mode-based analytical theory for acoustic propagation through inhomogeneities in an ocean waveguide. The predictions of the waveguide attenuation formulation are in agreement with measured reductions, where the position, size, and population density of the fish groups are characterized using OAWRS imagery as well as *in situ* echosounder measurements of the specific shoals occluding the propagation path. Negligible attenuation was observed in previous OAWRS surveys because the frequency of the acoustic signals was sufficiently far from the swimbladder resonance peak of the shoaling fish or the packing densities of the shoals were not sufficiently high.

10:20

3aAO2. Frequency-dependent sound scattering and extinction by fish schools. Christopher Feuillade (Inst. of Phys., Pontifical Catholic Univ. of Chile, Vicuna Mackenna 4860, Santiago, Metropolitan Region 7820436, Chile, chris.feuillade@gmail.com)

A low-frequency fish school acoustic scattering model, incorporating multiple scattering between fish, and also wave interference of their scattered fields, has been used in both back- and forward-scattering configurations to study fish school size and shape and fish abundance. There is a critical physical distinction between back- and forward-scattering which affects data analysis using this modeling approach. Frequency dependent interferences, which modify the back scattering amplitude, are absent in the forward scattering case. Studies of sound extinction by schools, based upon the inversion of transmission data, necessitate a forward scattering paradigm. Analysis of the 1995 Modal Lion experiment [Diachok, J. Acoust. Soc. Am. **105**, 2107–2128 (1999)] shows that this then leads to larger estimates of fish abundance than back-scattering analyses. A comparison with an effective medium approach shows good agreement in the forward direction, where interferences are minimal, indicating both methods are applicable for estimating fish abundance. A model comparison with back scattering data [Holliday, J. Acoust. Soc. Am. **51**, 1322–1332 (1972)] shows the effective medium method diverging strongly from the data when $k\ (=2\pi/\lambda) < 4s$ (where s is the average spacing between neighboring fish), while the scattering model continues to represent the data accurately. [Research supported by ONR.]

10:40

3aAO3. Bioacoustic attenuation spectroscopy in the Santa Barbara Channel and Yellow Sea: Inference of anchovy parameters from broadband transmission loss measurements. Orest Diachok (Poseidon Sound, 3272 Fox Mill Rd., Oakton, VA 22124, orestdia@aol.com) and Altan Turgut (Naval Res. Lab., Washington, DC)

Synoptic measurements of broadband transmission loss (TL), sound speed profiles, geo-acoustic properties of the bottom, and distributions of lengths, depths, and species of fish permitted isolation of the effects of bio-acoustic attenuation due to fish, the bottom and solitons on TL in the Santa Barbara Channel (SBC). Sound speed and geo-alpha in the bottom were derived from *chirp* sonar measurements. Differences between TL measurements and PE calculations of TL during a period, which was not affected by solitons, provided measures of excess attenuation due to bio-alpha versus frequency. Trawls revealed that 10.5 cm long anchovies were dominant. Resonances were evident at 1.8 kHz at 18 m and 1.1 kHz at 7 m due to dispersed, one year old, 10.5 cm anchovies, and at 0.5 kHz due to schools of 10.5 cm anchovies. A BAS experiment in the Yellow Sea revealed resonances due to dispersed 1-year old, 11 cm

anchovies, the dominant species/length in the Yellow Sea, and schools of 11 cm anchovies. The resonance frequencies were essentially the same (within 0.1 kHz) as in the SBC. Inferred depths of dispersed anchovies are consistent with concurrent echo-sounder data and historical measurements of preferred depths (Sobradillo, *et al*. 2021).

Contributed Paper

11:00

3aAO4. Simulation of bio-acoustic absorption and its effect on volume reverberation. Blake E. Simon (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu)

A significant density of pelagic fish and fish schools in underwater environments can lead to higher acoustic reverberation, thus affecting applications, such as environmental remote sensing or fish abundance estimation. Fish aggregations can also increase absorptivity in the water column and increase transmission loss. This phenomenon is described by bio-alpha, i.e., the attenuation coefficient within the scattering layer. Theoretical models and historical databases exist which describe volume scattering strength in underwater environments of interest. Additionally, effects of bio-alpha on transmission loss have been studied [for example, see Diachok, J. Acoust. Soc. Am. **105**, 2107–2128 (1999)]. However, less consideration has been given to quantifying the effect of bio-alpha on volume reverberation. Presented here is a simple volume reverberation model which directly accounts for biological absorptivity due to bio-alpha in the scattering layer. Representative cases for different frequencies and environments are presented. Based on the known diurnal behavior of many species of pelagic fish, it is shown in nighttime conditions that reverberation is high but decays quickly whereas during daytime reverberation is relatively lower with a relatively smaller rate of decay. [Work supported by the ARL:UT Internal Research and Development Program.]

11:20–11:40 Panel Discussion

WEDNESDAY MORNING, 21 MAY 2025 — STUDIO FOYER, 10:00 A.M. TO 12:00 NOON

Session 3aBA

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Award Poster Session

Kenneth B. Bader, Cochair
Dept. of Radiol., Univ. of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637

Kevin J. Haworth, Cochair
Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

All posters will be on display and all authors will be at their posters from 10:00 a.m. to 12:00 noon

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee.

Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers. All entries will be on display, and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

1aBA6. Functional ultrasound localization microscopy on freely moving rats
Student author: Yike Wang

1pBA2. Uncoupling a bi-disperse microbubble population for decreased ULM acquisition times: A proof-of-concept study
Student author: Giulia Tuccio

1pBA4. Gas vesicle expression in stem cells and the potential as a biomarker application
Student author: John Kim

2aBAb3. Second-harmonic generation in focused shear wave beams in tissue-like media with arbitrary amplitude shading at the source
Student author: Philip G. Kaufinger

2aBAb5. Three-dimensional vibration-controlled transient elastography in human subjects with myofascial trigger pain points
Student author: Maryam Satarpour

2pBAa4. Effect of the point spread function: Deconvolution strategies for accurate spatiotemporal cavitation doses by passive acoustic mapping
Student author: Abigail Collins

2pBAa5. Practical considerations in the calibration of array probes for quantitative cavitation imaging: Lessons learned from 16 attempts
Student author: Darcy Michael Dunn-Lawless

2pBAa9. Multiple signal classification in the time domain for passive cavitation imaging
Student author: Nathan Caso

2pBAb6. Finite element characterization of the effect of flow on acoustic radiation force-induced waves in blood vessels
Student author: Charles Capron

2pBAb7. Comparative assessment of arterial wave velocities measured by arterial dispersion ultrasound vibrometry (ADUV) and clinical arterial stiffness metrics
Student author: Md Aktharuzzaman

3pBA6. Effect of focused ultrasound-induced heat and cavitation on methicillin-resistant Staphylococcus aureus viability and the role of bacterial structural features in susceptibility to cavitation
Student author: Pratik A. Ambekar

4aBAa2. Targeted contrast enhanced ultrasound imaging to detect molecular changes in the preeclamptic rat placenta
Student author: Lili Shi

4pBAa1. Model-dependent modulation of radiotherapeutic efficacy with oxygen microbubbles and ultrasound
Student author: Phillip Durham

4pBAa10. Evaluating *in vivo* tissue stiffness of capillaries using microbubbles under an ultrasound field
Student author: Sae Jang

4pBAa11. Cavitation bubble nuclei in healthy vs. cancerous 3D cell cultures
Student author: Ferdousi Sabera Rawnaque

4pBAa6. Dose dependent safety and renal ultrasound contrast kinetics of oxygen microbubbles in healthy dogs
Student author: Jacob A. Mattern

4pBAb10. Simulating focused ultrasound propagation through heterogeneous biomedical materials with volume-surface integral equation methods and hierarchical matrix compression
Student author: Alberto Almuna Morales

4pBAb11. Ultrasonic evaluation of internal features in additively manufactured SS316L: Impact of process-induced microstructure heterogeneity
Student author: Harshith Kumar Adepu

4pBAb12. Quantifying the effect of cooling conditions on 3D printed PLA meso-structure and acoustic properties
Student author: Partha Pratim Pandit

4pBAb13. Development of a dynamic acoustic phantom for simulating human tissue
Student author: Amirhossein Yazdkhasti

4pBAb2. Estimating vascular density by using quantitative ultrasound in microbubble enhanced vessel networks
Student author: Parniyan Norouzzadeh

4pBAb6. Desktop 3D-printed metamaterials for MHz-range ultrasound devices
Student author: Alireza Tadibi

4pBAb8. Leveraging the parametric array effect for transcranial focused ultrasound interventions
Student author: Pradosh Pritam Dash

4pBAb9. Simulating performance of a through-transmit aberration correction method for transcranial focused ultrasound
Student author: Cooper L. Donovan

5aBA2. A novel quantitative ultrasound metric for osteoporosis diagnosis using instantaneous phase
Student author: Hugh E. Ferguson

5aBA3. Quantitative ultrasound-based characterization of placental microstructure during preeclampsia
Student author: Andrew Markel

5aBA5. Motion correction for improved ultrasound measurement of hyoid bone movements during swallowing
Student author: Nicholas S. Schoenleb

5aBA8. Ultrasonic characterization of the transmural structure of human scalps tissue
Student author: Catherine N. Prabish

5pBA1. Ultrasonic reflection and attenuation by the bone cortex—Implications for backscatter measurements of cancellous bone
Student author: Keith T. Hoffmeister

5pBA2. Enhancing ultrasound tissue characterization with the double Nakagami distribution model
Student author: Ladan Yazdani

5pBA4. Passive cavitation detection analysis of color Doppler twinkling from polymethyl methacrylate
Student author: Benjamin Wood

WEDNESDAY MORNING, 21 MAY 2025 STUDIO 6, 9:00 A.M. TO 11:20 A.M.

Session 3aCA

Computational Acoustics, Physical Acoustics, Acoustical Oceanography, and Underwater Acoustics: Parabolic Equation Methods Across Acoustics

Jennifer Cooper, Cochair
Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723

Michelle E. Swearingen, Cochair
Const. Eng. Res. Lab., U.S. Army ERDC, P.O. Box 9005, Champaign, IL 61826

Philippe Blanc-Benon, Cochair
CNRS, 36 Ave. Guy de Collongue, Ecully, 69134, France

Invited Papers

9:00

3aCA1. Parabolic models and computational approaches in high-power therapeutic ultrasound. Vera A. Khokhlova (Phys., Univ. of Washington, Seattle, USA/Moscow State Univ., Moscow 119991, va.khokhlova@gmail.com), Pavel B. Rosnitskiy (Dept. of Medicine, Univ. of Washington, Moscow, Russian Federation), Petr V. Yuldashev, Maria M. Karzova (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), and Oleg A. Sapozhnikov (Univ. of Washington, Seattle, USA/Moscow State Univ., Seattle, WA)

Ultrasound sources employed in high-power therapeutic medical systems are typically strongly focused, with f# = 1 or even lower. The initial design of such sources, acoustic characterization of existing ones, and corresponding treatment planning is a challenging problem, especially for therapies that rely on the formation of pressure shocks at the focus. One-way propagation models based on evolutionary differential equations have been shown to provide an effective way to simplify the analysis and obtain accurate solutions. Among them, the most popular is the nonlinear parabolic Khokhlov-Zabolotskaya (KZ) equation. The parabolic approximation inherently assumes a quasi-plane nature of the wave, so its direct applicability is limited to weakly focused beams, in practice, with f#>2.

However, an accurate analysis of nonlinear phenomena even in strongly focused beams based on the KZ equation in combination with an equivalent source model is possible by taking into account that in the focal region, where the wave is most affected by nonlinearity, the beam structure is close to a quasi-plane wave. Other approaches include transformation of the coordinate system following the geometry of beam focusing and extension to wide-angle parabolic models. This talk will overview the numerical tools that have been recently developed based on the proposed extensions.

9:20

3aCA2. Narrow- and wide-angle parabolic equations with arbitrary variations in the sound speed and medium velocity. Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), D. Keith Wilson, Michael Muhlestein (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Didier Dragna, and Jules Colas (LMFA, Ecole Centrale de Lyon, Ecully, France)

Acoustic wave propagation in inhomogeneous media such as the atmosphere and ocean can be efficiently modelled with parabolic equations. This presentation overviews recently derived narrow- and wide-angle parabolic equations for sound propagation in motionless and moving media, along with algorithms for their numerical implementation. These parabolic equations preserve the phase of the sound wave and are valid for arbitrary variations in the sound speed and arbitrary (subsonic) Mach number of the medium velocity. Within the ranges of their applicability, the parabolic equations considered exactly describe sound propagation in stratified moving media. These features are particularly important for long-range multipath sound propagation when the phase increments along different arrivals should be calculated accurately. Despite their generality, the narrow- and wide-angle parabolic equations are relatively simple. Moreover, they can be efficiently solved with available Crank-Nicholson numerical techniques. Example calculations are provided that demonstrate the numerical implementation of the narrow- and wide-angle parabolic equations and their accuracy as compared to parabolic equations based on the effective sound speed approximation.

Contributed Papers

9:40

3aCA3. Consequences of using the parabolic equation and nonlinear progressive wave equation in low-density atmospheres. Michelle E. Swearingen (U.S. Army Eng Res. and Development Ctr., P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil), Michael J. White, and Benjamin F. Racelis (U.S. Army Eng. Res. and Development Ctr., Champaign, IL)

Infrasound can travel up into the highest reaches of the earth's atmosphere. At these altitudes, the air density reduces to a value on the same order as the perturbation in density of a propagating wave. We review the numerical effects of the density profile for linear acoustics within a standard PE and consider the influences of the density profile on waves using an implementation of the Nonlinear Progressive Equation. [Approved for Public Release: Distribution is Unlimited.]

10:00–10:20 Break

10:20

3aCA4. Numerical simulation for three dimensions infrasound propagation in shadow zone using a one-way approach. Antoine Verdier (Institut Jean le Rond d'Alembert, Sorbonne Univ., 4 Pl. Jussieu, Paris 75005, France, antoine.verdier@sorbonne-universite.fr), Régis Marchiano (Institut Jean le Rond d'Alembert, Sorbonne Univ., Paris, France), Pierre Sochala (CEA, DAM, DIF, Arpajon, France), and Olaf Gainville (CEA, DAM, DIF, Bruyères-le-Châtel, France)

The propagation of infrasound through the atmosphere is influenced by factors such as temperature and wind speed gradients, topography, and geometry of the source. These factors can create shadow zones where geometric acoustic approximations are no longer valid. For instance, during the 2007 Carancas meteorite impact, the closest infrasound station was located in a shadow zone, and conventional ray-tracing models were unable to account for signals detected at this station. To address this challenge, our study proposes a 3-D numerical method that reconstructs the observed signals without relying on geometric acoustics assumptions, while avoiding the computational expense of full direct numerical simulations. Our approach is based on the FLHOWARD3D model, which employs a unidirectional approximation for nonlinear acoustic wave propagation in heterogeneous media. To take advantage of the unidirectional approximation, a domain coupling strategy was developed and rigorously validated. The method's performance in shadow zones is first evaluated by comparing it with analytical solutions. Subsequently, the method is applied to the Carancas scenario, incorporating both the nonlinear characteristics of the source and atmospheric heterogeneities. This work opens up prospects for improved infrasound prediction in shadow zones and enables the inference of key properties related to the source and propagation medium.

10:40

3aCA5. Atmospheric sound transmission loss at the coast: Modeling turbulence. Sophie Arruza (Dept. of Eng., East Carolina Univ., E 5th St., Greenville, NC 27858, arruzas22@students.ecu.edu), Matthew Stengrim (Dept. of Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, Diego Turo, John Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC)

This work presents progress in an ongoing effort to effectively represent turbulence in a numerical model of atmospheric sound propagation in a near-shore environment. Excess attenuation is predicted with a parabolic equation solver that incorporates turbulence as well as the combined effects of wind and temperature profiles on the sound speed profile. Literature-based turbulence parameters are used to inform model inputs. These previously reported turbulence metrics are compared with turbulence parameters derived from scanning Doppler LIDAR wind profiling measurements. A set of acoustic transmission loss measurements made with concurrent wind, temperature, and water surface roughness measurements provide ground truth for model output comparison. [Work supported by Office of Naval Research Award Nos. N00014-22-1-2492 and N00014-24-1-2437.]

11:00

3aCA6. Evaluation of long range acoustic model predictions using Monin-Obukhov similarity theory air temperature profiles. Matthew Stengrim (Eng., East Carolina Univ., 1000 East Fifth St., Greenville, NC 27858, stengrimm19@students.ecu.edu), Andrea Vecchiotti (Eng., East Carolina Univ., Washington, DC), Jeff Foeller, Sophie Arruza (Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

Refraction of sound in the atmosphere is largely governed by wind and air temperature. However, modeling sound propagation over long distances usually requires making a number of assumptions pertaining to the state of the atmosphere. In the near surface layer, Monin-Obukhov Similarity Theory (MOST) enables estimations of wind, air temperature, and humidity profiles with very little measurement input. This work examines how the assumption of MOST air temperature profile can affect acoustic model predictions over a range of 1 km. Over one years' worth of air temperature

profile, measurements over a water and marsh grass surface have been collected. Measurements were carried out by two arrays consisting of seven air temperature loggers spaced in 1 m intervals up to 7 m in elevation. Sound speed profiles based on both measured and modeled air temperature are implemented in a Crank-Nicholson Parabolic Equation (CNPE) solver. Predictions of sound pressure level made between each implementation of the CNPE model are compared over a wide range of weather conditions. [Work supported by Office of Naval Research Award N00014-22-1-2492.]

WEDNESDAY MORNING, 21 MAY 2025 BALCONY N, 9:00 A.M. TO 11:40 A.M.

Session 3aEA

Engineering Acoustics: Recording and Processing of Higher-Order Spatial Audio

Gary W. Elko, Cochair
mh acoustics, 25A Summit Ave., Summit, NJ 07901

Filippo Fazi, Cochair
ISVR, Univ. of Southampton, University Rd., Southampton SO171BJ, United Kingdom

Invited Papers

9:00

3aEA1. Sound field synthesis method using distributed spherical microphone arrays based on spherical harmonics. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp)

Sound field synthesis is one of the core techniques for realizing advanced virtual reality systems. The development of micro-electromechanical systems technology has enabled the use of compact spherical microphone arrays (SMAs) with numerous microphones to capture accurate sound field. While an SMA typically needs to be placed at the listening point, this placement is not always feasible in real-world scenarios. To address this issue, several methods have been proposed to reproduce sound field using distributed SMAs, without requiring them to be positioned at the listening point. One conventional technique involves translating the spherical harmonic (SH) center to the listening point by converting the expansion coefficients obtained from each SMA. However, the accuracy of the reconstructed sound field is often insufficient. In this presentation, we introduce our proposed sound field synthesis methods using distributed SMAs. In the methods, secondary sound pressures are calculated from the signals recorded by the SMAs. The sound pressure at the listening point is then reconstructed from these secondary sound pressures using SH-based approaches, such as the Auditory Display based on Virtual Sphere (ADVISE) model. The simulation results indicate that the proposed methods can synthesize sound field even when the listening point is far from SMAs.

9:20

3aEA2. Sound field diffuseness estimation based on the cross-spectral analysis of ambisonic signals. Filippo Fazi (ISVR, Univ. of Southampton, University Rd., Southampton, Hampshire SO171BJ, United Kingdom, Filippo.Fazi@soton.ac.uk), Nara Hahn, Mihai Orita, Bogdan Bacila, and Philip Nelson (ISVR, Univ. of Southampton, Southampton, United Kingdom)

A new method is presented for estimating the diffuseness of a sound field using the cross-spectral matrix of the Higher-Order Ambisonics signals captured with a microphone array. The diffuseness, which depends on both the isotropy and coherence of the sound waves/sources that constitute the sound field, is estimated by analysing the spread of the eigenvalues of the cross-spectral matrix over a range of frequencies. This method shares some similarities with the COMEDIE technique proposed by Epain and Jin in 2016, the main difference being the use of the cross-spectral matrix instead of the time-domain cross-correlation matrix analysed in COMEDIE. The proposed frequency-domain approach overcomes some of the limitations of the time-domain analysis, such as the overestimation of the diffuseness caused by correlated sources that are not time-aligned at the measurement position.

9:40

3aEA3. Sparse recovery beamforming using a combined linear and spherical array. Craig Jin (School of Elec. and Comput. Eng., The Univ. of Sydney, Maze Crescent, Bldg. J03, Sydney, New South Wales 2006, Australia, craig.jin@sydney.edu.au) and Shunxi Xu (School of Elec. and Comput. Eng., The Univ. of Sydney, Sydney, New South Wales, Australia)

Sparse recovery is a relatively well understood technique for spatial sound field analyses. In this work, we re-formulate sparse recovery as a beamforming method and provide a unified framework for both linear and spherical arrays. We provide a perspective that indicates sparse recovery beamforming is more suited to near-field analyses for linear arrays and far-field analyses for spherical arrays.

We, therefore, explore the dual use and combination of a linear and spherical array for sparse recovery beamforming and spatial sound field analyses. We analyse various sound fields using this approach and highlight where the combination of arrays provides utility.

10:00–10:20 Break

10:20

3aEA4. Comparison of processing methods for rigid spherical microphone arrays in spatial and spherical-harmonic domains for soundfield reconstruction. AMY BASTINE (School of Eng., Australian National Univ., 34 Marcus Clarke St., Academie House Unit 110, Canberra, Australian Capital Territory 2601, Australia, amy.bastine@anu.edu.au), Thushara D. Abhayapala, Prasanga Samarasinghe, and Jihui (Aimee) Zhang (School of Eng., Australian National Univ., Canberra, Australian Capital Territory, Australia)

Rigid Spherical Microphone Arrays (SMAs), such as commercially available Eigenmikes, are widely used for capturing 3-D sound fields. Various array processing methods have been developed to enhance the spatial extent and accuracy of soundfield reconstructions, particularly for virtual navigation applications. The traditional approach employs spherical-harmonic basis functions to decompose array measurements into the Higher-Order Ambisonics (HOAs) format, enabling soundfield reconstruction within the constraints of truncation order. Alternatively, measurements can be directly decomposed using spatial basis functions of plane-wave sources, point-sources, or mixed-wave sources via inverse filtering and compressive sensing techniques, exploiting the diversity introduced by the scattering properties of rigid SMAs. Hybrid methods also enable spatial basis decomposition from HOAs. Recently, physics-guided machine-learning models, such as the point-neuron framework, have emerged to ensure strict adherence of the equivalent source decompositions to the physical laws governed by the fundamental wave equation. While these methods have demonstrated success in specific scenarios, a unified evaluation is needed to compare their performance comprehensively. This paper examines their efficacy in terms of reconstruction accuracy, computational efficiency, and robustness to measurement noise and reverberation, highlighting trade-offs to guide practical applications in soundfield navigation.

10:40

3aEA5. Calibration requirements for Higher-Order Ambisonic (HOA) spatial microphone arrays. Gary W. Elko (mh acoustics, 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh acoustics, Fairfax, VT)

Spherical microphone arrays used for Higher-Order Ambisonics (HOA) spatial soundfield pickup are based on the spherical harmonic decomposition of the sound field at the measurement location. Practical design of spherical microphone arrays is based on limiting the number microphones to a reasonable number (<100) and minimizing spatial aliasing artifacts at higher frequencies due to discrete sampling positions when the average microphone spacing exceeds $1/2$ the acoustic wavelength. These two design objectives lead to a design where the HOA components are formed as estimates of higher-order spatial pressure differentials that are computed at frequencies where the acoustic wavelength is much larger than the size of the array (superdirectionality). It is well known that superdirectional arrays are sensitive to sensor calibration mismatch and self-noise. We will show simulation results for the magnitude and phase matching requirements needed to realize spherical HOA arrays as a function of the frequency, the number of microphones, and the array size.

Contributed Paper

11:00

3aEA6. A fly-inspired multi directional MEMS microphone. Xiaoyu Niu (Elec. and Comput. Eng., The Univ. of Texas at Austin, 2501 Speedway, EER 4.822, Austin, TX 78712, xyniu@utexas.edu), Donghwan Kim (Silicon Audio, Inc., Austin, TX), Zihuan Liu (Elec. and Comput. Eng., The University of Texas at Austin, Austin, TX), Yuqi Meng (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Ehsan Vatankhah (Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), and Neal A. Hall (ECE, Univ. of Texas, Austin, Austin, TX)

Inspired by the hearing mechanism of the fly *Ormia Ochracea*, we demonstrated a MEMS diaphragm capable of pressure and pressure gradient sensing in two orthogonal directions. The diaphragm is circular and suspended by multiple spring arms along its circumference. Each spring contains a piezoelectric film on its top surface. The structure has many orthogonal vibration modes. These modes are used for the concurrent measurement of sound pressure and pressure gradient along two orthogonal axes by way of summing and subtracting signals generated by the various sensing ports. Vibration modes of the MEMS are measured by applying broadband chirp voltage waveforms to a piezoelectric sensing port while observing the resulting motion using a scanning laser Doppler vibrometer (LDV). The device was then packaged for acoustic measurements by integrating each of several ports with a high input impedance Integrated Circuit Amplifier (ICA). Acoustical sensitivity and directivity were measured in a walk-in anechoic chamber. Compared with previous fly-inspired microphones, the presented prototype is the first with three measurands (p_0, dp/dx, and dp/dy) derived simultaneously from a single diaphragm.

Invited Paper

11:20

3aEA7. Higher-order ambisonics recordings in primary equatorial rainforests—A listening experience. David Monacchi (Sapienza Univ. of Rome, Via in Selci, 49, Roma, RM 00184, Italy, david.monacchi@gmail.com)

The project “Fragments of Extinction” has been working since 1998 at the crossroads between ecoacoustics research, technological innovation and sound art, recording the sound of the oldest and most biodiverse ecosystems of primary tropical forests in the Amazon, Africa, and South-East Asia. Through advanced higher-order ambisonics recording technologies, employed in these remote and extreme environments, the project is building an extensive sound archive of entire circadian cycles. These 24-h long spherical soundscape portraits are presented to audiences in the form of full-periphonic installations, to foster awareness about the extinction crisis and the

urgency of heritagization of the last sanctuaries of evolution, which high systemic integrity is surprisingly clear in their sonic processes. This lecture will outline the aims and challenges of the last expedition in Borneo in February 2023, which represented the most important recording campaign of the project, into—possibly—the biologically oldest tropical forest on Earth. 3 Terabytes of recordings have been carried out, for the first time, with a 64-channel microphone array, thus constituting the first attempt of its kind for storing the high-resolution spatial properties of a vanishing acoustic habitat. Depending on the hosting venue and the available multichannel system the recordings could be presented as a unique ambisonics 2D or 3D listening experience of an impressive dusk chorus sequence.

WEDNESDAY MORNING, 21 MAY 2025 SALON F/G, 11:00 A.M. TO 12:00 NOON

Session 3aED

Education in Acoustics: Education in Acoustics Prize Lecture

Shima Abadi, Chair
Univ. of Washington, 185 Stevens Way, Paul Allen Center – Room AE100R, Seattle, WA 98195

Chair's Introduction—11:00

Invited Paper

11:05

3aED1. Teaching acoustics as an engineering elective. David R. Dowling (Naval Architecture and Marine Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, drd@umich.edu)

Acoustics is a wonderful elective subject to teach because interested students arrive on the first day of class with two or more decades of experience operating an acoustic source (their voice) and utilizing multiple acoustic receivers (their ears). Thus, their curiosity and intuition are ripe for exploitation and can even be used to motivate the development of mathematical skills. My path to research and instruction in acoustics does not include any formal training in the subject. Instead, I encountered and learned acoustics at a pace and in manner of my choosing through research, consulting, and teaching. Thus, this lecture begins with a graphical retrospective of my career path punctuated by relevant dates and numbers. This career summary is followed by the statement and description of three main concepts that have guided my 30 + years of engineering science instruction: (i) let the truth be your only story, (ii) know your students and what motivates them, and (iii) engage in student-centric teaching. Descriptions, explanations, and examples related to these three concepts are presented and include: setting the tone of the course on the first day, surveying the students to determine their interests, and review of a few of my favorite homework problems.

Session 3aIDa

Interdisciplinary: Plenary Lecture: Selective Listening in Music: From Psychoacoustical Principles to Hearing Device Evaluation

Stefan Weinzierl, Chair
Einsteinufer 17c, Sekr. EN-8, Berlin 10587, Germany

Chair's Introduction—8:00

Invited Paper

8:05

3aIDa1. Selective listening in music: From psychoacoustical principles to hearing device evaluation. Kai Siedenburg (Inst. of Signal Processing and Speech Commun., Graz Univ. of Technol., Küpkersweg 74, Oldenburg 26129, Germany, kai.siedenburg@uol.de)

Music typically consists of a flurry of sounds from multiple instruments or voices that overlap in time and frequency. To infer musical structure, listeners, thus, need to select and track sounds from mixtures—a process also known as auditory scene analysis (ASA). ASA is determined by stimulus- and listener-specific properties: While certain instruments or voices naturally stand out in musical mixtures, individual listeners vary in their ability levels to hear out specific elements from mixtures. In the recent work, our team explored the psychoacoustical underpinnings of selective listening in music. In experiments with realistic musical material from popular music, this work identified acoustical features at the basis of vocal salience and specifically highlighted the role of pitch modulation and spectral distinctiveness. Selective listening in orchestral music turned out to be highly dependent on musical context and target sound but less affected by spatial acoustical variables. Using an efficient adaptive test approach, we observed individual differences in selective listening as a function of musical training, age, and hearing loss. The approach was further used to evaluate the performance of hearing devices, providing insight into their effectiveness in enhancing selective listening. Overall, our results underscore the need to develop tailored auditory solutions for listeners with diverse hearing profiles.

Session 3aIDb

Interdisciplinary: Student Council: Hot Topics in Acoustics

Chirag Gokani, Cochair
Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Heui Young Park, Cochair
Acoustics, The Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

May Pik Yu Chan, Cochair
Dept. of Linguistics, University of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104-6228

Natalie Kukshtel, Cochair
Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543

Chair's Introduction—10:55

Invited Papers

11:00

3aIDb1. From geoacoustic inversion to seabed tomography using a distributed network of sources and receivers. Julien Bonnel (Appl. Ocean Sci. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Ariel Vardi (Appl. Ocean Sci. and Eng., MIT-WHOI Joint Program in Oceanogr., Woods Hole & Cambridge, MA), John Leonard (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Understanding and predicting acoustic propagation at sea is critical for many marine applications, from noise pollution forecasts to underwater warfare. To do so in coastal waters requires knowledge of the seabed geoacoustic properties; estimating those from ocean acoustic data is called geoacoustic inversion. Historically, geoacoustic inversion considers acoustic propagation between a fixed (or linearly moving) source and a receiver (or array of receivers), usually leading to the estimation of a depth-dependent geoacoustic profile, assumed to be representative of the propagation track. In this talk, we will show how low-cost instrumentation and advanced signal processing methods (warping time-frequency analysis, trans-dimensional inversion, and machine learning) enable estimation of the spatial variability of seabed geoacoustic properties. Several examples will be presented, all based on data collected on the New England Mud Patches during the Seabed Characterization Experiments. We will notably illustrate how the proposed methods enable characterization of the spatial variability of the muddy seabed sediments at the scale of a single mud patch, as well as the inter-comparison of mud properties between several distant mud patches. [Work supported by the Office of Naval Research.]

11:20

3aIDb2. Liquid based tunable lenses – from concept to application. Sina Rostami (Phys., Univ. of Mississippi, 108 Lewis Hall, P.O. Box 1848, University, MS 38677, srostami@go.olemiss.edu) and Joel Mobley (Phys. and Astronomy, Univ. of Mississippi, University, MS)

Quasi-planar stepped lenses (QSLs) based on phasing have proven to perform closely to conventional refractive lenses for the generation of both focused and limited diffraction beams. These types of lenses have enabled applications, such as droplet extraction, trapping, and manipulation, as well as limited diffraction wavepacket generation. Ultrasonic beams formed in this manner have proven to closely match with simulations. These lenses function similarly to phased arrays but lack tunability and are restricted to a single frequency, which is inherent to their design. More recently, an approach using liquid based phasing has been developed that can be tuned to generate different beam types and focal depths while also enabling multi-frequency operation. In addition, it has a planar surface which can facilitate improved coupling to surfaces in relation to QSLs. The liquid lens generates the required phase modulation through the variation in speeds of sound in saltwater solutions. This talk traces the development of the liquid lens approach starting from the QSLs to the current designs. In addition to describing the current progress, we look ahead to the future prospects of these devices, including applications that may be of practical interest.

11:40

3aIDb3. Focused shear wave beams for enhanced soft tissue elastography. John M. Cormack (Div. of Cardiology, Dept. of Med., Univ. of Pittsburgh School of Medicine, Pittsburgh, PA 15261, jmc345@pitt.edu), Yu-Hsuan Chao (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Branch T. Archer (Chandra Family Dept. of Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), Philip G. Kaufinger (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX), Hansol O. Lee, Jaideep Behari (Dept. of Medicine, Univ. of Pittsburgh School of Medicine, Pittsburgh, PA), Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kang Kim (Dept. of Bioeng. and Med., Univ. of Pittsburgh, Pittsburgh, PA), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

The elastic response of soft tissues at audio frequencies is closely associated with disease-related changes in tissue microstructure and function. The propagation speed of shear waves in tissue, which is related to the shear modulus, is used to map tissue stiffness noninvasively, for example to assess breast lesion malignancy and liver disease severity. Conventional approaches for shear wave excitation are based on the acoustic radiation force or vibration of a small piston on the skin, both of which generate small-amplitude, diverging shear waves, and are limited in their ability for penetration to the required depths, especially in the assessment of liver stiffness because liver disease is often associated with obesity. Our group conceived and realized experimentally focused shear wave beams, generated by vibration of a concave piston source, which converge toward the measurement region, thereby maintaining high amplitude at the required depths. An overview is presented of recent progress in the application of focused shear wave elastography in the context of fatty liver disease assessment. Recent advances in the prediction of nonlinear effects in shear wave beam propagation will also be discussed. [P.G.K. was supported by the Chester M. McKinney Graduate Fellowship in Acoustics at ARL:UT.]

WEDNESDAY MORNING, 21 MAY 2025 GALERIE 5, 9:15 A.M. TO 11:40 A.M.

Session 3aNS

Noise: History of Acoustics and Evolution of Sound Measurement

Walter A. Montano, Cochair
Acoustics Research, ARQUICUST, Luis Clavarino 1227, Gualeguaychu E2820BSG, Argentina

Brandon Cudequest, Cochair
Threshold Acoustics, 141 W Jackson Blvd. Ste. 2080, Chicago, IL 60604

Chair's Introduction—9:15

Invited Papers

9:20

3aNS1. The evolution of noise awareness disseminated through the media until the first ASA anti-noise campaign. Walter A. Montano (Acoust. Res., ARQUICUST, Luis Clavarino 1227, Gualeguaychu, Entre Rios E2820BSG, Argentina, wmontano@arquicust.com)

Today, there are several worldwide initiatives to raise awareness of the noise problems in people and fauna, but little is known about the first efforts for its advocacy. Since journalism through the media became observers of reality, in the case of noise annoyance due to the lack of medical knowledge, the earliest complaints described the hardships that afflicted people, and only at the end of the 19th century clinical and technological criteria appeared. Therefore, also through some journalists, the media echoed the legal claims made by people, to invite them affected by noise to create societies against noise. Until 1930, a few initiatives were presented with the goal of promoting anti-noise actions, and after the emblematic report of noise levels in New York City recorded with a sound level meter, anti-noise campaigns took on another dimension. Members of the newly created ASA formed a group that dedicated themselves to investigating the problems of noise from technology, and later they realized that the best policy was to create an intermediate society, and thus, in 1940 the "National Noise Abatement Council" was born, whose first activity was the "Noise Abatement Week." In this paper, a brief summary of these early initiatives will be made, to focus on the work carried out by those ASA Members, some of them ASA founders.

9:40

3aNS2. Before the decibel—Sound measurements—Physical, aural, electrical. Noral D. Stewart (Retired, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, norals2020@sacnc.com) and Walter A. Montano (Acoust. Res., ARQUICUST, Gualeguaychu, Entre Rios, Argentina)

Efforts and methods to measure absolute or relative quantities of sound prior to establishment of the decibel will be discussed. Starting with Rayleigh, we will discuss early methods of physical measurement, aural observation methods at Riverbank Laboratories, early microphone development the origin of the audiometer, early contributions of Wente at Western Electric, the first practical portable direct measuring sound meter, and early efforts to evaluate community noise using audiometers.

10:00

3aNS3. The decibel and beyond—Early progress in sound measurement and analysis. Noral D. Stewart (Retired, 7330 Chapel Hill Rd., Ste. 201, Raleigh, NC 27607, norals2020@sacnc.com) and Walter A. Montano (Acoust. Res., ARQUICUST, Gualeguaychu, Entre Rios, Argentina)

The origins of the decibel and the progress in sound measurement instrumentation that followed in the next two decades will be discussed, including early sound level meters and microphones, standardization and early standardized sound level meters, calibration, early frequency analyzers, and the first compact sound level meters.

10:20–10:40 Break

10:40

3aNS4. Human analyzers and Helmholtz resonators: A prehistory of the sound level meter. Zachary Weiss (Noise Control Eng., LLC, 85 Rangeway Rd., Bldg. 2, Billerica, MA 01862, zfweiss@noise-control.com)

Though Helmholtz resonators now find widespread application within silencers and sensitive acoustic spaces, they bear their name from a scientist who pursued neither of these ends. Rather, Hermann Von Helmholtz employed his namesake resonators as a convienent tool to analyze individual tonal components of complex musical sounds, essentially positioning the human observer as a signal analyzer augmented by a bank of tuned resonators comprising variable filters within the signal chain. In the lead up to the development of the sound level meter, both of these roles were ultimately replaced by electroacoustic technologies. However, this was not immediate, and echoes of Helmholtz's apparatus were seen in such measurement techniques as Wallace Sabine's aural determination of reverberation time, the resonant enclosue of the Tucker hot-wire microphone, and binaural methods of sound ranging during World War I. This presentation traces the physical and human antecedents to the modern electronic sound level meter, concluding with a few specialized cases where they can still find use today.

Contributed Papers

11:00

3aNS5. Measurement of transmission loss of symmetric systems by using a 2-microphone standing wave tube. Guochenhao Song (Ray W. Herrick Lab, Purdue Univ., West Lafayette, IN), Tongyang Shi (Institue of Acoust. Chinese Acad. of Sci., Beijing, China), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S. Russell St., West Lafayette, IN 47907-2099, bolton@purdue.edu)

Transmission loss is a key property for evaluating the sound insulation performance of acoustic systems in applications ranging from architectural acoustics to automotive noise control. In ASTM E2611, a 4-microphone impedance tube setup is standardized for measuring transmission loss based on transfer matrix theory. By comparison with the 2-microphone impedance procedure, the 4-microphone measurement requires a more expensive setup and a more complex testing procedure. In contrast, the 2-microphone setup standardized in ASTM E1050 is simpler and more efficient but measures only the surface normal impedance, which relates to sound absorption rather than sound insulation. To bridge this gap, for symmetric acoustic systems, a closed-form expression has been derived in the present work to compute transmission loss based on two surface normal impedance measurements with different backing conditions, thus offering a practical and efficient alternative to the 4-microphone procedure. The accuracy of this approach has been validated through comparisons with transmission loss predictions obtained from both simulations and real measurements. This method simplifies the transmission loss testing procedure and so expands the application of the existing 2-microphone impedance tube configuration.

11:20

3aNS6. A brief history of the standing wave tube. J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907-2099, bolton@purdue.edu)

The standing wave tube is widely used to measure the absorption coefficient and surface normal impedance of materials, most often in accordance with ASTM standard E1050, in which a so-called two-microphone procedure is described. However, the standing wave tube method has a long and interesting history, having been used to measure acoustic absorption coefficients for more than 120 years, its first development being credited by several authors to Tuma in 1902. The initial pure tone procedures were entirely non-electronic and relied on sensitive flames or discs to quantify sound pressure. The first electronic implementation appeared in the 1920s and was based on using a "hot wire microphone" to measure the sound field. The pure tone procedure involving a traversing probe tube found its ultimate expression in work by Beranek (1947), Brüel and Kjaer (1955), and Champoux and Stinson (1991). The "modern" two-microphone, broadband procedure was introduced by Seybert and Ross in 1977 and has since been universally adopted as an efficient alternative to the pure tone procedure. The presentation will conclude with a description of the extension of the two-microphone procedure to four-microphone, transmission loss measurements, and recent multi-microphone procedures that extend the upper frequency limit of plane wave measurements.

Session 3aPAa

Physical Acoustics and Structural Acoustics and Vibration: Mesoscopics in Acoustics and Elasticity I

John Yoritomo, Cochair
Acoustics, Naval Research Lab., 4555 Overlook Ave. SW, Washington, D.C. 20375

Marco Scalerandi, Cochair
Dept. of Appl. Sci. Technol., Politecnico di Torino - Disat - Corso Duca degli Abruzzi 24, Torino 10134, Italy

Sandrine T. Rakotonarivo, Cochair
Aix-Marseille Univ. and Lab. of Mech. and Acoust., LMA - UMR 7031 AMU - CNRS - Centrale Marseille, 4 impasse Nikola Tesla, Marseille 13453, France

Invited Papers

9:00

3aPAa1. Nonlinear elastic properties of intact and fractured Westerly granite. Jacques Riviere (Penn State Univ., 212 Earth Eng. Sci. Bldg., University Park, PA 16802, jvr5626@psu.edu), Hermann Klinghammer, Derek Elsworth, and Parisa Shokouhi (Penn State Univ., University Park, PA)

The objective of this work is to unravel the physical mechanisms responsible for the nonlinear elastic response of rocks and, in particular, to understand the role of fractures. Cylindrical samples of Westerly granite, either intact or transected perpendicular to the long-axis by a mated fracture, are subjected to a pump-probe protocol. This protocol applies axial and confining stress oscillations of amplitude $\pm$0.5 MPa and pump frequency $f_0 \approx 8.3$ mHz, while the elastic state of the sample is probed with S-wave, 500-kHz transducers at each end of the sample. The quasi-static axial stress is also varied stepwise, between 1 and 17 MPa, both upward and downward, while the confining pressure is kept constant (4 MPa). Two nonlinear elastic parameters are extracted: the offset R_0, tracks the average wave-speed change induced by the oscillation, while R_1 tracks wavespeed changes at f_0. Congruent with a previous study, and counterintuitively, R_1 is smaller for the fractured samples. R_0 is positive (average stiffening) during axial stress up-steps, and negative (average softening) during down steps. Moreover, this effect is more pronounced for the fractured samples, that is, R_0 is larger in absolute value for fractured samples. These results are discussed in light of previous measurements conducted on fractured rocks.

9:20

3aPAa2. Ultrasound scattering and absorption in a cracked rock. Hao Zhou (College of Geophys., Chengdu Univ. of Technol., Chengdu, China), Xiaoping Jia (Institut Langevin, ESPCI Paris, 1 rue Jussieu, Paris 75005, France, xiaoping.jia@espci.fr), Li-Yun Fu (School of Geosciences, China Univ. of Petroleum (East), Qingdao, China), and Arnaud Tourin (Institut Langevin, ESPCI Paris, Paris, France)

Ultrasonic experiments were performed on room-dry and wet shale samples—composites consisting of sand grains and clay with extremely low porosity and permeability, and exhibiting a layered texture. Wetting caused significant shear wave scattering and a reduction in shear wave velocity, likely due to moisture-induced microcracks from the swelling of clay minerals. To analyze these effects, we used the Monte Carlo method to solve the radiative transfer equation for finite-size samples, extracting the scattering mean-free path and absorption length from ballistic and coda waves. The frequency-dependent scattering mean-free path indicated that wet shale could be modeled as a medium with randomly oriented and located cracks, characterized by a low crack density and power-law crack length distribution. This was supported by 2-D finite-difference simulations of shear wave scattering in a similar cracked medium. As confining pressure increased, microcracks in the shale closed, leading to higher shear wave velocity and scattering mean-free path. Interestingly, the absorption length was found to decrease under pressure, which contrasts with the behavior typically observed in sandstones (e.g., squirt flow dissipation). This result aligns with observations in weakly wet granular media, where water films trapped by asperities at grain surfaces play a key role.

9:40

3aPAa3. Coda-wave interferometric and lock-in measurements of slow dynamic nonlinearity. Richard Weaver (Dept. of Phys., Univ. of Illinois, 1110 W Green St., Urbana, IL 61801, r-weaver@illinois.edu) and John Yoritomo (Acoust., Naval Res. Lab, Washington, DC)

We study the universal linear-in-log-time recoveries of material moduli in rock and cements after minor conditioning (by strains of order 10^{-6}), recoveries that proceed in the laboratory over decades in time from msecs to minutes. Digital-lock-in and coda-wave-interferometric procedures reveal changes in ultrasonic wave-speed with high precision and fine time resolution. These methods are particularly useful for separating the intrinsic material contributions to early time relaxation spectra from the obfuscating effects of

conditioning ring-down. We apply these procedures to (1) examine very early and very late times, where recovery must deviate from log-linearity, (2) elucidate dependence on conditioning strains, and strain rates, (3) measure the effect of conditioning durations and thresholds, and (4) detect dependence on environment. We confirm reports of a conditioning threshold in Berea sandstone, at about one micro-strain, below which there is no slow dynamics and above which SD is linear in pump strain. Our measurements challenge reports of roll-offs in relaxation spectra at early times. They also fail to support speculations of dependence on conditioning strain *rate*—at least in our parameter regime with strains of order 10^{-6} and pump frequencies of order kHz. [Work supported by the US DOE Award No. DE-SC0021056.]

10:00–10:20 Break

10:20

3aPAa4. Coupled synchrotron x-ray imaging and static/dynamic acoustoelastic testing to understand the nonlinear elasticity of fractured rock under stress. Parisa Shokouhi (Eng. Sci. and Mech., The Penn State Univ., 212 EES Bldg., University Park, PA 16802, pxs990@psu.edu), Evan Bozek, Prabhav Borate (Penn State Univ., University Park, PA), Mark Rivers (Univ. of Chicago, Chicago, IL), Derek Elsworth, and Jacques Riviere (Penn State Univ., University Park, PA)

Seismic characterization of fractures requires quantitative mapping between *in-situ* fracture characteristics and the corresponding elastodynamic properties. Laboratory experiments coupling X-ray imaging and ultrasonic testing are conducted to quantify how bulk/fracture deformation, porosity, and contact area/distribution influence the elastodynamic response of stressed fractured rock. The stress-dependency of the ultrasonic wave speed, amplitude, and frequency are analyzed in relation to the changes in true contact area, contact size distribution, and fracture aperture gleaned from high resolution *in situ* 3-D and 2-D synchrotron X-ray imaging during quasi-static and dynamic loading of Westerly granite and Berea sandstone samples with saw-cut fractures normal to the loading axis. A subset of fractured Westerly granite samples is thermally damaged to decouple the influence of bulk damage and fault on the ultrasonic signatures. We confirm connections between ultrasonic transmission and contact area and find a correspondence between the transmitted wave frequency and fracture aperture. We observe the lowest acoustic nonlinearity parameter ß for the sample with the largest estimated contact area; the two samples with similar contact areas but different contact size distributions yield similar ß irrespective of the bulk stiffness. These findings have implications in improving the interpretability of seismic attributes in relation to subsurface fracture properties.

Contributed Papers

10:40

3aPAa5. A multirelaxation model for conditioning and relaxation of modulus in sandstones introducing fast and slow dynamics contributions. Marco Scalerandi (Dept. of Appl. Sci. and Technol., Politecnico di Torino- Disat-Corso Duca degli Abruzzi 24, Torino 10134, Italy, MARCO.SCALERANDI@POLITO.IT), Jan Kober, and Radovan Zeman (Czech Acad. of Sci., Inst. of Thermomechanics, Prague, Czechia)

Elastic modulus of sandstones exhibits a non trivial dependence on strain. Wavespeed decreases when a strain is applied, with a complex hysteretic/butterfly dependence on strain amplitude and the softening is not instantaneous but increases progressively with time (conditioning). On a time scale of several minutes, velocity recovers the original value when strain is removed (relaxation). Here, we propose a multirelaxation model, which allows reproducing hysteresis, cumulative conditioning and relaxation. We introduce a non-equilibrium strain, given as the superposition of contributions which are evolving on different time scales (from very fast to very slow) and weighted according to a given distribution of relaxation times. Each component evolves in time due to a conditioning term and a relaxation term, acting simultaneously. The balance between them determines the state of the material at a given time and strain amplitude. The ratio of relaxation times and wave period allows to identify which components contribute to fast and which to slow dynamics. We validate the model by comparison with experimental data and apply it to obtain predictions for cases, which are difficult to realize in experiments.

11:00

3aPAa6. Ultrasonic propagation and slow dynamic nonlinear elasticity in saturated bead packs. John Yoritomo (Acoust., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, john.yoritomo@nrl.navy.mil), Benjamin Dzikowicz (Acoust., Naval Res. Lab., Washington, DC), and Grace Burkholder (Phys., Bethel Univ., St. Paul, MN)

Nonlinear Mesoscopic Elasticity (NME) refers to a class of materials that display nonlinear elastic behaviors that cannot be captured by the classical theory, where the stress field is related to powers of the strain field. The designation "mesoscopic" comes from the fact that NME materials are composed of grain elements that are large compared to the atomic scale (and thus, can be contrasted with materials like aluminum or quartz that have atomic elasticity) but small compared to sample size. Of the NME behaviors, slow dynamic nonlinear elasticity, or simply slow dynamics (SD), has received particular attention. There is a prevailing thought that moisture plays a key role in SD. Some models predict that the slow dynamic response should disappear in a saturated material. To test this prediction, we probe saturated glass bead packs with ultrasound and assess changes in the bead pack using coda wave interferometry. SD is induced using a low frequency harmonic pump. Our results are compared with similar measurements of dry bead packs. We find that saturated bead packs also exhibit SD and, with some caveats, the SD response is unaffected by saturation.

11:20

3aPAa7. Contactless characterization of the adhesion of thermoplastic welded composite assemblies with Lamb waves based inverse method. Loïc Girardot (Laboratoire d'Acoustique de l'Université du Mans / IRT Jules Verne, 1 Ave. Olivier Messiaen, Le Mans 72000, France, loic.girardot@live.fr), Mourad Bentahar, Silvio Montresor (Laboratoire d'Acoustique de l'Université du Mans, Le Mans Univ., Le Mans, France), and Nicolas Terrien (IRT Jules Verne, Nantes, France)

Interest in detecting defects and evaluating the mechanical strength of assemblies using non-destructive methods has grown considerably during the last years. In this context, the present work describes a characterization method based on the propagation of Lamb waves to quantify the adhesion of ThermoPlastic (TP) welded composite assemblies. The propagating ultrasonic guided waves within the multilayered anisotropic composites are excited and detected with the help of Air-Coupled ultrasonic Transducers (ACT). Then, an adapted analytical model is used to investigate the influence of the mechanical properties corresponding to the welding layer. Theoretical and experimental data are compared by considering the wavenumber dispersion curves, where guided modes are considered over different frequency domains. Finally, the welding layer stiffness properties are estimated using an optimization method on various welded samples of different qualities.

Session 3aPAb

Physical Acoustics and Computational Acoustics: Session in Honor of Richard Raspet I

Gregory W. Lyons, Cochair
gregwlyons@gmail.com

Keith Attenborough, Cochair
School of Eng. and Innovation, The Open Univ., Milton Keynes MK7 6AA, United Kingdom

Jeremy Webster, Cochair
Los Alamos National Labs, LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544

W. C. K. Alberts, Cochair
CCDC-Army Research Laboratory, 2800 Powder Mill Rd., Adelphi, MD 20783

Chair's Introduction—9:15

Invited Papers

9:20

3aPAb1. Richard Raspet, the construction engineering research laboratory, and beyond. Victor W. Sparrow (Penn State, 411 ECoRE Bldg., 556 White Course Dr., University Park, PA 16802, vws1@psu.edu)

Richard Raspet was a consummate researcher and mentor to many students at the University of Illinois, Urbana-Champaign during his employment at the Construction Engineering Research Laboratory, now known as a laboratory of the U.S. Army Engineer Research and Development Center (ERDC). This talk will provide an overview of some of the projects that Dr. Raspet undertook during the 1980s, with a particular focus on his work with students. Time permitting, the discussion will extend to his work at the University of Mississippi on sonic booms and thermoacoustics in the 1990s. The present author is honored to have been the first of Richard's many Ph.D. students, and his personal influence was just as profound and lasting as that of his many technical contributions. [Work supported by the Penn State College of Engineering through its United Technologies Corporation Professorship in Acoustics.]

9:40

3aPAb2. Richard Raspet—Professor, advisor, mentor, and friend. Mark W. Sprague (Phys., East Carolina Univ., M.S. 563, Greenville, NC 27858, spraguem@ecu.edu)

I worked and studied under Richard Raspet in the late 1980s and early 1990s at the University of Mississippi. This presentation will recount memories of my time with Rich at Ole Miss. The Physical Acoustics Research Group at Ole Miss consisted of Raspet, Henry Bass, and James Sabatier. We studied long-range atmospheric sound propagation and did both computational and experimental studies. Our laboratory was located in the deteriorating Old Band Building at Ole Miss, but we moved to the brand-new National Center for Physical Acoustics facility in the early 1990s. Rich was a respected physics faculty member, who taught graduate courses in Acoustics, Classical Mechanics, and Statistical Mechanics. Graduate students wanted him to teach their courses, especially the difficult ones. His instruction was rigorous and thorough. He always found a way for students to relate to obscure theoretical material. Rich studied many aspects of sound propagation including propagation modeling using the fast field program, normal modes, the parabolic equation, and ray theory. He studied complex ground impedance, atmospheric turbulence, wind and temperature gradients, and their effects on sound propagation in the atmosphere. He also studied thermoacoustics and made several contributions to that field.

Contributed Paper

10:00

3aPAb3. Outdoor sound propagation studies by Rich Raspet in the 1980's. Michael J. White (U.S. Army Engineer Res. and Development Ctr., US Army ERDC/CERL, 2902 Farber Dr., Champaign, IL 61822-1072, michael.j.white@usace.army.mil)

In 1979, Rich Raspet began working at the U.S. Army Construction Engineering Research Laboratory to help mitigate the noise produced by the Army. He and colleagues conducted experiments in sound production by helicopters and explosive charges and experiments in sound propagation at several US and NATO military training installations. His experiences whetted his appetite for the phenomena of shock waves, refraction of sound by wind and temperature gradients, reflection of sound by porous ground impedance, diffraction by turbulence, and much more, joining the National Center for Physical Acoustics in 1987. Results from one of his experiments started this presenter on a similar journey.

10:20–10:40 Break

Invited Papers

10:40

3aPAb4. Bikes, beers, and some biomedical acoustics: My interactions with Rich Raspet. Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu)

Rich Raspet was a dedicated researcher and teacher in the field of physical acoustics. This talk will recall a former student's interactions with Rich and a final homework problem related to the treatment of venous clots. Improved outcomes have been observed when clot-busting thrombolytic drugs are combined with histotripsy, a focused ultrasound therapy that breaks down tissue via bubble activity. To gain insight into the mechanisms of this combination approach, an *in silico* model of bubble/thrombolytic/clot interaction was developed. The methods to calculate bubble dynamics and drug/clot interaction were known, but additional insights were needed to model delivery of the drug. After many dead ends, the student reached out to his former teacher once again for guidance. Rich was able to distill the problem down to its essential elements, which led to adoption of the perfusion-diffusion equation to complete the calculation. Findings from the resultant model indicated histotripsy increases the penetration of thrombolytic drug into the clot relative to just the drug alone. These analyses were validated by *in vitro* measurements and suggest that the observed enhancement in drug activity was the result of histotripsy-induces changes to tissue perfusivity.

11:00

3aPAb5. Two perspectives on Rich Raspet. Cecille Labuda (Phys. and Astronomy, Univ. of Mississippi, 108 Lewis Hall, University, MS 38677, cpembert@olemiss.edu)

In this talk, I will present two perspectives of Rich Raspet; one from the point of view as a graduate student and the other from the point of view as a faculty colleague. As a PhD student in the University of Mississippi Physics program, he was my instructor in two courses and gave me advice whenever I requested it. Later, as a faculty member in the same department, I became Rich's colleague where I continued to learn much from him. I will reflect on what it was like to know Rich in both modes and what I learned from him as a student and as a faculty colleague.

11:20

3aPAb6. Contributions by Richard Raspet on two aspects of the acoustics of rigid framed porous media. Keith Attenborough (School of Eng. and Innovation, The Open Univ., Milton Keynes MK7 6AA, United Kingdom, keith.attenborough@open.ac.uk)

Two papers co-authored by Richard Raspet are about the surface impedance of grounds with exponential porosity profiles [R. Raspet and J. M. Sabatier, J. Acoust. Soc. Am. 99, 147–152 (1996)] and sound propagation in porous media containing main capillary pores with smaller pores in their walls [W. P. Arnott, J. M. Sabatier, and R. Raspet, J. Acoust. Soc. Am. 90, 3299–3306 (1991)]. The variable porosity ground impedance model enables better fits to outdoor short-range measurements over many types of grassland than the widely used semi-empirical single parameter model. Moreover, it has been found to represent the surface impedance of artificial surfaces such as felt. The theory for pores with porous walls which was developed in the context of thermoacoustic stack designs is the forerunner of several designs involving slit like pores with microporous walls, sinusoidal walls or periodically varying widths in the somewhat different context of creating useful sound absorbers by additive manufacture.

11:40

3aPAb7. Richard Raspet's legacy in porous media and thermoacoustics. William Slaton (Phys., Astronomy, & Eng., The Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72034, wvslaton@uca.edu)

Richard Raspet significantly advanced the fields of sound propagation in porous media and thermoacoustics, bridging these domains through innovative research. His seminal work unified Biot's theoretical framework for porous media with Rott's thermoacoustic theory, allowing comprehensive modeling of sound propagation and energy flow in pores with various geometries. Raspet's insights revealed the interplay between thermoviscous dissipation functions and established a universal methodology for applying porous media models to thermoacoustics. This framework supported advancements in thermoacoustic devices, including engines and refrigerators, by enabling the use of random and fibrous media such as fiberglass and metallic foams. Furthermore, Raspet pioneered the exploration of evaporation–condensation effects, demonstrating their critical role in enhancing thermoacoustic efficiency and energy density, leading to patented innovations. His contributions laid a foundational understanding that continues to propel both theoretical and applied research in acoustics, earning him recognition as a transformative figure in physical acoustics.

Session 3aSA

Structural Acoustics and Vibration and Musical Acoustics: Friction Acoustics

Trevor Jerome, Cochair
Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., Bldg 3 #329, West Bethesda, MD 20817

Kathryn Matlack, Cochair
Univ. of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801

Vasileios Chatziioannou, Cochair
Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria

Eric Rokni, Cochair
Physics, High Point Univ., One University Parkway, High Point, NC 27268

Chair's Introduction—8:55

Invited Papers

9:00

3aSA1. Developing acoustic emission and machine learning techniques for condition monitoring of rubbing contacts. Robert Gutierrez (Mech. Eng., Imperial College London, Imperial College London, Exhibition Rd., London SW7 2AZ, United Kingdom, robert.gutierrez16@imperial.ac.uk) and Tom Reddyhoff (Mech. Eng., Imperial College London, London, United Kingdom)

Acoustic emission (AE) waves are high frequency mechanical stress waves, which propagate through a material from some source mechanism. AE has been shown to be a rich source of tribological information and as it is non-invasive and relatively low-cost, it is well suited for condition monitoring of rubbing contacts. However, it is not widely implemented within industry, as AE signals are difficult to interpret and the relationship between friction behaviour and sound is highly complex. Herein, AE and coefficient of friction (CoF) have been recorded from oscillating ball-on-disc sliding tests. AE spectrogram and histogram data are used to train machine learning (ML) regression models to predict CoF. Strong predictions are given for new tests at different conditions. Aluminium scratch tests are also presented, demonstrating correlations between AE and scratch image features, such as cracks, giving an insight into the wear mechanisms producing AE. Finally, sliding tests with different oil additives have been explored. AE data combined with CNN techniques are used to build classification models for identifying oil composition from on AE data. The classification models show a high accuracy and precision. Overall, AE has proven to be a powerful condition monitoring tool, with potential to bring large economic savings to industry by improving machine efficiency and preventing failures/unnecessary maintenance.

9:20

3aSA2. Elasto-plastic friction modeling of bow–string interaction. Ewa Matusiak (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1/II, Wien 1030, Austria, matusiak@mdw.ac.at), Vasileios Chatziioannou (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Vienna, Austria), and Maarten van Walstijn (SARC, School of Electronics, Elec. Eng., and Comput. Sci., Queen's Univ. Belfast, Belfast, United Kingdom)

One approach for simulating bow-string interaction is to use an elasto-plastic friction model. In this study, such a model is refined to guarantee passivity, and a stable numerical scheme is derived that inherits the energy balance of the underlying continuous model. The model is used to simulate the motion of a vibrating string under frictional excitation by a finite-width bow, incorporating both bow hair compliance and string torsional motion. Using inverse modeling to obtain parameter values for the elasto-plastic friction model, it is possible to reconstruct measured transient signals. By examining Guettler diagrams of simulated data, a dependency of the underlying frictional profile on the bow force and acceleration emerges. This observation is in agreement with prior research indicating that static and dynamic friction coefficients vary within a Guettler diagram and points to the fact that the elasto-plastic model should be modified to account for that dynamical nature. [This research was funded in whole or in part by the Austrian Science Fund (FWF) [10.55776/P34852]]

9:40

3aSA3. Friction force reconstruction in bowed cello strings. Alessio Lampis (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, lampis@mdw.ac.at), Ewa Matusiak, Alexander Mayer, and Vasileios Chatziioannou (Dept. of Music Acoust. (IWK), mdw - Univ. of Music and Performing Arts Vienna, Vienna, Austria)

When playing a bowed-string instrument, a complex interaction occurs between the rosin-coated bow hair and the string. The resulting frictional force that excites the string has interested researchers since the first observations of Helmholtz and Raman. Despite extensive research, we still lack a clear formulation of the physical laws governing this force. Notably, Schumacher, Garoff, and Woodhouse developed an inverse calculation method that reconstructs the friction force at the bowing point by measuring forces at the string's terminations. Using this method, they gathered valuable empirical evidence of the friction force behavior by studying a violin string excited with a glass rod. In our work, we applied the inverse method to a more complex scenario, including a cello string excited with a real bow. By analyzing the reconstructed friction forces from both measured and simulated data across various bowing conditions, we gathered further insights into the frictional interaction. [This research was funded in whole or in part by the Austrian Science Fund (FWF) [10.55776/P34852]]

10:00

3aSA4. Experimental study of frictional interfaces using solitary waves in 1-D monoatomic granular crystals. Jean Myung Jung (The Grainger College of Eng., Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 409 S 3rd St., Champaign, IL 61820, jmjung3@illinois.edu), Alfredo Fantetti (Dept. of Mech. Eng., Imperial College London, London, United Kingdom), Alexander F. Vakakis, and Kathryn Matlack (The Grainger College of Eng., Dept. of Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

This research addresses critical challenges in understanding the complex mechanics in frictional interfaces. Interfaces in machinery subjected to vibrations are difficult to study due to limited experimental data on contact states like stick, slip, and separation. To address this, we exploit solitary waves in granular crystals as a novel method to monitor friction in vibrating surfaces. Solitary waves exist in nonlinear media and are highly sensitive to contact mechanics due to their localized point contact and short interaction timescales. Their sensitivity to boundary conditions, short interaction timescales, and localized point contact suggest the potential for nondestructive evaluation of friction, offering advantages over conventional ultrasonic techniques. This study focuses on effects of vibrating frictional interface at the boundary of a chain of rigid spheres and its impact on solitary wave speed. The experiment demands precise control and measurements to address the challenges by solitary waves' sensitivity to contact variations. Experimental results show measurable changes in solitary wave speed during stick, slip, and the evolution of contact parameters by wear, above the noise floor, demonstrating the potential for nondestructive evaluation. Anticipated outcomes include deeper understanding of interfacial mechanics, new characterization tools, and innovative acoustic metamaterials, with implications for industrial and naval applications.

10:20–10:40 Break

10:40

3aSA5. Visualizing nonlinearity and non-repeatability in bolted joints through digital image correlation. Nicholas Pomianek (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, pomianek@bu.edu), Trevor Jerome (NSWC, Carderock Div., Bethesda, MD), Enrique Gutierrez-Wing, and James G. McDaniel (Mech. Eng., Boston Univ., Boston, MA)

The dynamics of bolted joints are known to be highly complex in terms of nonlinearity and non-repeatability. This has been established primarily through traditional experimental modal analysis techniques that measure the system response using accelerometers or laser doppler vibrometers. These single-point system identification techniques are powerful tools for detecting nonlinear behavior and non-repeatability but they do not interrogate the underlying physical processes that cause this behavior. High-speed 3D Digital Image Correlation (DIC) is applied to a corner-bolted square plate system under dynamic load to identify the specific region of the frictional interface responsible for nonlinear behavior. Non-repeatability is similarly investigated using DIC by comparing non-repeatability in measured displacement fields to non-repeatability in dynamic response. Multiple plate thicknesses are tested to determine the applicability of the results to generalized systems and whether the mechanical properties of the bolts dominate the repeatability behavior of the joint. DIC and modal testing results are compared to a finite element model to evaluate the performance of current predictive modeling techniques. [Work supported by the Office of Naval Research under Grant No. N00174-22-1-0015.]

11:00

3aSA6. Measurement of local contact stiffness using ultrasound. R. J. Osborne (Mech. Eng., Univ. of South Florida, 4202 E. Fowler Ave., ENG 030, Tampa, FL 33620, rjosborn@usf.edu), Matthew R. Brake (Mech. Eng., Rice Univ., Houston, TX), and Drithi Shetty (Mech. Eng., Univ. of South Florida, Tampa, FL)

Friction in built-up structures has a nonlinear impact on the overall dynamics and is difficult to predict due to the different phenomena occurring at various length scales. A major challenge is observing and quantifying the interfacial behavior *in-situ*. One solution that has long been studied is the use of ultrasound to measure the interfacial contact stiffness. However, existing research requires complete immersion of the setup in water or permanently attaching the transducer to the test article using epoxy to ensure good, consistent coupling between the transducer and metal. Immersion would change the interface conditions whereas epoxy-based coupling would mean that an array of transducers is required to measure larger contact surfaces. This work proposes a cost-effective, roving-transducer method that uses a conformable delay-line to couple the transducer to the metal surface. The accuracy of the proposed method is tested against static measurements of contact stiffness from a universal testing machine. Furthermore, the contact stiffness across the interface of two bolted stainless-steel blocks is measured using the proposed approach.

11:20

3aSA7. A stick slip model of the singing wineglass. Alexander M. Mertz (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, amm9385@psu.edu), Megan F. Orzolek (NSWCCD, University Park, PA), and Michael L. Jonson (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The characterization of friction is a multiscale problem involving a balance between asperity contact and fluid film forces. Periodic stick-slip phenomena are often associated with the friction force that causes mechanical vibration. This type of vibration can cause noise to radiate via structural modes of the system. Many mechanical systems have connections that involve friction, including bolted joints, bearings, and brakes. Avoiding stick slip conditions minimizes wear and vibration, while extending the life of the component. Understanding the exact conditions that stick slip occurs can be difficult, as it requires a characterization of the friction forces and the loss mechanisms in a complex mechanical system. Classic single degree of freedom models predict the stability criterion to be a function of the system damping, normal load, and the derivative of friction coefficient with respect to sliding speed. Many musical instruments utilize the stick slip phenomenon to produce sound. In this work, we study the stick slip phenomenon with a singing wineglass. The dynamic friction force and radiated pressure is measured under a variety of speed and load combinations. Friction models of increasing complexity are used to demonstrate the main features of the dynamic force and radiated pressure spectra.

WEDNESDAY MORNING, 21 MAY 2025 SALON H, 8:55 A.M. TO 11:20 A.M.

Session 3aSC

Speech Communication and Psychological and Physiological Acoustics: Speech Perception Beyond Intelligibility I

Melissa Baese-Berk, Cochair
Linguistics, Univ. of Chicago, 1115 E 58th St., Rosenwald Hall Rm. 203, Chicago, IL 60657

Susannah V. Levi, Cochair
Communicative Sciences and Disorders, New York Univ., 665 Broadway, 9th Fl., New York, NY 10012

Chair's Introduction—8:55

Invited Papers

9:00

3aSC1. Is speech processing faster for familiar talkers? Sandy Abu El Adas (BCBL, San-sebastian, Gipuzkoa, Spain), Melissa Baese-Berk (Linguist, Univ. of Chicago, Chicago, IL), Nadine Lavan (Biological Psych., Queen Mary Univ. of London, London, United Kingdom), Susannah V. Levi (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, svlevi@nyu.edu), and Carolyn McGettigan (Speech, Hearing & Phonetic Sci., Univ. College London, London, United Kingdom)

When listening to speech, intelligibility (measured as percent words or phonemes correct) is increased for familiar talkers, both for talkers who are naturally familiar with the listener and for talkers who become familiar through lab-based training. These findings are robust and have been shown for individual words and sentences and for younger and older listeners. In the current study, we test the extent to which talker familiarity affects speech processing time. Listeners are familiarized with three talkers in a talker identification task. To test differences in processing speech, listeners then complete a lexical decision task with both familiarized and unfamiliar speakers. We predict that listeners will be faster at making lexical decisions for familiar speakers compared to unfamiliar speakers. Data collection is ongoing and results of this study will inform how talker familiarity impacts processing more broadly speaking.

9:20

3aSC2. Processing L2-accented speech: lessons from pupillometry and the dual task paradigm. Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, One Brookings Pl, Campus Box 1125, St. Louis, MO 63130, kvanengen@wustl.edu) and Mel Mallard (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

In most conversations, people correctly recognize the words their interlocutor produces. The work required to do this, however, can vary widely among speakers, listeners, and situations. Factors, such as hearing loss, environmental noise, language proficiency, or familiarity with each other's ways of speaking, can significantly affect how much effort is required to understand speech. This presentation will discuss approaches to studying this work in the context of processing of L2-accented speech. Specifically, it will focus on lessons learned from pupillometry (a physiological measure) and the dual task paradigm (a behavioral measure), which we are using to elucidate relationships among listening effort, speech intelligibility, and individual listener characteristics.

9:40

3aSC3. Exploring dysarthric speech perception through listening effort. Micah E. Hirsch (Speech, Lang., and Hearing Sci., Boston Univ., 10 Orkney Rd., Apt C, Brighton, MA 02135, mehirsch@bu.edu) and Kaitlin L. Lansford (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL)

Listening effort is an emerging construct in dysarthric speech perception research and has the potential to complement intelligibility measures. The current study evaluated listening effort using pupillometry and perceived listening effort (PLE) ratings. Listeners completed a speech perception task in which they heard a speaker with dysarthria and a neurotypical speaker while an eye-tracking system measured their pupil dilation. Listeners also provided PLE ratings. Results showed that listeners had greater pupil dilation and rated PLE higher when listening to the speaker with dysarthria, even when the speech was accurately perceived. Additionally, incorrectly perceived trials were associated with higher pupil dilation and PLE ratings. Although the two measures of listening effort showed similar overall patterns, differences emerged, particularly in the interaction between dysarthria presence and perceptual accuracy. Finally, pupil dilation tended to decrease across trials while PLE ratings remained static. Overall, the findings suggest that listening effort is a valuable perceptual outcome to consider alongside speech intelligibility and may contribute to a more comprehensive understanding of the communication challenges associated with dysarthria. Further research is needed to better understand individual listener differences in listening effort and to establish the reliability and validity of the measures used to capture listening effort.

10:00–10:20 Break

10:20

3aSC4. Talker-based asymmetries in memory for spoken sentences. William S. Clapp (Linguist, Stanford Univ., 450 Jane Stanford Way, Bldg. 460, Stanford, CA 94305, wsclapp@stanford.edu) and Meghan Sumner (Linguist, Stanford Univ., Stanford, CA)

It is well-established that memory plays a central role in the human ability to understand speech, but not all experiences with speech are remembered equally well. One hypothesis about how these asymmetries emerge is that representation strength depends on how listeners allocate cognitive resources to the speech signal, partially based on the social characteristics of the talker. To test this, we conducted three recognition memory experiments with 12 diverse, but roughly equally non-standard talkers (i.e., no speakers of mainstream American English). We manipulated attention at encoding, as well as retrieval modality. Participants heard spoken sentences at study with different test blocks: auditorily presented sentences (Exp. 1), orthographically presented sentences (Exp. 2), and images (Exp. 3). In all three experiments, memory was stronger in Full than Divided Attention. Crucially, we also found that memory performance depended heavily on the talker and that talker interacted with voice of repetition (Exp. 1) and attention (Exps. 1, 2, and 3) in complex ways. These results point to a highly dynamic, context-sensitive network of speech representations where encoding and recognition behaviors are patterned by resource allocation in addition to frequency and typicality. We discuss implications for understanding voice-based biases in everyday situations.

Contributed Paper

10:40

3aSC5. Situation-specific listening effort as a marker of speech perception success. Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall Rm. 115, Minneapolis, MN 55455, mwinn@umn.edu)

Correct repetition of speech does not indicate the various effortful processes involved, such as mentally repairing words that were missed, ignoring words because they are already known, or targeting specific words containing key information. Across a series of studies, these abilities were tested using stimuli that gave listeners the opportunity to treat the same speech content with different strategies based on situational needs and cues. Momentary changes in listening effort and sensory gain were revealed by changes in pupil dilation and suppression of microsaccades linked to key stimulus landmarks. Typical-hearing listeners consistently exerted effort at specific times, such as the moment after missing a target word or when hearing repetition of a stimulus previously missed, while also reducing effort during speech that was already heard or irrelevant to the task. However, listeners with cochlear implants instead showed signatures of sustained effort that persisted after stimulus presentation, was not specific to key moments of information, and which did not decrease for irrelevant or redundant speech. These results suggest that listener-driven effort can be situationally dependent, and the ease and quickness of regulating effort is a dimension of success that would not be revealed by measures of word repetition accuracy.

3a WED. AM

Invited Paper

11:00

3aSC6. Predictors of individual differences in accented speech perception. Portia Washington (Psychol. Sci., Univ. of Connecticut, 2 Alethia Dr., Unit 1271, Storrs, CT 06269, portia.n.washington@uconn.edu) and Emily B. Myers (Psychol. Sci. and Speech, Lang. and Hearing Sci., Univ. of Connecticut, Storrs, CT)

Speech perception relies on quick mapping of spectral-temporal cues onto linguistically meaningful units. This becomes difficult when there is a mismatch between the listener's phonological representations and phonetic cues produced by a speaker with an unfamiliar accent. This difficulty is similar to other adverse listening conditions and the ability to resolve phonetic ambiguity has been linked to individual differences in phonetic category representations. In this study we measure phonetic category structure and accent perception accuracy for 85 adults (55 younger and 30 older) to explore the impact of bottom-up processing on accent perception. We use a discrete Visual Analog Scale (VAS) phonetic decision task requiring listeners to rate tokens on a continuum spanning canonical endpoints and ambiguous tokens. Phonetic category structure is modeled by using a four-parameter logistic regression (slope, boundary, minimum, and maximum) to estimate each individual's unique behavioral function. Interestingly, slope (or gradience interpreted as increased phonetic sensitivity) did not predict accent perception accuracy, but the error of responses to that function (consistency) did. Age was not a significant predictor of accent performance in this sample. Implications of response consistency, slope, and the stability of speech perception processes in healthy aging will be discussed.

WEDNESDAY MORNING, 21 MAY 2025 GALERIE 1, 8:55 A.M. TO 12:00 NOON

Session 3aSP

Signal Processing in Acoustics, Acoustical Oceanography and Computational Acoustics: Physics-Inspired Neural Networks (PINNs) in Underwater Acoustics

Ananya Sen Gupta, Cochair
Dept. of Electr. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242

Monika Aggarwal, Cochair
IIT Delhi, IIT Delhi Main Rd, IIT Campus, Hauz Khas, New Delhi 110016, India

Chair's Introduction—8:55

Invited Papers

9:00

3aSP1. Leveraging overcomplete autoencoder for improved direction-of-arrival estimation using sensor array. RASHIDA K (Electronics, Cochin Univ. of Sci. and Technol., Thrikkakkara, Errnakulam, Kerala 682021, India, rashidasuhail@gmail.com), Minu A Pillai (Electronics and Commun. Eng., Indian Inst. of Information Technol., Kottayam, Ernakulam, Kerala, India), and Supriya M H (Electronics, Cochin Univ. of Sci. and Technol., Ernakulam, Kerala, India)

The Direction of Arrival (DoA) estimation using linear sensor arrays is a critical task in array signal processing, with applications spanning SONAR, RADAR, and more. Traditional methods often struggle with the non-linear and high-dimensional nature of the data involved. This paper proposes a novel approach that leverages the strengths of both deep learning and machine learning classifiers. By combining an overcomplete Auto Encoder (AE) with a Support Vector Machine (SVM), the proposed method effectively extracts intricate features from the high-dimensional input signals, enabling accurate DoA estimation. The model was trained and evaluated using field data collected from the Indian Ocean. Sensor delays were introduced experimentally to the collected single-channel data by positioning acoustic sources at various angles within 0 to 180-degree relative to a linear microphone array with 16 channels. In DOA estimation, where experimental data are often noisy or missing features, the AE learns a detailed latent space representation that helps to smooth the data before it is classified. While AE + SVM is not new, its application to DoA estimation with noisy, incomplete, or sparse experimental data is novel. The overcomplete AE addresses the unique challenges of DoA estimation by improving feature quality and

enabling more accurate, robust predictions. Experimental results demonstrate the efficacy of this unified model, achieving a remarkable 94.7% accuracy in DoA estimation tasks.

9:20

3aSP2. In pursuit of learning ocean dynamics from basin scale acoustics: Insights from the Kauai Beacon transmissions. Nicholas C. Durofchalk (Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA, nicholas.durofchalk@nps.edu), Andrew J. Christensen, Timothy Linhardt, Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Kay L. Gemba (Phys., Naval Postgrad. School, Monterey, CA)

The low frequency Kauai Beacon (KB) source, situated at 1000 m depth on the north shore of Kauai, provides a unique opportunity to investigate basin-scale ocean dynamics and underwater acoustic propagation through the lens of controlled, long range transmissions (3,500 + km) of broadband m-sequence signals to fixed receivers, such as those of the Comprehensive Nuclear Test Ban Organization (CTBTO) hydroacoustic monitoring station at Wake Island. This presentation focuses on an extensive dataset of received and simulated signals, spanning more than a year, and explores how travel time fluctuations on the order of a second could be related to significant oceanographic events or changes to the environment. Simulated data are synthesized using physics-based acoustic propagation software and ocean state estimates from reanalysis data, providing machine learning frameworks an opportunity to directly train on the mapping between the ocean state and the received signal. We then leverage these simulated datasets to train neural networks, enabling comparisons with models trained on the measured receptions. This approach elucidates the ability of machine learning to uncover unmodeled physical effects while maintaining coherence with the underlying physics.

9:40

3aSP3. Machine learning-based forecasting of ocean channel impulse responses over basin-scale distances motivated by prior knowledge of the waveguide propagation physics. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, NJ), Nicholas C. Durofchalk (Naval Postgrad. School, Monterey, CA), Andrew J. Christensen, Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Kay L. Gemba (Naval Postgrad. School, Monterey, CA)

The underwater acoustic channel is challenging to predict given the uncertainty of the oceanic environment. This is particularly difficult across basic-scale distances due to the intricacies and interactions between the time-varying oceanic processes that influence the ocean state and, therefore, the channel impulse response. From an application perspective, such channel forecasting knowledge can vastly improve the quality of interpretation of sensor measurements as the impact of the channel delay and delay-Doppler spread can be accurately compensated for in sensor network data. Similar gains may be expected for target-specific interpretation of sonar pings by harnessing the accurate forecasting of channel impulse response from the sonar transceiver to the target and back. In this work, we will focus on ideas to improve machine learning-based forecasting of ocean channel impulse responses over basin-scale distances. Specifically, we will investigate the use of constraints during training on the ML predictions motivated by prior knowledge of the waveguide propagation physics. These include penalty functions that force causality, constraining ray path structures, expected channel spreading etc. The approaches are demonstrated on simulated and real measurements of channel impulse responses using M-sequence transmissions from the 75 Hz Kauai beacon received at Wake Island.

Contributed Paper

10:00

3aSP4. Automatic sonar target recognition using regularized wavelet neural networks. Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, 3100 Seamans Ctr., Iowa City, IA 52242, andrew-christensen@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Convolutional Neural Networks (CNNs) have gained prominence in the sonar signal processing community due to their ability to capture complex patterns in acoustic data. However, in automatic sonar target recognition, the limited data availability often leads to overfitting, making network training challenging. Recent studies have shown that CNNs tend to learn filters that resemble wavelets, suggesting that incorporating wavelet-based priors into the design of convolutional neural networks could reduce the learning burden. Building on this insight, we propose a wavelet neural network that replaces learned filters in a CNN with a tight wavelet frame, learning only how to combine wavelet scales. Combining a tight wavelet frame with regularization constraints provides effective control over overfitting of the proposed wavelet neural network. The proposed approach is evaluated on real-world sonar target echo return datasets. [This talk will present research funded by DoD Navy (NEEC) (Grant No. N00174201001) and the ONR (Grant numbers N000142112420 and N000142312503)]

10:20–10:40 Break

Invited Papers

10:40

3aSP5. Quantum underwater communication: Bridging the gap between quantum technologies and marine applications. Monika Aggarwal (IIT Delhi, IIT Delhi Main Rd., IIT Campus, Hauz Khas, New Delhi 110016, India, maggarwal@care.iitd.ac.in)

Underwater communication has traditionally depended on acoustic methods, which are limited in terms of bandwidth, data rates, and security. To address the growing demand for faster, more secure, and reliable communication in underwater environments, our work explores quantum communication as a transformative solution. Quantum communication offers significant advantages, such as highly secure data transmission, resistance to eavesdropping, and techniques like quantum key distribution (QKD). Our research focuses on applying quantum principles to underwater scenarios, integrating quantum systems with traditional acoustic communication to create hybrid frameworks that overcome existing limitations. Key challenges include photon propagation constraints, environmental noise, and quantum decoherence in underwater channels. Through this work, we aim to develop technologies that enable secure data exchange, support underwater IoT networks, and enhance marine exploration. By addressing the challenges and harnessing the potential of quantum communication, this research seeks to redefine underwater communication and open new avenues in marine connectivity.

11:00

3aSP6. GAN-based synthetic data generation for underwater acoustic channel modeling. Farheen Fauziya (Elec. and Comput. Eng., Univ. of Iowa, 1130 Seamans Ctr., 103 South Capitol St., Iowa, IA 52242, farheen-fauziya@uiowa.edu)

Underwater acoustic communications is the most effective technique for communicating underwater. However, the environment is very challenging severely limiting the capabilities of communications system. Availability of data can mitigate this problem, especially with the maturing for ML techniques for channel estimation and equalization. Such works are hindered by extreme difficulty in collecting enough labeled training data. While data availability has always been an issue, traditional receiver design was greatly supported by some usable synthesis models such as Bellhop. In recent times, such models having given way to ML based synthesis techniques such a Variational Autoencoders (VAEs), Generative Adversarial Networks (GANs), and more recently the diffusion model especially for images. GANs are a cutting-edge class of deep learning models where two networks, a generator and a discriminator, are trained in opposition to produce highly realistic data. These models provide greater variability in the sample set, thus resulting in more comprehensive dataset for representing the challenging and divers underwater acoustic channel. In this work, we explore the feasibility of GAN to accomplish the task of generating diverse and representative data samples for underwater acoustic communication. The synthesized data can be used for training ML models, e.g., transformer-based equalizers, with the data generated by the GAN's generator serving as the training input. [Work supported by Office of Naval Research Grant No. N000142312503.]

Contributed Paper

11:20

3aSP7. Convex optimization of shallow neural networks with applications to automatic sonar target recognition. Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, 3100 Seamans Ctr., Iowa City, IA 52242, andrew-christensen@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Neural networks are powerful tools for automatic sonar target recognition, capable of modeling the complex dynamics of the ocean from large amounts of data. However, training these neural networks often poses challenges due to their inherent non-convexity. As a result, they are highly sensitive to parameter initialization and often converge to suboptimal local minima. To address these issues, we propose a framework that reformulates two-layer multi-class ReLU neural networks as convex optimization problems solvable in polynomial time. Our approach can incorporate regularization constraints and domain-specific priors, enhancing both interpretability and robustness to overfitting. The effectiveness of this convex neural network framework is demonstrated on real-world sonar target echo return datasets. [This talk will present research funded by DoD Navy (NEEC) Grant No. N00174201001 and the ONR grant numbers N000142112420 and N000142312503.]

11:40–12:00 Panel Discussion

Session 3aUW

Underwater Acoustics and Acoustical Oceanography: Boundary Interactions Including Shear Wave Effects in Underwater Acoustics

Anatoliy N. Ivakin, Cochair
Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Peter H. Dahl, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98115-7834

Chair's Introduction—8:55

Invited Papers

9:00

3aUW1. Estimation of sediment shear speed using interface waves in the New England Mudpatch. James H. Miller (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., URI Ocean Eng., Narragansett, RI 02882, miller@uri.edu), Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD San Diego, La Jolla, CA), and Cecelia Schneider (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Stoneley and Scholte interface waves were generated in the New England Mud Patch in May 2022 by the Interface Wave Sediment Profiler (iWaSP). The iWaSP is a piezoelectric bender beam transducer which vibrates the seabed about 70 m below the sea surface. The range between the iWaSP and the receiving Ocean Bottom Recorders (OBX) with 3-axis geophones and hydrophone was about 40 m. The speed of these interface waves is approximately 90% of the shear wave speed in the higher-speed elastic medium. Arrivals were detected on the geophones with speeds ranging from 39 to 133 m/s for a mud thickness of about 6 m. However, the top 50 to 60 cm is much softer than the underlying mud and we term that the "surface layer." We hypothesize that these interface waves propagated on the surface-layer-mud (Scholte) and the mud-sand interfaces (Stoneley). This work shows detections of Scholte and Stoneley waves and their arrival times provide estimates of the shear speeds of the layers. Modeling of these waves was performed using finite element modeling in COMSOL. Finally, an updated shear speed model for New England Mudpatch sediments is presented. [Work supported by ONR.]

9:20

3aUW2. Estimation of shear wave speed at the New England Mud Patch using vector sensor measurements. Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, gpotty@uri.edu), James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Ying-Tsong Lin (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

Horizontal to vertical spectral ratio (HVSR) of the Scholte wave particle motion can be used to estimate ocean bottom shear properties. This study uses data from two ocean bottom recorders (OBX) deployed during the Seabed Characterization Experiment (SBCEX 2022) in the New England Mud Patch (NEMP). Sediment thickness data from seismic surveys and core data were used to constrain the shear speed estimation approach. The bottom model estimated by matching the HVSR and ellipticity angle from a model with the data compares well with two previous inversions from the same location. The current inversion estimates shear speed in the sublayers within the mud layer which were identified by the seismic surveys and core data. The HVSR based method is a fast and robust inversion approach for estimating the thickness of the sediment layers and the shear wave speeds. The data used for this study consist primarily of ambient seismic field mixed with potential far field anthropogenic noise sources (e.g., ships, sources deployed as part of experiment). [Work supported by the Office of Naval Research, code 322 OA.]

9:40

3aUW3. Estimates of compressional and shear speed attenuation in the 10–50 Hz frequency range based on underwater noise from a merchant ship. Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98115-7834, dahl@apl.washington.edu), David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA), and Robert W. Drinnan (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Observations of underwater noise emission from a merchant ship are made as a function range in waters of depth 77 m on the New England Mud Patch (NEMP), as part of the 2022 Seabed Characterization Experiment with primary objective being study of sound propagation involving seafloors consisting of fine-grained sediments. In this region the seafloor is characterized by a mud-like layer of order 10 m thickness, commencing at the water–sediment interface. The ship traversed the NEMP producing a 30 km observation transect for which the water and mud-layer depth slowly varied with range. Data are forward-modeled using adiabatic normal modes based on a range-varying geoacoustic model for the upper sediments that incorporates stratigraphic information from the experimental area. For the

deep sediment structure, the compressional speed is a constant, sand-like layer of speed 1810 m/s for O(100) m, after which speed increases before ultimately terminating at a basalt basement of speed 5000 m/s. For simplicity, compressional attenuation is estimated upon assumption of realistic values for the Poisson ratio and shear speed quality factor Q. For frequencies near 10 Hz, estimated compressional wave attenuation is reduced by approximately 30% relative to corresponding estimates that exclude elastic effects.

10:00

3aUW4. Bayesian matched-field inversion for shear and compressional geoacoustic profiles at the New England Mud Patch. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, School of Earth and Ocean Sci., University of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), David P. Knobles (The Platt Inst., Austin, TX), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

This paper estimates depth-dependent profiles of shear- and compressional-wave geoacoustic properties for seabed sediments at the New England Mud Patch through matched-field inversion of broadband (20-2000 Hz) acoustic data recorded at a 14-element vertical line array due to a combustive sound source. Trans-dimensional Bayesian inversion is applied to sample probabilistically over the number of seabed layers and corresponding layer depths and geoacoustic properties, as well as over the order and parameters of an autoregressive error model. This approach, based on a parallel-tempering implementation of birth/death reversible-jump Markov-chain Monte Carlo sampling, combines objective, data-driven model selection and quantitative parameter/uncertainty estimation. Results indicate low shear-wave speeds (~30 m/s) with small uncertainties over most of the upper mud layer, increasing in underlying transition and sand layers, with values in good agreement with *in situ* probe measurements (for the mud) and nominal values (for sand). The compressional-wave attenuation profile is well estimated but shear-wave attenuation is poorly constrained. Comparison of results for inversions both with and without shear-wave parameters and consideration of inter-parameter correlations indicate that estimates of compressional-wave parameters, including attenuation, are not substantially influenced by shear-wave effects, with the possible exception of the sand layer.

Contributed Papers

10:20

3aUW5. Interface scattering from broadband sources in the New England Mud Patch. Jade F. Lopez Case (Ocean Eng., Univ. of Rhode Island, 215 S Ferry Rd., Narragansett, RI 32837, jadelcase@uri.edu), James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

The New England Mud Patch (NEMP) has been a region of interest in underwater acoustics for the past decade. Numerous geo-acoustic inversion methods have been used to estimate the compressional and shear wave speeds, and densities of four distinguishable sediment layers. While exact values vary, it is known that the upper-most layer is a relatively thin, fluid-like layer (layer 1) of mud. Below that lies more rigid mud (layer 2) that has varying physical properties with depth, followed by a sand layer (layer 3) of approximately 10 m. Below these sediment layers, an elastic half-space layer is assumed for modeling purposes. The Seabed Characterization Experiment (SBCEX22) utilized Ocean Bottom Recorders (OBX's) to provide acoustic and seismic measurements taken at the water–sediment interface. The resulting acoustic pressure and particle velocity measurements indicate the presence of frequency dependent reverberation, likely from roughness at the mud-sand interface between layer 2 and layer 3. Analysis of the shear potential in the elastic layers of the seabed suggests that the frequency dependency seen in these data can be attributed to bottom loss from shear effects. Data from the NEMP and modeling results from a wavenumber integration model will be shown.

10:40–11:00 Break

11:00

3aUW6. Modeling substrate-borne vibrations generated by offshore monopile installation. Cristian Graupe (INSPIRE Environ., Jamestown, RI, cristian@inspireenvironmental.com), Matthew Milone (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), Ying-Tsong Lin (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Kathy Vigness-Raposa (INSPIRE Environ., Newport, RI), James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, Sterling, VA)

Impact and vibratory pile driving for increasingly large offshore monopiles generate low frequency substrate-borne vibrations that radiate from the pile. These waves contain compression and shear wave components that propagate in sediments with poroelastic properties, potentially impacting benthic fauna around the monopile installation site. Additionally, these waves are coupled to the water column and can propagate along the interface between the water and seafloor, exciting additional acoustic waves and contributing to increased sound levels in the water column. Currently, no methods exist to mitigate the effects of these interface waves on marine ecosystems, so understanding how they are generated and the extent to which they propagate is imperative for assessing noise impacts on all marine species. The generation of these vibroacoustic waves is investigated for both vibratory and impact pile driving in a simulated 3D marine environment using numerical modelling. Near field models developed using the SPECFEM3D-Cartesian package are coupled to a Parabolic Equation model adapted to include poroelastic wave motion for far field propagation. Sediment properties are derived from data collected during recent benthic surveys. Predictions of particle motion for simulated ground waves are compared to measurements of ground vibration taken at recent or ongoing offshore windfarm installation sites.

11:20

3aUW7. Comparison of observed and predicted reflection loss in the Beaufort Sea using a multilayer, rough, acousto-elastic reflection coefficient model. Jonathan Levay (The MIT-WHOI Joint Program in Oceanography/Appl. Ocean Sci. and Eng., 266 Woods Holes Rd., MS #9, Woods Hole, MA 02543, jwlevay@mit.edu), Gil Averbuch (Woods Hole Oceanographic Inst., Woods Hole, MA), John A. Colosi (Woods Hole Oceanographic Inst., Monterey, CA), Matthew Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., San Diego, CA)

The Canada Basin Acoustic Propagation Experiment (CANAPE) conducted during 2016–2017 utilized a 150-km radius, seven-mooring acoustic tomography array to examine acoustic propagation in the new Arctic. Broadband acoustic transmissions with center frequencies of 172.5, 250–255, and 275 Hz revealed identifiable and trackable ray-like arrivals with grazing angles of 10–20 degrees that reflect off the ice 3–10 times. Worcester *et al*. (2024) [J. Acoust. Soc. Am. 156, 4181–4192] showed that the maximum excess transmission loss per surface reflection, defined as the increase in transmission loss relative to open water conditions, varies from 2–6 dB and is strongly frequency and angle dependent. The loss scales roughly with ice thickness. We have developed a four-layer (water, skeletal ice, solid ice, and air), acousto-elastic reflection loss model that incorporates roughness using the Rayleigh formula. When combined with *in-situ* observations of ice thickness and roughness, as well as physical parameters for ice from literature, the model shows good agreement with the observations during the maximum ice thickness period in June. Comparisons will be made for the

ice growth and melting phases, sensitivity to ice parameters will be investigated, and the results will be interpreted in terms of first year ice rheology.

11:40

3aUW8. Computational analysis of under-ice propagation in deep Arctic Ocean environments using elastic parabolic equation solutions. Scott D. Frank (Mathematics, Marist Univ., 3399 North Ave., Marist College Mathematics, Poughkeepsie, NY 12601, scott.frank@marist.edu) and Anatoliy N. Ivakin (Univ. of Washington, Seattle, WA)

Deep Arctic environments have specific conditions that affect acoustic propagation. The presence of an overlying ice layer, as well as single- and double-ducted water sound speed profiles cause variations in travel-time and transmission loss. Compared to other methods, parabolic equation solutions incorporating these environmental challenges provide an effective approach for analysis of their impact on received acoustics signals. Particularly, range dependent variations in the thermohaline structure and ice layer thickness can be difficult to model with normal mode or spectral methods. Practical use of ray-based models for such complicated environments is usually limited by the requirement for acoustic frequencies be sufficiently high. Another significant complication in arctic geo-acoustic modeling is caused by shear elasticity of the ice cover that normally supports both compressional and shear waves. Here, to quantify contributions of shear effects in range-dependent arctic environments, we compare calculations of under-ice transmission loss made using fluid and elastic parabolic equation techniques. The results are presented for various combinations of ice layer and water duct parameters, as well as various source and receiver depths, ranges, and sound frequencies. Specific examples are presented for configurations and environments similar to those found at recent experiments in the Beaufort Sea. [Work supported by ONR.]

WEDNESDAY AFTERNOON, 21 MAY 2025 GALERIE 2, 1:20 P.M. TO 2:40 P.M.

Session 3pAA

Architectural Acoustics, Noise, Psychological and Physiological Acoustics, and Speech Communication: At the Intersection of Speech and Architecture IV

Kenneth Good, Cochair
Architecture Acoustics, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17601

Evelyn Hoglund, Cochair
Speech and Hearing, Ohio State Univ., 104a Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210

Pasquale Bottalico, Cochair
Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820

Contributed Papers

1:20

3pAA1. Acoustic evaluation of penetrations in drywall constructions. Alexander Müller (Architecture and Civil Eng., Tech. Univ. pf Appl. Sci. Würzburg-Schweinfurt, Würzburg, Germany) and Normen Langner (Architecture and Civil Eng., Tech. Univ. pf Appl. Sci. Würzburg-Schweinfurt, Röntgenring 8, Faculty of Architecture and Civil Eng., Würzburg 97070, Germany, normen.langner@thws.de)

In the design of drywall constructions for office buildings with airborne sound insulation requirements, penetrations, such as cables, cable bundles, or ventilation ducts, are typically not included in acoustic calculations. For this reason, a research project was conducted to examine the impact of various types of penetrations on the weighted sound reduction index (R_w) of different drywall partitions. Laboratory measurements were performed on four drywall constructions, testing over 180 modifications. Starting with properly installed, penetration-free walls, different penetrations were introduced, and their effects on R_w were measured. The results demonstrated significant reductions in sound insulation, depending on the penetration type and installation quality. Key factors included the precision of the opening and the size of the unfilled residual cross-sectional area. The ongoing analysis aims to determine whether the initial wall sound insulation correlates with penetration effects, such as greater sound insulation amplifying the impact of penetration on its reduction. Based on these findings, it is recommended to incorporate an additional safety margin in the design phase, tailored to the specific type of penetration, to better address its potential impact on sound insulation.

1:40

3pAA2. A field survey on speech privacy in open-plan offices: Quantitative assessment of speech leakage during working hours. Naoya Maruyama (Kumamoto Univ., 2-39-1 Kurokami Chuo-ku, Kumamoto 860-8555, Japan, maruyama@arch.kumamoto-u.ac.jp), Ainun Nadiroh, Miyu Maruyama, and Keiji Kawai (Kumamoto Univ., Kumamoto-shi, Japan)

Open-plan offices are widely implemented due to their open design and facilitation of communication. However, their acoustic characteristics may pose challenges to speech privacy. This study aims to assess the current state of speech privacy in two offices through field investigations. Although the two offices have comparable room volumes, they differ significantly in occupancy levels, suggesting varying risks of speech leakage due to differences in seating density. The investigations included acoustic measurements in unoccupied rooms, noise level measurement, and sound recordings during

working hours, as well as a questionnaire survey of office workers. Additionally, this study explored a quantitative approach to evaluating speech leakage using audio recordings collected during working hours. The results suggest that speech transmission is affected by seat density. Furthermore, a new method was proposed, employing speech recognition technology to quantitatively evaluate the extent of speech leakage from recorded workplace sounds. The effectiveness of this method was demonstrated, highlighting its potential for assessing speech privacy in real-world office settings. These findings underscore the importance of considering seating density and background noise in the design of open-plan offices to enhance acoustic environments and speech privacy.

Invited Papers

2:00

3pAA3. Free speech, intelligibility, and privacy: A court case study using forensics, standards, and legal approaches. Part 1. David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net), Katie Schwartzmann, and Annie K. Cleveland (School of Law, Tulane Univ., New Orleans, LA)

This court case study combines the use of testing and standards for privacy/intelligibility, sound transmission, and soundscape to develop a forensic re-creation of an incident that involves amplified and unamplified sound sources to verify perception of speech and consider context. The legal approach to applying the science and also legal arguments for what constitutes allowable free speech will also be presented.

2:20

3pAA4. Free speech, intelligibility, and privacy: A court case study using forensics, standards, and legal approaches. Part 2. Katie Schwartzmann (School of Law, Tulane Univ., New Orleans, LA), David s. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net), and Annie K. Cleveland (School of Law, Tulane Univ., New Orleans, LA)

This court case study combines the use of testing and standards for privacy/intelligibility, sound transmission, and soundscape to develop a forensic re-creation of an incident that involves amplified and unamplified sound sources to verify perception of speech and consider context. The legal approach to applying the science and also legal arguments for what constitutes allowable free speech will also be presented.

WEDNESDAY AFTERNOON, 21 MAY 2025 STUDIOS 9/10, 1:00 P.M. TO 2:00 P.M.

Session 3pAOa

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

David R. Barclay, Chair
Dept. of Oceanography, Dalhousie Univ., P.O. Box 15000, Halifax B3H 4R2, Canada

Chair's Introduction—1:00

Invited Paper

1:05

3pAOa1. Development of acoustic remote sensing of seagrass ecosystems and understanding of their climate impacts. Megan Ballard (Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu)

Seagrasses provide a multitude of ecosystem services and act as important carbon sinks. However, seagrass habitats are declining globally, and they are among the most threatened ecosystems on earth. For these reasons, long-term and continuous measurements of seagrass parameters are of primary importance for ecosystem health assessment and sustainable management. This talk will present results from both active and passive acoustical methods for ecosystem monitoring in seagrass meadows. Examples of both techniques will be presented based on data collected as part of a two-year continuous deployment of an acoustical measurement system operating in a seagrass bed dominated

by Thalassia testudinum (turtle grass) in Corpus Christi Bay, Texas. From a propagation perspective using a broadband acoustic source, gas bodies contained within the seagrass tissue as well as photosynthetic-driven bubble production results in attenuation and scattering of sound that produces increased transmission loss. For the passive approach, the detachment of gas bubbles from the plants is an important component of the ambient soundscape. The data show annual trends related to the seasonal growth pattern of Thalassia as well as diurnal trends correlated with photosynthetically active radiation. [Work supported by ONR, NSF, ARPA-E.]

WEDNESDAY AFTERNOON, 21 MAY 2025 STUDIOS 9/10, 2:20 P.M. TO 3:20 P.M.

Session 3pAOb

Acoustical Oceanography: Decadal Survey for Ocean Acoustics

Marcia Isakson, Chair
Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnett Rd., Austin, TX 78758

Invited Paper

2:20

3pAOb1. Decadal Survey for Ocean Acoustics. Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnett Rd., Austin, TX 78758, misakson@arlut.utexas.edu)

Understanding and anticipating change in the ocean, and how it will affect marine ecosystems and humans, has never been more urgent. However, at the start of this new decade (2025-2035), U.S. investment in ocean science, engineering, and technology is not keeping pace with growing societal needs, even as U.S. competitors are increasing investments in ocean science and advancing their capacities. At the request of the National Science Foundation (NSF), this report provides advice on how to focus investments in ocean research, infrastructure, and workforce to meet national and global challenges in the coming decade and beyond, and in doing so, enhance national security, scientific leadership, and economic competitiveness through a thriving blue economy.

Session 3pBA

Biomedical Acoustics: General Topics in Biomedical Acoustics: Cavitation

Jacob C. Elliott, Chair
The Pennsylvania State Univ., 325 ECoRE Bldg., University Park, PA 16802

Contributed Papers

1:00

3pBA1. Machine learning-assisted closed-loop focused ultrasound for targeted nanoparticle delivery and liquid biopsy in the brain. Hohyun Lee (Mech. Eng., Georgia Inst. of Technol., 901 State St NW, Atlanta, GA 30332, hlee649@gatech.edu), Victor Menezes, Chulyong Kim (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Cynthia M. Baseman (College of Computing, Georgia Inst. of Technol., Atlanta, GA), Jae Hyun Kim (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Samhita Padmanabhan (Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA), Pranav Premdas (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), Naima Djeddar, Anton Bryksin (Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA), Nikhil Pandey, Pavlos Anastasiadis, Anthony Kim, Graeme F. Woodworth (Neurosurgery, Univ. of Maryland School of Medicine, Baltimore, MD), and Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA)

Focused ultrasound (FUS) combined with microbubbles (MB) provide a unique physical method to reversibly increase the permeability of the BBB and improve drug delivery. Real-time methods to monitor and control the MB dynamics is increasingly important as safe and effective treatment of larger brain regions over multiple sessions are becoming clinical priority. However, for current control methods, safety remains a concern because it primarily relies on reacting to MB collapse (i.e., strong wideband emissions—WBE). We hypothesize that machine learning (ML) can assist controllers to predict the onset of strong WBE (>6 dB) and adjust the exposure before WBE occurs, thereby attaining safer BBB opening without compromising efficacy. To test our hypothesis, we trained a multi-layer perceptron (MLP) model using past BBB opening acoustic emission dataset in rodents (54 040 total data). Subsequently, we integrated MLP to closed-loop controller and assessed its ability to predict WBE in real-time during BBB opening. In our training/testing dataset, MLP model showed 95% accuracy of predicting WBE; its application to real-time controller showed more than 90% reduction in total WBE events compared to current state-of-art closed-loop controller without compromising safety (assessed using immunohistochemistry) while retaining high BBB permeability (measured using dynamic contrast enhanced MRI)

1:20

3pBA2. Can focused ultrasound sensitize androgen-independent prostate cancer cells to antiandrogen therapy? Jeannette Nyiramana (Biomedical Eng., Tulane Univ., New Orleans, LA), Jeannette B. Nyiramana (School of Medicine, Tulane Univ., 1430 Tulane Ave. New Orleans, LA 70112, amageed@tulane.edu), and Damir B. Khismatullin (Biomedical Eng., Tulane Univ., New Orleans, LA)

Antiandrogen therapy, such as Enzalutamide (Enz), is used for the treatment of prostate cancer. Enz works as antagonist of the androgen receptor (AR) preventing or reducing the binding of the hormones such as testosterone and dihydrotestosterone in the prostate gland and elsewhere in the body. This inhibits cancer progression and induces cancer cell death. However, Enz and other antiandrogens are only effective for a period of time, after which prostate cancer cells develop resistance to this medication. A 22Rv1 cell line is a model for this resistant, androgen-independent prostate cancer, in which native AR is replaced with AR splice variants that do not contain binding sites for hormones or antiandrogens. The AR splice variants cause AR activation without ligation, leading to uncontrollable cell growth even under exposure to antiandrogen medication. In this work, we show that the mechanical disruption of the cytoskeleton and nuclear chromatin by focused ultrasound resets AR in 22Rv1 cells, thus stopping cancer cell growth and breaking their resistance to Enz.

1:40

3pBA3. The history of single bubble sonoluminescence. Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 4662 175th Ave SE, Bellievue, WA 98006, lacuw@uw.edu)

In 1990, a paper by Crum and Gaitan was presented at the 12th International Symposium on Nonlinear Acoustics, held in Austin, Texas. This paper reported on the discovery of light emissions from a single, stable, bubble, acoustically levitated in a glass flask. This phenomenon was later known as Single Bubble Sonoluminescence (SBSL). Subsequently, this discovery became of international interest, with hundreds of papers published attempting to explain this remarkable behavior. Papers suggested that this was a demonstration of the Dynamic Casimir Effect, that it mimicked the collapse of black holes, and one paper even reported evidence for nuclear fusion. Eventually, the behavior was explained by noting that the bubble acted as miniature chemical reactor, converting the air enclosed in the bubble to a soluble nitrogen molecular complex. Recently, the author was presented with the copy of an abstract, suggesting that this phenomenon was observed nearly 30 years earlier by Katsuya Yosioka and Akira Omura at the Institute of Scientific Research, Osaka University, Japan. The author wishes to acknowledge this paper, to discuss it contents, and to suggest that what Yosioka ad Omura observed was not true SBSL.

2:00

3pBA4. Enhancing particle transport in the brain with oscillatory fluid flows: Models and simulations. Raghu Raghavan (Sonovance, Inc., 4203 Somerset Pl., Baltimore, MD 21210, raghu@sonovance.com)

Can larger drug distributions (e.g., gene therapy needed for monogenic correction of rare brain diseases) be achieved by supplementing direct delivery of the drug into the brain with sonication, steered and focused to effectively push the infusion away from the infusion catheter? Both better steering towards desired targets and potentially larger distribution volumes with appropriate protocols are desirable goals. These techniques may also enable effective intrathecal delivery which is less invasive than direct infusion. The desired velocities of advection and dispersivities (beyond the usually negligible molecular diffusivities) of the therapeutic particles necessary for clinical significance are quite small (1–10 microns/second, 1000–2000 microns squared/second) but still difficult to attain. The mechanisms involved include streaming and augmented dispersion, both very sensitive to boundary conditions: Further development is needed to correct literature in this field because it has numerous errors and omissions that may invalidate the conclusions therein. We discuss the theory, its subtleties, and three possible applications to enhance drug distributions: (i) intrathecal and (ii) intraparenchymal delivery to the brain and (iii) other intraparenchymal deliveries, e.g., to liver. The calculations offer room for optimism, but the

developments needed are considerable. We conclude with a "control diagram" for needed developments.

2:20

3pBA5. Effect of elastic modulus and fiber spacing on boiling histotripsy bubble dynamics in collagenous tissues. Jacob C. Elliott (Graduate Program in Acoust., Penn State Univ., Res. West, State College, PA 16801, jce29@psu.edu) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Collagenous tissues, such as tendon, are resistant to fractionation by boiling histotripsy (BH), possibly from their high elasticity and close fiber spacing. To account for parallel collagen fibers, a numerical bubble model was adapted from blood vessel applications (Hay, 2012) to predict bubble expansion/collapse during BH exposure and compared to experimental results. *Ex vivo* cervine Achilles tendons ($n = 12$) were harvested, and assigned to tendinopathic (soaked in 1% w/v collagenase for 7 days) or healthy (soaked in PBS for 7 days) groups. Tendons were exposed to either load-to-failure mechanical testing or 1.5-MHz BH ($p+ = 127$ MPa/$p- = 35$ MPa; 10-ms pulses at 1-Hz). During BH, bubble activity was monitored using passive cavitation imaging (Philips/ATL L7-4; Vantage® ultrasound system). Model inputs were determined from load-to-failure tests and fiber spacings from microscopy of fibrin gels. Results showed tendinopathic tendons had significantly higher cavitation amplitudes than healthy tendons (0.22 ± 0.04 and 0.08 ± 0.03 V^2, respectively; $p = 0.009$). Additionally, the numerical model indicated that an order-of-magnitude reduction in elastic modulus increased bubble expansion by only 0.1%. However, an order-of-magnitude increase in fiber spacing resulted in a 57% increase in bubble expansion. These results suggest that fiber spacing may influence BH bubble dynamics more than elastic modulus. [Work supported by NIHR01EB032860.]

2:40

3pBA6. Effect of focused ultrasound-induced heat and cavitation on methicillin-resistant Staphycoccus aureus viability and the role of bacterial structural features in susceptibility to cavitation. Pratik A. Ambekar (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, pambek@uw.edu), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Vera A. Khokhlova (Dept. of Medicine, Univ. of Washington, Moscow), Gilles P. Thomas, David Giraud, kaizer Contreras, Daniel Leotta, Matthew Bruce, Stephanie Totten, Jeff Thiel (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Keith Chan (Vantage Radiology and Diagnostic Services, Seattle, WA), W. Conrad Liles (Dept. of Medicine, Univ. of Washington, Seattle, WA), Evan P. Dellinger (Dept. of Surgery, Univ. of Washington, Seattle, WA), Adeyinka Adedipe (Dept. of Emergency Medicine, Univ. of Washington, Seattle, WA), Wayne Monsky (Dept. of Radiology, Univ. of Washington, Seattle, WA), and Thomas Matula (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Methicillin-resistant Staphylococcus aureus (MRSA) is a gram-positive sphere-shaped bacterium and a common cause of abscesses. Focused ultrasound-induced cavitation can kill bacteria and potentially treat abscesses non-invasively by reducing bacterial populations. As an initial study to evaluate this approach, 10 mL MRSA suspensions were used as a simplified abscess model and treated in polycarbonate vials coupled to a focused ultrasound transducer (H-161, Sonic Concepts, Bothell, WA, F# 0.9). The pulse parameters were 1.2 MHz frequency with 20 μs pulse duration at 2% duty cycle, and 40-min treatments were conducted. The coupling water was heated to 37 °C, with input power adjusted to raise and maintain the sample temperature to 50 °C throughout the treatment. The cavitation dose was determined from PCD broadband noise using a co-axially aligned imaging probe (P4-2), while thermal dose (CEM50) was determined from thermocouple temperature measurements. Control experiments involving heating to 50 °C without ultrasound were also conducted. Results showed logarithmic relationship between MRSA inactivation and thermal dose, with cavitation contributing insignificantly to the effect. To understand this result in contrast to statistically significant inactivation of E. coli (gram-negative rod-shaped) by cavitation, we are investigating the effect of bacterial structural features on susceptibility to focused ultrasound-induced cavitation.

Session 3pNS

Noise: General Topics in Noise: Community Noise

Andrew Christian, Chair
Structural Acoustics Branch, NASA Langley Research Center, 2 N. Dryden St., M/S 463, Hampton, VA 23681

Contributed Papers

1:00

3pNS1. Strategy in tackling road traffic noise in Hong Kong: Past, present, and future. Ka-wai CHAN (Environ. Protection Dept., The Government of the Hong Kong Special Administrative Region of the People's Republic of China, Hong Kong, Hong Kong), Huimin YANG (Environ. Protection Dept., The Government of the Hong Kong Special Administrative Region of the People's Republic of China, 26/F, Southorn Ctr., 130 Hennessy Rd., Wan Chai, Hong Kong 999077, Hong Kong, selenahmyang@epd.gov.hk), Yau-hang LEE, Chi-wing LAW, and Chee Kwan Lee (Environ. Protection Dept., The Government of the Hong Kong Special Administrative Region of the People's Republic of China, Hong Kong, Hong Kong)

Hong Kong is one of the cities that have the highest population density in the world with more than 2200 km of road within a built-up area of approximately 287 sq. km. Road traffic noise is inevitably a major environmental concern in such a densely populated city with complex road network. The Government of Hong Kong Special Administrative Region has all along committed to adopting a multi-pronged approach in mitigating road traffic noise impact in Hong Kong to build a livable city. In parallel, the Environmental Protection Department (EPD) has conducted strategic noise mapping regularly to reveal the population exposed to excessive road traffic noise since Year 2000. Despite the continuous increase in population, road network and numbers of registered vehicles from Year 2000 to 2020, the local population exposed to excessive traffic noise has contrarily dropped by about 40%. This paper will present the overall strategy in tackling road traffic noise problem in Hong Kong and its effectiveness over the past 20 years. The plan to further ameliorate road traffic noise will also be discussed.

1:20

3pNS2. Comparative study of the acoustic landscape of selected sites on Spitsbergen in Longyearbyen area during the "Bright" winter and summer seasons. Jerzy Wiciak (Dept. of Mech. and Vibroacoustics, AGH Univ. of Krakow, al. Mickiewicza 30, Krakow 30-059, Poland, jerzy.wiciak@agh.edu.pl), Dorota Mlynarczyk, Pawel Malecki, and Janusz Piechowicz (Dept. of Mech. and Vibroacoustics, AGH Univ. of Krakow, Krakow, Poland)

The Arctic is a unique place in terms of acoustics and nature. It is currently experiencing an increase in average annual temperature of about 1.7 °C, This is twice as much as in the rest of the globe. On July 5, 2020, the Svalbard archipelago recorded a record temperature of 21.7 °C, which is also the highest temperature ever recorded in the European part of the Arctic. Rapid climate change is an additional factor making it a worthwhile place to record the sounds that occur. The article discusses the results of a survey and acoustic measurements carried out in Spitsbergen near Longyearbyen, the largest town in the area. The results obtained made it possible to determine preferences for tourist activities in the Longyearbyen area during the polar pre-winter and summer. The sites of acoustic measurements were selected on the basis of the analysis of the preferences of the people surveyed. Acoustic analyses included: analysis of time courses of sound pressure level A, equivalent sound level A, spectra, and spectrograms.

1:40

3pNS3. Metamaterials application on low-height noise barrier for railways: Challenges of real-world scenarios. Domenico De Salvio (Dept. of Industrial Eng., Univ. of Bologna, Viale Del Risorgimento, 2, Bologna 40136, Italy, domenico.desalvio2@unibo.it) and Massimo Garai (Dept. of Indus. Eng., Univ. of Bologna, Bologna, Italy)

Addressing noise in the living environment is essential to improving public health and urban quality of life. Railway noise poses significant challenges in densely populated areas, necessitating effective mitigation measures. This study explores using metamaterials to enhance the acoustic performance of low-height noise barriers (LHNB) for railways, focusing on geometrical constraints and environmental durability. Metamaterials offer tailored acoustic properties without increasing barrier size. However, their design is challenging, especially for low-frequency absorption, and outdoor conditions demand materials resistant to weathering and long-term degradation. Two acoustic metamaterials—Neck-embedded Helmholtz Resonators (NEHR) and channel resonators—are integrated into an LHNB to target noise attenuation across 100 to 5000 Hz. NEHRs are optimized for low-frequency performance, while channel resonators provide broader frequency absorption with tailored geometrical designs. These elements ensure a lightweight, compact solution without compromising acoustic efficiency. Analytical models and numerical simulations are validated experimentally to demonstrate the efficacy of this hybrid approach. Results highlight significant noise reduction while maintaining structural and aesthetic feasibility. This work advances the design and optimization of acoustic metamaterials, offering practical, scalable solutions for sustainable noise control in urban and transport environments.

2:00

3pNS4. Battery energy storage systems noise modeling and mitigating techniques. Tricia Pellerin (Tetra Tech, 10 Post Office Square, Ste. 1100, Boston, MA 02109, tricia.pellerin@tetratech.com), Kevin Fowler (Tetra Tech, Chicago, IL), Chris Hulik (Tetra Tech, Rochester, NY), and Yona Simonson (Tetra Tech, Boston, MA)

Battery energy storage systems (BESS) facilities are designed to store and manage electricity for later use. Whether permitted in conjunction with a solar and/or wind energy development, or as a standalone project, installation of BESS facilities is increasing all over the country. BESS may appear relatively innocuous, but they can generate elevated sound levels and are often proposed in confined locations close to communities. The primary noise sources associated with BESS include the battery and inverter cooling systems as well as electrical noise produced by the inverters and transformers. Manufacturer specifications provide maximum sound emission levels for the BESS, which typically need to be further refined by considering load rating or directionality. Even in the absence of applicable regulations, it would be advisable to model offsite noise impacts produced by BESS facilities to minimize the potential for future noise complaints. If properly assessed during the permitting process, modeling will provide an indication of possible noise-related issues as well as allow for design considerations and/or mitigation measures that can be implemented to reduce the overall noise impacts to the surrounding communities. This paper will provide details about how to model BESS facilities, noise issues and siting considerations, and noise mitigation options.

Session 3pPAa

Physical Acoustics, Education in Acoustics and Structural Acoustics and Vibration: Nonlinear Waves in Architected Solids

Michael R. Haberman, Cochair
Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Vincent Tournat, Cochair
Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans 72000, France

Chair's Introduction—12:55

Invited Papers

1:00

3pPAa1. Transition waves in a beam coupled to a bistable foundation with a symmetric energy landscape. Dengge Jin, Samuele Ferracin (Univ. of Pennsylvania, Philadelphia, PA), Vincent Tournat (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), and Jordan Raney (Univ. of Pennsylvania, Towne Bldg., Rm. 274, 220 S. 33rd St., Philadelphia, PA 19104-6272, raney@seas.upenn.edu)

Multistable metamaterials, capable of adopting multiple stable configurations, offer versatile control over the shape and mechanical properties of systems. Transition waves (TWs), which propagate spatially through state transitions, provide a promising mechanism for reconfiguring such materials. While TWs in systems with asymmetric energy landscapes propagate stably by transitioning from higher to lower energy states, they require additional energy input to reset the structure. This work explores a multistable system with a reconfigurable surface shape, comprising an elastic slender beam coupled to a foundation of bistable elements, each with a symmetric energy landscape. The symmetric landscape enables TWs with tunable speeds, energy, and propagation distances, which can be controlled by boundary impact speed. The stop position of the TW forms a stable domain wall, separating regions of the beam in distinct stable states. Leveraging the relationship between impact speed and propagation distance, we propose a dynamic strategy for reversible surface shape reconfiguration using sequential impulses to generate targeted configurations. The feasibility of this strategy for shape control of multistable systems has been confirmed experimentally, using buckled double-beams as bistable elements.

1:20

3pPAa2. Nonlinear elastodynamics of dilational metamaterials. Zeb Rocklin (Phys., Georgia Inst. of Technol., School of Physics/ Georgia Inst. of Technol., North Ave., Atlanta, GA 30332, zeb.rocklin@physics.gatech.edu), Neel Singh (Phys., Georgia Inst. of Technol., Atlanta, GA), Audrey A. Watkins, Giovanni Bordiga (Harvard, Cambridge, MA), Vincent Tournat (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), and Katia Bertoldi (Harvard, Cambridge, MA)

Architected structures such as checkerboard patterns of squares joined by thin ligaments at their corners are capable of nonlinear deformations in which the structure contracts globally in the plane while individual elements counter-rotate. However, recent work shows that under generic driving such structures display a wide variety of nonlinear waves rather than this single global mode. We show that such excitations can be understood in terms of conformal symmetry, in which a wide variety of nonlinear (conformal) deformations are nearly degenerate in energy. This symmetry leads to low-energy surface waves whose dispersion is controlled by the structure's bulk modulus and by the size of the architected unit cell. Simulations and experiments also provide evidence of new conserved quantities, the spatially complex generalizations of linear and angular momentum. This approach provides a new platform for creating and controlling novel nonlinear waves for energy harvesting, sensing, and mode conversion. Conformal symmetry grants a degree of analytic control and predictability for characterizing novel waves in complex structures. [Work supported by Army Research Office (MURI # W911NF2210219)]

1:40

3pPAa3. Design and realization of nonlinear mechanical metamaterial cloaks. Giovanni Bordiga, Jean-Gabriel Argaud, Audrey A. Watkins (John A. Paulson School of Eng. and Appl. Sci., Harvard Univ., Cambridge, MA), Vincent Tournat (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Laboratoire d'Acoustique de l'Université du Mans (LAUM), Le Mans 72000, France, vincent.tournat@univ-lemans.fr), and Katia Bertoldi (John A. Paulson School of Eng. and Appl. Sci., Harvard Univ., Cambridge, MA)

Based on flexible mechanical metamaterial platforms that have been proven to be experimentally realizable, that support nonlinear deformations and waves, and that possess a large number of adjustable geometrical design parameters, we propose here to realize mechanical metamaterials for cloaking in the nonlinear regime. Mechanical cloaking in the linear regime has been previously achieved using various strategies, from transformation elasticity to data-driven optimization approaches. However, cloaking in structures that support large-amplitude nonlinear elastic responses remains an open challenge due to the difficulty to design the nonlinear elastic properties of metamaterials. The key point is to implement differentiable simulations for our metamaterial dynamical response and solve an optimization problem to systematically find optimal cloaking structures. The latter is then fabricated and tested under both static and dynamic excitations. Specifically, we realize highly deformable structures capable of hosting "undetectable" inclusions, shielding point excitations, and creating stress-free regions within metamaterial domains. Potential applications include shielding against unwanted vibrations, protecting sensitive sensors, and generating desired haptic feedback in highly deformable robotic systems.

2:00

3pPAa4. Kinks and breathers in mechanical metamaterials. Georgios Theocharis (CNRS-LAUM, Le Mans Univ., Le Mans, France, georgiostheocharis@gmail.com)

Flexible elastic metamaterials (flexEM) are architected structures known for their ability to support large elastic deformations, with key features, such as buckling under critical loads, bistability, and zero modes. By leveraging concepts from condensed matter physics—such as symmetry breaking, topological band theory, and frustration—flexEMs enable the observation of rich nonlinear phenomena previously studied in other settings, including conducting polymers, atomic wires, and ferroelectric phase transitions. This talk, building on recent progress in understanding the static behavior of bistable flexEMs, will explore their dynamic response. Using simplified mass-spring lattice models, which effectively capture their elastodynamic behavior, I will focus on nonlinear waves, such as kink solutions and breathers, their collisions, and interactions with linear phonon wavepackets. By investigating these nonlinear dynamics, we aim to propose strategies for dynamically controlling the structure of flexEMs—such as manipulating kink motion to overcome the Peierls barrier or triggering transition waves through breather collisions. The ultimate goal is to deepen our understanding of flexEM dynamics and provide insight into the design of reconfigurable metamaterials for applications in advanced mechanical devices, soft robotics.

2:20–2:40 Break

Contributed Papers

2:40

3pPAa5. Machine learning-based generation of discrete-element models for nonlinear wave propagation in elasto-plastic metamaterials. William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, PO Box 9767, Austin, TX 78766-9767, william.willis@utexas.edu), Michael R. Haberman, and Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Nonlinear elastic metamaterials (NLEMs) support a variety of dynamic phenomena that enable the manipulation of large-deformation elastic waves. Full-scale dynamic simulation of NLEMs is often prohibitively expensive due to the importance of complex, sub-wavelength geometry. Low-order effective medium models based on mass-spring lattices [Wallen *et al.*, arXiv:2407.20434 (2024)] have recently been developed to capture history-dependent effects of plasticity for 1-D simulation of nonlinear wave propagation in NLEMs. However, the model developed therein requires significant preparatory effort to obtain empirical constitutive relations and their derivatives via a complex, ad-hoc curve-fitting procedure. Here, an alternative method is proposed whereby trained deep neural networks provide the constitutive relations, allowing for application of automatic differentiation methods to obtain derivatives for implicit solution of the differential-algebraic equations of motion. The networks are trained using cyclical force–displacement data from a finite-element model of a unit cell of interest, which exhibits buckling under elasto-plastic deformation. The trained neural networks are then incorporated into the discrete-element framework of Wallen *et al*. to simulate wave propagation in a chain of unit cells.

3:00

3pPAa6. Leveraging nonlinear hysteretic response for structural characterization. Paul Geimer (Los Alamos National Lab., P.O. Box 1663, M.S. G755, Los Alamos, NM 87545, pgeimer@lanl.gov), Andrew Delorey, Rajarshi Bose, and Timothy J. Ulrich (Los Alamos National Lab., Los Alamos, NM)

Nonlinear resonance ultrasonic spectroscopy (NRUS) is a nondestructive testing method that has been demonstrated to be highly sensitive to the presence of damage, especially in cases of progressive microcracking such as in mechanical or thermal fatigue. In controlled laboratory experiments, a characteristic strain-dependent hysteretic NRUS response can be determined via tracking of resonance frequencies. However, such results can be limited in their scope and reliant on controlled excitation sources. In recent NRUS experiments, we have incorporated machine learning to help address these limitations. Here, we present the nonlinear response of samples under pseudo-random excitation and discuss neural network predictions of structural status and experimental conditions, through training on 3-D frequency response functions. Using sparse sampling of a dataset which included mode- and stress-dependent nonlinear responses of a damaged component, structural status predictions using convolutional and graph neural network architectures had an accuracy of 85–90% in many cases, nearly four times better than random chance. This work is motivated by ongoing development of nuclear microreactors, which operate under more extreme conditions than conventional reactor designs, necessitating new control systems and sensor technologies to monitor for structural anomalies on the component level.

Session 3pPAb

Physical Acoustics and Computational Acoustics: Session in Honor of Richard Raspet II

Gregory W. Lyons, Cochair
gregwlyons@gmail.com

Keith Attenborough, Cochair
School of Eng. and Innovation, Open Univ., Milton Keynes MK7 6AA, United Kingdom

Jeremy Webster, Cochair
Los Alamos National Labs, LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544

W. C. K. Alberts, Cochair
CCDC-Army Research Lab., 2800 Powder Mill Rd., Adelphi, MD 20783

Chair's Introduction—12:55

Invited Papers

1:00

3pPAb1. Thermoacoustic properties of fibrous materials. Carl Jensen (Appl. Physical Sci., 530 Virginia Rd., Concord, MA 01742, crjensen21@gmail.com)

This work was done with Richard Raspet to confirm a parameterized model of the thermoacoustic properties of porous materials that he had proposed with Roh *et al*. J. Acoust. Soc. Am. 121, 1413–1422 (2007). Determining the thermoacoustic properties of these materials had proved difficult for previous students of Dr. Raspet's so I proposed to build a three-dimensional thermo-hydrodynamic simulation to interrogate their properties directlyi—the sort of thing you'd only attempt as a graduate student. Neither Rich nor I had experience with high-performance computing or computational fluid dynamics, but we did our best to examine the available numerical methods that captured the appropriate physics, find example cases to test the model's behavior, and, I think to Rich's surprise, eventually delivered a full simulation that interrogated the complicated physics of random media. Rich was a good sounding board as I tried to test numerical issues I did not understand, always supportive of my effort and always ready to listen to me complain. This work, and my PhD, could not have happened without him.

1:20

3pPAb2. Richard Raspet: Turbulence, acoustics, and a lifetime of scientific pursuit. Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, nmurray@olemiss.edu)

This review examines the collaboration between the author and Dr. Richard Raspet regarding turbulence contributions to dynamic wall-pressure signals. The narrative interweaves professional analysis with personal reflections, tracing a path from early connections through the author's grandfather, Dr. F. D. Shields, to shared cycling adventures, the author's Ph.D. hooding ceremony, collaborative research work, and ultimately the author's assumption of leadership at the National Center for Physical Acoustics. A poignant moment in 2023, sharing coffee with Dr. Raspet and Dr. Sabatier, provides a fitting culmination to this journey. Throughout a lifetime of scientific pursuit, Dr. Raspet exemplified an unwavering childlike curiosity that led naturally to greatness, rather than pursuing greatness as an end in itself. Henry David Thoreau's words resonate with Dr. Raspet's approach: "I wished to live deliberately, to front only the essential facts of life, and see if I could not learn what it had to teach." Dr. Raspet's steadfast dedication to the love and pursuit of nature and science created an atmosphere of awe that elevated the humanity of all who knew him.

1:40

3pPAb3. Richard Raspet's pioneering contribution and theoretical foundation to the study of wind noise. Jiao Yu (Liaoning Petrochemical Univ., 1 Dandong Rd., West Section, Fushun, Liaoning 113001, China, yujiaojoy@hotmail.com)

Richard Raspet conducted pioneering research in the field of wind noise and established a series of theories with significant influence. His theoretical foundation encompasses an in-depth analysis of the mechanisms behind wind noise generation, revealing the intricate processes of wind-wind and wind-object interactions that produce noise from the perspective of the intersection of fluid mechanics and acoustics. In terms of theoretical research, Raspet developed mathematical models capable of accurately predicting wind noise spectra based on wind speed measurements in outdoor turbulent flows, providing a powerful tool for wind noise prediction and spectral contribution analysis. His research has garnered wide attention in academia, fostering the advancement of wind noise studies and impacting

a variety of fields with practical engineering applications, significantly contributing to the reduction of wind noise. This presentation aims to detail Richard Raspet's pioneering contributions and theoretical foundations, offering insights and inspiration for future wind noise research. [Work supported by NSFC 12074160.]

2:00

3pPAb4. An overview of Richard Raspet's contributions to our understanding of low frequency wind noise. Jeremy Webster (Los Alamos National Labs, LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544, jwebster@lanl.gov)

Richard Raspet's research in wind noise took the acoustics community from a rudimentary understanding of how low-frequency wind noise is generated in pressure sensors to a solid theoretical understanding rooted in first principles of fluid dynamics. This work spanned three decades starting with quantitative studies and progressing through the development of a framework within which the complex interactions of turbulent pressures can be parsed and understood. This talk will present a non-technical overview of this work, which serves as a testament to Richard's tenacious approach to problem solving. It was the author's pleasure to have been a part of this challenging journey along with the graduate students who contributed considerably along the way.

2:20–2:40 Break

2:40

3pPAb5. Collaborative research with Richard (Rich) Raspet on wind induced ground vibrations. Craig J. Hickey (NCPA, Univ. of Mississippi, NCPA, 145 Hill Dr., University, MS 38677, chickey@olemiss.edu), Vahid Naderyan (Itasca, IL), Mohammad Mohammadi (Knowles Electronics, LLC, Itasca, IL, USA, Itasca, IL), and Md Abdus Samad (NCPA, Univ. of Mississippi, University, MS)

Dr Richard (Rich) Raspet was a colleague and friend for over 30 years, and I had the privilege of benefiting from his mentoring as a post-doctoral fellow, co-advising graduate students, working on collaborative research projects, and friendship. Rich was extremely passionate and thorough in his endeavors, both personal and professional. Working with Rich was "intense" and usually involved a dossier of handwritten derivations under my door waiting for me in the morning, followed by discussions over coffee, and subsequent discussions later in the day (or days) after I'd had time to digest his formulations. Rich focused on hard work and continuous forward progress on the problem at hand. In this presentation, I will summarize our progress on our final research on *wind induced ground vibrations*. Our goal was to understand this interaction to exploit wind noise as a useful seismic source. Rich was a strong advocate of publishing, and I often joked that "all his graduate students still owed him a publication." It is fitting that I share our final correspondences where I appear to be obligated for two papers!

Contributed Paper

3:00

3pPAb6. Richard Raspet's legacy in planetary exploration. Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, 240 Hebrard Blvd, Broussard Hall, Rm. 103, Lafayette, LA 70503-2067, andi.petculescu@louisiana.edu)

Richard Raspet's legacy in acoustics and physics is immense. His theoretical and experimental work on acoustic sensors in adverse conditions has helped scientific and engineering progress in numerous areas. Here, I focus on some of the implications of Richard's work in planetary exploration. For example, his work on microphones in the presence of winds was used for the Mars InSight wind-noise filter and the development of the microphone of the SuperCam instrument of the Mars Perseverance rover. Richard's research on ground motion produced by wind was used in isolating the purely seismic component of Mars's surface motion detected by the InSight seismic sensors. In a similar vein, his results on wind-generated low-frequency noise inside porous domes has led to a major study to explore the feasibility of deploying custom-designed porous domes on Mars as alternatives for rosette-type wind-noise filters for infrasound measurements. On a personal level, it was Richard who suggested, during a session break of an ASA meeting, that it may be worthwhile addressing infrasound absorption and dispersion in Earth's lower thermosphere; this brief interlude made helped cement my interest in atmospheric acoustics on Earth and beyond.

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

G. Christopher Stecker, Chair
Center for Hearing Research, Boys Town National Research Hospital,
555 N 30th St., Omaha, NE 68131

Chair's Introduction—1:00

Invited Paper

1:05

3pPP1. Advances in auditory neuroscience: Lessons learned from the last four decades. Fan-Gang Zeng (Univ. of California Irvine, Irvine, CA, fzeng@hs.uci.edu)

Auditory neuroscience came of age in the late 1980s, as part of the broader maturation of the neuroscience field, particularly marked by the establishment of the National Institute on Deafness and Other Communication Disorders in 1988. In this Hartmann Prize Lecture, I will share not only my research experiences and interactions as a graduate student in the late 1980s, but also my perspective on the advances in auditory neuroscience over the last four decades. Several paradigm shifts are evident, including the shift in research focus from the ear to the brain, the change in approach from linear to nonlinear systems analysis, and, importantly, the movement from treating basic and clinical questions as separate domains to fostering a fruitful interplay between basic and translational research. To illustrate these shifts, I will provide a global overview supported by literature review and analysis, along with specific examples drawn from psychophysics, physiology, cochlear implants, and gene therapy for hearing restoration. In honoring William M. Hartmann's engineering contributions, I will take a systems analysis approach, incorporating computational modeling and machine learning, to highlight both the advances and the opportunities in auditory neuroscience.

Session 3pSA

Structural Acoustics and Vibration: Constrained Layer Damping

Ian C. Bacon, Cochair
Physics & Astronomy, Brigham Young Univ., 333 W 100 S, Provo, UT 84601

Michael L. Dickerson, Cochair
MD Acoustics, 170 S William Dillard Dr., Ste 103, Gilbert, AZ 85233

Benjamin M. Shafer, Cochair
PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Chair's Introduction—12:55

Invited Papers

1:00

3pSA1. Scaling effects of frequency and material properties on the design of constrained layer damping. Benjamin S. Beck (Fluid Dynam and Acoust. Office, Penn State Appl. Res. Lab., P.O. Box 30, MS 3200D, State College, PA 16804, benbeck@psu.edu)

Engineers can prefer to non-dimensionalize effects to allow for understanding of a response of systems at various configurations, sizes, or speeds. Great insight can be found while interrogating the physics across independent variables. This work outlines the relationship between the physical scale and the stiffness and damping properties of viscoelastic materials within constrained layer damping when applied to thin structures. Previous work has shown that peak frequency of the damping of viscoelastic materials can be shifted to account for changes in scale of systems when damping is applied without a constraining layer. However, when the constraining layer is applied, the dominant deformation within the damping layer becomes shear instead of bending which controls the stiffness of the underlying thin, flexible structure. Here the methods of non-dimensionalizing the stiffness, loss factor, and geometry of a constrained layer system is presented along with a discussion on the effects of frequency to optimize the system for maximum vibration attenuation.

1:20

3pSA2. Treating noise in modern buildings using constrained-layer damping and current standardization. Benjamin M. Shafer (PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Constrained-layer damping (CLD) has been utilized to treat noise and vibration since the 1950s. The application of CLD in building materials such as gypsum wallboard, since the early 2000s, has been experimentally shown to have a significant effect on the sound transmission loss through building partitions over different one-third octave band frequency ranges, depending on the remaining building structure where it is applied. 20 years of experimental research indicate that while CLD, as applied in building construction, may drastically affect the sound transmission loss of partitions, many of these changes in sound isolation may fail to be observed when assessing performance using the sound transmission class (STC) according to standardized methodology. An overview of experimental data and observation of CLD as applied in modern buildings will be discussed as well as how the current standardized classification of sound isolation affects the understanding and application of this treatment in buildings. Possible solutions within the current standards body will be presented as well.

Contributed Paper

1:40

3pSA3. Subproblem optimization for constrained layer damping for a panel. Matthew Luu (Penn State, 446 Bluecourse Dr. (Apt. 907), State College, PA 16803, mbl5743@psu.edu) and Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., State College, PA)

This work explores the use of subproblem decomposition based optimization for constrained layer damping placement and construction within a structural acoustics system. In particular, how Benders decomposition with substructuring can be used to efficiently solve a structural acoustic optimization (SAO) problem compared to traditional methods. The methods are implemented in a parallel computing framework where the different subproblems may be distributed across multiple processors within a high-performance computing environment. The subproblem approach is shown to successfully compute an optimal design aimed at structural acoustic response of the system and comparisons to traditional optimization methods are made.

Session 3pSC

Speech Communication and Psychological and Physiological Acoustics: Speech Perception Beyond Intelligibility II

Melissa Baese-Berk, Cochair
Linguistics, Univ. of Chicago, 1115 E 58th St., Rosenwald Hall Rm. 203, Chicago, IL 60657

Susannah V. Levi, Cochair
Communicative Sciences and Disorders, New York Univ., 665 Broadway, 9th Fl., New York, NY 10012

Invited Paper

1:20

3pSC1. AAVE in the courtroom: Speaker role influences speaker perceptions. Sharese D. King (Linguist, Univ. of Chicago, 1115 E. 58th St., Chicago, IL 60637, sharesek@uchicago.edu) and Marisa Casillas (Comparative Human Development, Univ. of Chicago, Chicago, IL)

A growing body of work on the study of AAVE and its role in legal outcomes shows that speakers' whose voices are recognized as Black in such contexts are negatively evaluated. AAVE speakers are heard as less educated, comprehensible, and believable and are considered more likely to have been involved in a gang (Dunbar, King & Vaughn 2024; Rickford & King 2016). Related work has also shown effects of dialect on housing opportunities (Wright 2023; Purnell, Idsardi, & Baugh 1999) and job prospects (Hoffman, Kalluri, Jurafsky, & King 2024). Within the courtroom context, prior work has focused on AAVE speakers as witnesses and defendants. The current study examines how the specific legal role that a speaker fills alters perceptions of Black speech. A 35-year-old African American professor recorded 2-minute "role" passages—a mock closing argument ("attorney") and a mock testimony ("expert witness")—in both AAVE and Mainstream American English. Listeners linked AAVE to decreased clarity, increased identifications as Black, decreased education (especially in the expert witness role), and increased ratings of involvement in organized crime. Listeners identified the speaker as Black less often in the "attorney" role, shedding light on how dialect and courtroom status mitigate perceptions of Black speech.

Contributed Papers

1:40

3pSC2. Effects of dialect familiarity and dialect exposure on cross-dialect lexical processing. Marie Bissell (Univ. of Texas at Arlington, Arlington, TX), Cynthia G. Clopper (Ohio State Univ., Columbus, OH), and Abby Walker (Virginia Tech, 181 Turner St. NW, Blacksburg, VA 24061, ajwalker@vt.edu)

Dialect exposure has been shown to affect lexical processing, including faster and more accurate processing of familiar dialects than unfamiliar dialects. In addition, listeners with lifetime exposure to multiple dialects (i.e., multi-dialectal listeners) exhibit a more flexible processing strategy than listeners with lifetime exposure to a single dialect (i.e., mono-dialectal listeners), such that multi-dialectal listeners delay choosing among potential word candidates to allow them to resolve dialect-specific phonological ambiguity. The goal of the current study was to explore how dialect familiarity and multiple-dialect exposure interact in lexical processing. Mono- and multi-dialectal listeners in the Midland and Southern U.S. dialect regions participated in a cross-modal lexical decision task with auditory primes produced by General and Southern American English talkers. The prime-target pairs were selected to be phonologically ambiguous across the two stimulus dialects. Responses were faster overall for matching prime-target pairs (e.g., side, side) than unrelated pairs (e.g., side, pet), consistent with form-priming facilitation, and slower overall for competing prime-target pairs (e.g., side, sod) than unrelated pairs, consistent with minimal pair inhibition. The results suggest that whereas specific dialect familiarity is the primary predictor of facilitation, overall dialect exposure is the primary predictor of inhibition.

2:00

3pSC3. Intelligibility and listening effort for native and non-native French. Morgan Robertson (Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, morgan.robertson@ku.edu) and Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS)

Native and non-native listeners can identify words in semantically anomalous sentences more accurately when produced clearly by native and non-native speakers (Jung & Dmitrieva, 2023). The present study examines intelligibility and listening effort for two types of clear speech: General and Competitor. Stimuli for this experiment consist of productions from an experiment in which native and non-native French speakers interacted with a faux computer program that 'guessed' what they said with controlled responses (Maniwa *et al.*, 2009). When wrong, participants reproduced the words more clearly. When the program's response was "???" (What did you say?), clear productions were coded as General. When the program's response was a competitor (e.g., target: reine [ʁɛn], response: reins [ʁɛ̃]; target: beurre[bœʁ], response: peur [pœʁ]; target: dit [di], response: du [dy]), clear productions were coded as Competitor. In this perception experiment, participants are tasked with a 2AFC Identification-in-Noise task and an AX Discrimination-in-Noise task. Participants' accuracy will reflect intelligibility. Competitor tokens are expected to show the greatest intelligibility benefit. Participants' reaction times will reflect listening effort. While native clear speech is expected to require less listening effort, non-native clear speech may require more due to different clear-speech strategies employed. [Work supported by NSF Grant 2416141.]

2:20

3pSC4. Alignment of prosodic focus affects listening effort. Abbey L. Thomas (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., Minneapolis, MN 55455, abbeyt@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

One of the most persistent problems in studying listening effort is distinguishing differences in effort without the confound of differences in speech intelligibility. We address this issue using a design where listening conditions contain similar acoustic content, but where comprehension effort is mediated by prosody. Stimuli were question-and-answer pairs where the answer corrects or affirms information from the question. In most trials, the answer had prosodic focus on the novel information. However, in select trials, the prosodic focus was incorrectly placed on already-known information. We hypothesized that inappropriate prosody would not affect intelligibility, but would elicit lingering listening effort, marked by elevated pupil dilation in the moments after stimulus presentation. We present results from 18 listeners with typical hearing, and discuss potential implications for listeners with cochlear implants who struggle to hear prosodic contours. This experimental design with novel discourse-level stimuli holds promise for distinguishing effortful auditory encoding from effortful comprehension as well as for exploring how the benefits of prosody in speech perception extend beyond improving accuracy, processing speed, and recall, to make listening easier.

WEDNESDAY AFTERNOON, 21 MAY 2025 STUDIO FOYER, 1:00 P.M. TO 3:00 P.M.

Session 3pSP

Signal Processing in Acoustics: Signal Processing Poster Session

William F. Jenkins, Chair
Scripps Institution of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

Contributed Papers

3pSP1. A bi-axial velocity-sensor colocated with a pressure-sensor—Direction-finding imprecision due to the incident source's directional spreading. Mark L. Velasco (Dept. of Electronics and Comput. Eng., De La Salle Univ., Manila, NCR, Philippines), Kainam T. Wong (College of Comput. Studies, De La Salle Univ., Beihang Univ., New Main Bldg., D-1107, 37 Xueyuan Rd., Beijing 100083, China, ktwong@ieee.org), and Yue I. Wu (College of Electronics and Information Eng., Sichuan Univ., Chengdu, Sichuan, China)

This paper investigates the direction-finding precision, when the incident source suffers spatial/direction spreading, obtainable from a triad consisting of a biaxial velocity-sensor colocated with a pressure-sensor. Such a planar array has a low physical profile, thus easing affixation onto a mobile platform. The three component-sensors' spatial colocation mathematically decouples the incident wavefield's time-frequency dimensions from the polar-azimuthal directional dimensions. The statistical measure of direction-finding precision here is the Cramer-Rao lower bound of any unbiased estimate's variance.

3pSP2. Data-driven listening intention detection for hearables. Ruksana Giurda (Sonova AG, Eichwisstrasse 32, Hombrechtikon 8634, Switzerland, ruksana.giurda@sonova.com), Pascal Baumann, Seraina Glaus, and Simone Lionetti (HSLU, Rotkreuz, Switzerland)

Commercially available hearing aids are capable to automatically adapt the built-in "sound enhancing" technology according to the "acoustic scene." Individualization of this technology can be done with assisted or manual intervention. However, without continuous self-adjustment the mapping of "acoustic scene" and "sound enhancement" setting stays fixed, i.e., the hearing device does not adapt to an acute switch of the user's "listening intention." This ability, to selectively attend and switch attention between various acoustic sources, is a landmark of the healthy hearing system and seriously degraded even with a mild hearing deficit. In this work, we developed a system based on "listening intention detection" which can integrate data from multiple sensors to classify complex acoustic environments, infer the acoustic source attended by the listener, and adjust to individual user preferences. This is achieved through a comprehensive analysis of the user's surroundings and complemented with implicit feedback signals from the user himself, without introducing invasive sensors. The algorithm can be adjusted to meet specific technical constraints for edge devices.

3pSP3. Enhancing speech quality in hearing aids using matched filtering and log-spectral Wiener techniques for localization and binaural rendering in noisy environments. Fan-Jie Kung (Elec. Eng., National Taipei Univ. of Technol., No. 1, Sec. 3, Zhongxiao E. Rd., Da'an Dist., Taipei 106344, Taiwan, fjkung@mail.ntut.edu.tw)

This paper attempts to develop an improved real-time hearing aid system that integrates a microphone array for speech enhancement, sound source localization, and binaural rendering, specifically targeting individuals with hearing impairments in noisy environments like cocktail parties. The system leverages the benefits of microphone arrays, such as noise suppression and source localization. A matched filter is initially applied to minimize uncorrelated noise, and a log-spectral amplitude (LSA) Wiener filter is then employed to further reduce remaining noise and interference. The clean signal obtained is serves as the foundation for accurate direction-of-arrival (DOA) estimation. Four estimation techniques—steered-response power phase transform (SRP-PHAT), multiple signal classification (MUSIC), cosine similarity, and diagonal unloading (DU)—are assessed through Monte Carlo simulations. Following the DOA estimation, head-related transfer functions (HRTFs) are applied to the signal to produce binaural rendering. This proposed integrated system improves the clarity of speech and

enhances listening quality in complex acoustic environments, offering benefits for individuals with hearing loss.

3pSP4. Sound field recording and reproduction method using distributed spherical microphone arrays based on kernel ridge regression. Keita Yanagiya (Res. Inst. of Elec. Commun. / Graduate School of Information Sci., Tohoku Univ., 2–1–1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, yanagiya.keita.t2@dc.tohoku.ac.jp), Shuichi Sakamoto, and Seungmoo Jung (Res. Inst. of Elec. Commun./ Dept. of Eng., Tohoku Univ., Sendai, Japan)

The demand for sound field synthesis methods capable of reproducing wide areas is increasing. Ueno *et al*. (2018) proposed a technique that uses distributed microphones based on kernel ridge regression. However, it is difficult to synchronize all signals recorded by the microphones. In this study, we applied kernel ridge regression to signals captured by distributed spherical microphone arrays (SMAs) and proposed a sound field recording and reconstruction method that minimizes the effects of signal asynchronization and misalignment. In the proposed method, multiple SMAs are spatially arranged around the listening point to record the sound signals. The recorded signals are then transformed into spherical harmonic coefficients. Using these coefficients, secondary sound pressures at arbitrary positions are calculated. Finally, the sound field is reconstructed by applying kernel ridge regression to the calculated secondary sound pressures. The results of numerical simulations demonstrated that the proposed method reproduces an accurate sound field around the listening point. However, the findings also highlighted the importance of considering both the precision of the secondary sound pressures and the sufficiency of spatial information in the reconstruction region to achieve high reconstruction accuracy. These two factors involve a trade-off, especially in the high-frequency areas.

3pSP5. Augmented sound-image perception using pre-virtual-leading hypersonic signals with bass frequency envelopes. Ryota Imanaka (Ritsumeikan Univ., 2-150, Iwakura-cho, Ibaraki, Osaka 567-8570, Japan, is0545rs@ed.ritsumei.ac.jp), Yuting Geng (Ritsumeikan Univ., Ibarakishi, Osaka, Japan), Masato Nakayama (Osaka Sangyo Univ., Daitoshi, Osaka, Japan), and Takanobu Nishiura (Ritsumeikan Univ., Ibaraki, Osaka, Japan)

Recently, stereo audio technologies have been proposed to control sound-image localization based on binaural effects. The precedence effect, which occurs when sounds arrive from different directions with a slight time difference, can let listeners perceive a sound-image in the direction of the first arriving sound. Moreover, hypersonic effect suggests that inaudible high-frequency sounds influence audible sound perception. Considering the potential of ultrasound in sound-image control, we have previously proposed a sound-image augmentation method with ultrasonic signals named pre-virtual-lead signals. In this method, a sound-image is constructed with stereo technology, while the pre-virtual-lead signals are used to provide a virtual perception. The direction perception is augmented because the pre-virtual-lead signals can emphasize the direction of sound-image. The signals are designed as amplitude-modulated ultrasounds, based on the tactile perception of ultrasounds. This method has been demonstrated to be effective in limited conditions, where remains the possibility of improving left-right perception. This paper focuses on envelopes of pre-virtual-lead signals and proposes a sound-image control method using pre-virtual-lead signals with envelope frequencies in bass band. Previously, envelope frequencies were chosen based on vibrotactile sensitivity, but here we experimentally investigate the proper envelope frequencies in bass band considering the effect in auditory.

3pSP6. Evaluation of noise reduction performance of active noise control with optical laser microphone in reverberant environments. Maoto Mizutani (Ritsumeikan Univ., 2-150, Iwakura-cho, Ibaraki, Osaka 567-8570, Japan, is0580kk@ed.ritsumei.ac.jp), Kenta Iwai, Takanobu Nishiura (Ritsumeikan Univ., Ibaraki, Osaka, Japan), and Yoshiharu Soeta (National Inst. of Adv. Industrial Sci. and Technol., Ikeda, Osaka, Japan)

A feedforward active noise control system has a constraint on the processing and sound propagation delay. It is called the causality constraint. The feedforward active noise control system with an optical laser microphone has been proposed to relax the causality constraint. The optical laser microphone acquires the vibration velocity of the target noise with the speed of light. Therefore, the causality constraint is relaxed. However, the acquired signal contains little the reverberation characteristics of the room. Therefore, the noise reduction performance of the system is degraded in reverberant environment. Hence, this paper proposes a multichannel feedforward active noise control system for noise reduction in reverberant environments. The proposed system has the optical laser microphone and an air-conducted microphone as reference microphones. The unique point of this system is the additional air-conducted microphone, which acquires the reverberation components. The proposed system processes signals acquired by each microphone with separate noise control filters. This simultaneously relaxes the causality constraint and takes into account the reflection characteristics of the room. In this paper, the noise reduction performance of the proposed system is evaluated by the real-world experiments.

3pSP7. A deep reinforcement learning-based noise detection network for electric telescopic rods in electric tailgate systems. Zhehui Zhu (Tongji Univ., Cao'an Rd., Shanghai 201804, China, 2131577@tongji.edu.cn), Lijun Zhang, Kaikun Pei, and Dejian Meng (Tongji Univ., Shanghai, China)

As the automotive industry moves toward greater intelligence, electric tailgate systems have gained widespread adoption, offering features such as remote control and intelligent opening. However, noise issues in the electric telescopic rods, which serve as key actuators, negatively impact tailgate performance. This paper proposes a Noise Acoustic Detection Network (NADNet) for detecting noise in electric telescopic rods, based on Deep Reinforcement Learning (DRL). NADNet combines multi-dimensional features to accurately identify high-frequency electromagnetic noise, low-frequency impact noise, and periodic structural noise. Unlike traditional methods, NADNet dynamically adjusts noise detection strategies using DRL, allowing for automatic identification and classification of noise types in various noise environments without relying on labeled data. To overcome the challenge of limited data samples, NADNet utilizes synthetic samples generated through time/frequency transformations and data mixing techniques, improving classification accuracy. Experimental results demonstrate that NADNet outperforms traditional methods in noise detection and classification tasks for telescopic rods, providing effective data support for noise optimization in electric tailgate systems.

3pSP8. Machine learning on synthetic acoustic data using adversarial domain adaptation technique. Daniel Pereira (Appl. Acoust. Team, Los Alamos National Lab., 125 Central Park Square, Los Alamos, NM 87544, pereira.ufrgs@gmail.com)

Acousti Resonance Spectroscopy (ARS) measurements are attractive because they are highly sensitive to deviations in the structure's material, geometry, and environmental conditions. However, retrieving the property of interest is challenging due to the complicated changes to the spectra as the property of interest changes and sensitivity of the spectra to changes in other properties simultaneously. Deep learning on ARS measurements has shown to efficient and robust to identify trends that correlate with the property of interest. In many applications, it is infeasible to collect enough experimental ARS data to train a model so strategies to augment experimental data remain an open issue. Synthetic data from simulations are typically easy to acquire. Therefore, learning from synthetic data is very desirable in this context. We introduce a novel approach leveraging the Finite Element Method (FEM) framework to generate synthetic spectra in pressure estimation tasks. The FEM simulations were performed to mimic various vessel configurations at different pressures, representing 100% of the dataset used to train of the CNN model. As a crucial step, we implemented a domain adaptation technique to mitigate discrepancies between real and synthetic spectra.

3pSP9. A gated recurrent unit approach for improving audio quality in hearing induction loop systems. Ruiteng Li (Waseda Univ., 3 Chome-4-1 Okubo, Tokyo, Japan 169-8555, Japan, ruiteng.li@toki.waseda.jp) and Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan)

Hearing induction loop systems often encounter uneven field distribution, metal-induced signal loss, and overspill, which may reduce speech intelligibility. This study presents a digital signal processing framework

centered on gated recurrent unit (GRU) networks to address these challenges. Real-world noise measurements, including interference from power and data cables, are integrated into the training process to represent practical field conditions. By monitoring the signal-to-noise ratio, the GRU-based approach aims to reduce interference while preserving key speech cues. A simplified attention mechanism and spectral feature extraction are also incorporated, allowing the model to highlight crucial time-frequency segments. Preliminary tests, conducted in both simulated and real-world scenarios, suggest improvements in noise mitigation and intelligibility compared with baseline methods. These findings may help align theoretical low-spill designs with actual performance, ultimately supporting broader accessibility in venues such as lecture halls and event spaces.

3pSP10. Auditory-inspired adaptive frequency tracking and harmonic grouping. Vijay Peddinti (NUWC, Howell St., Newport, RI 02841, vijaykumar.peddinti.civ@us.navy.mil)

One of the best sound analyzers to date is the human (mammalian) auditory system, which has evolved through millions of years by resolving classification problems. It is a versatile, elegant. and powerful sound processing unit. It excels in detecting, estimating, and classifying multiple targets simultaneously even in noisy environments. Hence, mimicking even some known features of the auditory system could be beneficial in developing improved frequency tracking and classification algorithms. An auditory inspired adaptive synchrony capture filterbank (SCFB) signal processing architecture for tracking signal frequency components was proposed in an earlier paper [JASA-2013]. The SCFB exhibits many desirable properties for processing speech, music, and other complex sounds. The algorithm was modified using adaptive tuning parameters, and a generalized way to determine and suppress voiced and unvoiced (silent) regions. This modified algorithm estimates frequencies with higher accuracy even in the presence of closely spaced input tones [ASA-2024 presentation]. Recent work extended mimicking the auditory system further and in the process made the algorithm computationally efficient. In addition, the estimated frequency tracks are harmonically grouped (or clustered), which is a first step in sound source (or signal) separation. Preliminary analysis shows promising results. This presentation will focus on these latest updates.

3pSP11. Correlation analysis of room impulse responses for use in musical signal processing. Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, 617 Comput. Studies Bldg., 160 Trustee Rd., RC 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

Many important features of musical signals can be extracted from the auto- and cross-correlation functions of a recorded sound. In particular, both binaural localization and pitch estimation rely heavily on the locations of peaks in these correlation functions. However, in reverberant environments, these correlations depend on the respective autocorrelation functions of both the source and acoustic filter. Since the source signal's autocorrelation is often better understood or more easily estimated than that of the room, modelling the effects of room acoustics on these parameter estimates requires a more detailed understanding of room impulse response correlation functions. This work investigates the correlation properties of both monophonic and binaural room impulse responses from multiple databases, spanning a wide range of room sizes and configurations. Specifically, the cross-correlations between impulse responses recorded at different locations in the same room and between channels of binaural impulse responses are analyzed, with emphasis on the contribution of early reflections, low frequency modes, and late reverberation to these correlations. Finally, examples are used to illustrate the impact of these elements on the resulting pitch and localization cues.

3pSP12. Transforming and analyzing astrophysical signals through sound. Shaun Pies (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, spies@uno.edu) and Kendal Leftwich (Phys., Univ. of New Orleans, New Orleans, LA)

Astrophysical signals, such as pulsar emissions and gravitational waves, often contain valuable information about the fundamental properties of the universe. Translating these signals into the auditory domain provides an alternative approach for analyzing and interpreting their underlying structures. This project explores the conversion of astrophysical signals into audio waveforms and their subsequent analysis using acoustic signal processing techniques. By resampling and frequency-shifting the data, phenomena with inaudible timescales—such as pulsar spin periods or gravitational wave frequencies—are transformed into the human-audible range, preserving their temporal and frequency characteristics. Key astrophysical phenomena are studied acoustically, including pulsar timing irregularities and gravitational wave chirps. The resulting audio signals are analyzed to identify subtle patterns, such as glitches, harmonic resonances, or signal modulations, which may otherwise be obscured in traditional visual representations. This approach not only enhances scientific understanding but also facilitates public outreach by presenting complex astrophysical phenomena in an accessible and engaging format. Additionally, acoustic signal processing methods, such as spectrograms and wavelet analysis, are employed to draw parallels between the auditory and astrophysical domains. The study demonstrates the utility of sound as a tool for both research and education, offering new perspectives on the universe's dynamic processes.

3pSP13. Comparing the robustness to parameter mismatch of Infotaxis and Maximum A Posteriori search strategy. Muhammad Mudassir Jawaid (Elec. and Comput. Eng., UMass Dartmouth, 80 Cross Rd., Rockwood Apt., Dartmouth, MA 02747, mmudassir@umassd.edu) and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

This paper compares the robustness of Infotaxis and Maximum A Posteriori (MAP) search strategies under parameter mismatch scenarios. Vergassola *et al.* (2007) originally introduced Infotaxis as a passive sensing strategy inspired by moth odor tracking. Infotaxis maximizes mutual information by choosing measurements to minimize the expected uncertainty in the target's location. As a result, Infotaxis balances the exploration of its environment to gain new information with the exploitation of existing information. In contrast, MAP is a highly exploitative search strategy that makes decisions by relying on available information to guide the search. Recent studies extended Infotaxis' application to active sensing, demonstrating its superiority over MAP in reducing search duration at detection probability (P_D) below 0.8. To further investigate the robustness of Infotaxis and MAP, this study introduces parameter mismatches between the Bayesian update of the state vector and the sensor model. These mismatches occur when the estimated detection probability (η_D) differ from the sensor's actual P_D. We evaluate three sensor calibration conditions — over-calibrated ($P_D > \eta_D$), properly calibrated ($P_D = \eta_D$), and under-calibrated ($P_D < \eta_D$)—using 10 000 Monte Carlo simulations. Infotaxis consistently performs better than MAP under low P_D (<0.85) scenarios. Infotaxis also excels in under-calibrated conditions. [Work supported by ONR MURI program.]

3pSP14. Impact of beamwidth on Infotaxis search strategy performance. Marvin O. Mboya (Dept. of Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, mmboya@umassd.edu) and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

This presentation explores the impact of increasing active sonar beamwidth on the infotaxis search strategy (Vergassola, 2007) on the length of searches to find targets. Our model discretizes the search space into a one-dimensional grid. The associated state vector contains the probability that each grid cell contains the target. The search is modeled as a three-step iterative process: choosing the next sensing location (search strategy), measuring the environment, and Bayesian update of the state vector. The measurement is simplified as a Binary Hypothesis Test for the cell(s) within the sonar beam. Infotaxis search strategy is considered which maximizes the expected rate of information gain. Keith (2022) previously focused on a narrow sonar beam measuring a single grid cell. Increasing the beamwidth measures several grid cells at once, requiring a revised Bayesian update rule. The detection probability within the wider beam decreases moving away from the main response axis modeling the reduced power transmitted in off-axis directions. Simulations comparing different beamwidth infotaxis searches found that increasing the beamwidth allows infotaxis to reduce the expected number of iterations to find the target. [Funded by ONR MURI Program.]

3pSP15. Adaptive algorithm for holographic processing of broadband hydroacoustic signals. Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst. RAS, General Phys. Inst. RAS, Russia, Moscow, Vavilov St., GSP-1, d. 38, Moscow 119991, Russian Federation, kumiov@yandex.ru) and Sergey A. Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

Within the framework of numerical modeling, the adaptive algorithm for holographic signal processing of a sound source in a shallow water waveguide is verified. The sound source creates a stable interference pattern of intensity distribution (interferogram) at the receiver point in the frequency-time domain. The 2-D Fourier Transformation (2-D-FT) is applied to analyze the interferogram of the source moving in the waveguide. The 2-D-FT of the interferogram is referred to as a hologram (S. Pereselkov and V. Kuz'kin, JASA 151(2), 666–676). Adaptation within the framework of holographic signal processing is understood as the evaluation of processing parameters depending on specific propagation conditions. These processing parameters are aimed at the optimal estimation of the radial velocity and distance of the source. The evaluation of processing parameters implemented at the source detection step enables the determination of differences in the horizontal mode wavenumbers and their derivatives for various combinations of sound field modes. The proposed adaptation algorithm addresses the challenge of low-noise source localization in shallow water where acoustic calibration is not possible or where data about hydroacoustic conditions is unavailable. The adaptive algorithm significantly expands the applicability of holographic signal processing for the localization of low-noise sound sources. [This research was supported by a grant from the Russian Science Foundation (Grant No. 23-61-10024)]

WEDNESDAY AFTERNOON, 21 MAY 2025 STUDIOS 7/8, 1:00 P.M. TO 3:00 P.M.

Session 3pUW

Underwater Acoustics: Acoustics of Marine Sediments

Derek Olson, Chair
Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943

Contributed Papers

1:00

3pUW1. Spatial variability of geoacoustic parameters in SBCEX17. Florian Meyer (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093, florian.meyer@ieee.org) and William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA)

Since the ocean covers more than two-thirds of the planet and mud covers most of the seabed, understanding the acoustic propagation of mud is essential for acoustic performance prediction. We aim to determine the acoustic properties and their spatial variability on the New England Mud Patch by using data collected during the SBCEX17 experiment. In particular, we use tonal signals between 53 and 953 Hz emitted by sources towed on circular tracks and recorded by vertical line arrays. For Bayesian estimation, we use a new implementation of Metropolis-Hastings Markov chain Monte Carlo (MCMC) sampling that combines adaptive covariance estimation, sequential sampling in eigenvector space, and parallel tempering. Acoustic sound propagation is modeled by an adiabatic modes model that can provide a good tradeoff between inversion speed and modeling accuracy in the moderately range-dependent SBCEX17 environment. Our description of the seafloor includes a mud layer in which the sound speed increases moderately with depth and a thinner mud-sand transition layer where sound speed increases strongly due to a sand content that increases with depth. Preliminary results match well with established results based on broadband reflection-coefficient data. In addition to the spatial variability, we investigate the effects on uncertainty quantification of geoacoustic parameters when switching from a range-independent propagation model to the considered adiabatic modes model.

1:20

3pUW2. Geoacoustic inversion using seabed reflection-coefficient data variations across frequency. Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, BC V8W2Y2, Canada, minj@uvic.ca), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Jan Dettmer (Earth, Energy, and Environment, Univ. of Calgary, Calgary, AB, Canada)

Seabed reflection-coefficient (RC) data as a function of grazing angle and frequency provide rich information content to estimate seabed sediment layering structure and associated geoacoustic properties. To date, Bayesian inversions of wide-angle, broadband RCs have provided high-resolution geoacoustic profiles and uncertainty estimates using data sets based primarily on the Bragg resonance pattern across closely-spaced angles at a small number of frequencies. This paper considers instead the use of RCs across closely-spaced frequencies (including magnitude variations and fringe patterns), which could potentially provide higher information content on certain geoacoustic properties, such as attenuation and its frequency dependence. To investigate this, inversions are carried out based on RCs as a function of frequency from 1–6 kHz at a small number of grazing angles (multiple angles are required to overcome the ambiguity between layer sound speed and thickness). Inversions are evaluated and compared for specific choices of grazing-angle regime, including low angles, high angles, and near the critical angle. [Work supported by the Office of Naval Research.]

1:40

3pUW3. Deployment of a single frequency acoustic attenuation system for measuring fine suspended sediments in stream channels. Brian Carpenter (National Ctr. for Physical Acoust., The Univ. of MS, 145 Hill Dr., University, MS 38677, wocarpen@olemiss.edu), Bradley Goodwiller (School of Eng., The Univ. of MS, University, MS), Daniel Wren (ARS-NSL, USDA, Oxford, MS), and John D. Heffington (National Ctr. for Physical Acoust., The Univ. of MS, University, MS)

The use of ultrasonic acoustic technology to measure the concentration of fine suspended sediments has the potential to greatly increase the temporal and spatial resolution of sediment measurements while reducing the need for personnel to be present at gauging stations during storm events. In collaboration with the U.S. Department of Agriculture, The National Center for Physical Acoustics at The University of Mississippi has developed a single frequency acoustic attenuation system to monitor the concentration of suspended fine sediments (less than 60 micrometers in diameter) in rivers and streams. The field unit consists of two immersion ultrasonic transducers measuring attenuation of 20 Megahertz acoustic signals propagated through suspended particles. The system was operated in the Goodwin Creek Watershed near Batesville, Mississippi, USA from November 2019-Februry 2023. The calibrated acoustic measurements of fine sediment concentration agreement well with discharge and pump samples while providing greatly improved time resolution in the data. The basic design parameters, calibration methodology, and examples of results of field use of the prototype will be presented.

2:00

3pUW4. Frequency-dependent attenuation and multiple scattering in granular materials. Colton A. Kawamura (Phys., Naval Postgrad. School, 209 Emerald Hill, Franklin, NC 28734, colton.kawamura1@nps.edu), Abe Clark (Naval Postgrad. School, Monterey, CA), and Derek Olson (Oceanogr., Naval Postgrad. School, Monterey, CA)

We employ discrete element simulations to investigate the phase speed and attenuation of acoustic waves in dissipative granular packings, a topic with broad implications for geophysical and engineering applications. The study focuses on grain-scale mechanisms underlying the anomalous frequency dependence observed in experiments and previous simulations. Our results reveal that this anomalous attenuation arises from non-affine grain motion, characterized by oscillations both along the wave's polarization and in perpendicular directions. As granular packings approach more realistic configurations—with stiffer particles and minimal inter-grain overlap—perpendicular oscillations increase in amplitude and phase inhomogeneity. Through theoretical analysis, we demonstrate that these transverse motions account for the observed anomalous attenuation scaling, offering a grain-scale perspective on why attenuation exceeds expectations in dissipative granular packings.

2:20

3pUW5. Optimal transport for full waveform seabed property inversion. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu)

Processing waveforms measured at one or more hydrophones in the ocean provides us with a means for conducting geoacoustic inversion. By comparing these waveforms to replica waveforms calculated with a sound propagation model and minimizing the distance between real and replica data, one can identify the most suitable property values for the propagation medium. Typically, distance measures such as the Euclidean distance are employed. However, such measures present us with multiple local minima in addition to the sought after global minimum. Search approaches are challenging to implement, as "trapping" in regions of local minima can hinder global optimization. Optimal transport, relying on the minimization of different similarity measures, provides us with a function that is easier to navigate, being characterized by different properties from those of the Euclidean distance. In this work, we illustrate how optimal transport can facilitate inversion for seabed parameter estimation and we compare results employing optimal transport to those obtained from conventional approaches. [Work supported by ONR.]

2:40

3pUW6. A fast complex ray approximation for modeling scattering from layered seafloors. Derek Olson (Oceanogr., Naval Postgrad. School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943, dolson@nps.edu), Dajun Tang (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR)

Most measurements of seafloor scattering are performed using compact sources. When the seafloor exhibits complex geoacoustic structure, such as layering with roughness, the plane wave sonar equation model is inaccurate, and modeling these measurements must be performed in the time-domain. Recent work on this topic has resulted in a small-roughness perturbation theory approach that models the time-evolution of a pulse and explicitly takes into account the layering structure. This method requires numerical evaluation of three highly oscillatory integrals for each layer interface and can be prohibitively expensive to use in inverse methods. Here, we compare several approximations to this approach. The first is to use a modified version of the free space Green's function to account for the extra phase traveling through the sediment. The second is a to use the complex ray approximation developed by Westwood to the layered media Green's function. In this approximation, up to two solutions are found for each depth and range point in the integral. The approximations are compared with the exact solution and implications for modeling real data are discussed.

Silver Medal in Physical Acoustics

ASA
ACOUSTICAL SOCIETY
OF AMERICA

Richard Raspet

2024

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Isadore Rudnick	1975	Gregory W. Swift	2000
Martin Greenspan	1977	Philip L. Marston	2003
Herbert J. McSkimin	1979	Henry E. Bass	2006
David T. Blackstock	1985	Peter J. Westervelt	2008
Mack A. Breazeale	1988	Andrea Prosperetti	2012
Allan D. Pierce	1991	Evgenia A. Zabolotskaya	2017
Julian D. Maynard	1994	James M. Sabatier	2019
Robert E. Apfel	1997	Michael R. Moldover	2020

ENCOMIUM FOR RICHARD RASPET

… for contributions to understanding outdoor sound propagation, acoustics of porous media, and thermoacoustics

21 MAY 2025 • NEW ORLEANS, LOUISIANA

Richard Raspet was born in upstate New York and moved with his family to Starkville, Mississippi where he graduated from Starkville (Mississippi) High School. He earned a B.S. with distinction in Physics from Rhodes College and M.S. and Ph.D. degrees in Physics from the University of Mississippi.

He joined the Physics faculty of the University of Mississippi in September 1987 as an Associate Professor and was promoted to Full Professor in 1994. He also held an appointment as a National Center for Physical Acoustics Principal Scientist. Before joining the Physics Department, he was a Physicist on the Acoustic Team at the U.S. Army Construction Engineering Research Laboratory in Champaign, Illinois and was appointed Adjunct Professor of Electrical and Computer Engineering at the University of Illinois.

Rich was a leader in experimental, theoretical, and numerical research in the propagation of shock waves and blast noise. He contributed extensively to outdoor sound propagation above a ground of finite impedance in the presence of wind and temperature gradients and turbulence. He also contributed to the mitigation of blast noise and the interaction of sound with porous media. In more recent years his research focused on wind noise reduction and wind noise theory at audible and infrasonic frequencies.

His research on shock wave and blast noise propagation includes the effect of finite ground impedance, the effect of vibrational relaxation rise times, the reduction of blast noise with aqueous foam, and diffraction of the explosive transient. Dr. Raspet was a leading expert on shock waves and blast noise. He authored the chapter, "Shock waves, blast noise and sonic booms," in the *Encyclopedia of Acoustics*, edited by M. Crocker.

His research on outdoor sound propagation addressed a broad array of topics, including the scattering of sound by atmospheric turbulence, the acoustic surface wave above a complex impedance surface, the propagation of sound in a shadow zone, and the influence of wind and temperature gradients and turbulence on sound propagation. Of particular importance, he introduced the numerical method called the Fast Field Program (FFP) to the field of outdoor sound propagation. This was a major milestone to the field. It allowed for the development of different implementations of the FFP and development of various alternative numerical techniques by research groups in many countries. Though many of the groups share results at international meetings, the codes were generally developed independently. Thus, Members of NATO Panel 3 Research Study Group 11 began an effort to compare predictions. The goals were to identify any differences and sources of differences and to provide a set of results which future researchers can use to check new models and code. This led to the publication of Benchmark cases in *JASA*. Dr. Raspet was a major contributor to this effort, both in defining the Benchmark cases and in calculating all the Benchmark cases using the FFP.

Rich published several papers on thermoacoustic engines, an indication of the breadth of his work. He conducted foundational work uniting the Biot theory of sound propagation in porous materials to theoretical work in thermoacoustics. In this work the acoustic field quantities and second order energy flow were considered for sound propagation in arbitrary shaped pores. Determining the complex wave number and complex impedance for the thermoacoustic porous element supporting a temperature gradient, known as the 'stack', allowed for familiar modeling from the perspective of sound propagation in porous media, mainly impedance and pressure translation through the stack and heat exchangers as well as open resonator elements of the thermoacoustic device. Pore shapes of different regular and arbitrary geometry and boundary layer effects were computed and compared in a model thermoacoustic engine. Pore shape factors are used in these models to scale the porous thermoviscous effects between random and circular pores and are readily adapted to the field of thermoacoustics. He extended this approach to include random porous media and fibrous media to support experimental efforts to use stacked screens, fiberglass, and reticulated vitreous carbon and aluminum foams as thermoacoustic elements.

In addition, Rich was instrumental in leading the effort to include the effects of evaporation-condensation on sound propagation in porous media with and without a temperature gradient. He experimentally observed that the presence of a volatile liquid-vapor in an inert gas mixture greatly changed the temperature gradient needed for the onset of spontaneous thermoacoustic oscillations in an open-air engine. The earlier theoretical work on inert gas-vapor effects on sound propagation in porous media with temperature gradients enabled a foundation to study the use of evaporation-condensation effects in thermoacoustic engines and refrigerators.

Rich was a Fellow of the Acoustical Society of America (ASA) and a member of Sigma Pi Sigma, Phi Kappa Phi, Sigma Xi, and the ASA Physical Acoustics Technical Committee He served as an Associate Editor of *The Journal of the Acoustical Society of America* (JASA) and as a member of the editorial board of *Applied Acoustics*. He was a reviewer for JASA, *Noise Control Engineering*, *Nature, Science, The Journal of Geophysical Research, Climate and Physics of the Atmosphere, Applied Acoustics, and Acta Acustica*.

Rich's research and teaching career has been widely recognized. While working at the US Army Construction Engineering Research Laboratory, he received numerous awards recognizing Sustained Superior Performance and Researcher of the Year Award. In 2007, he received the University of Mississippi's Distinguished Faculty Fellowship Award for his contributions to teaching. He mentored many undergraduate, graduate and post-graduate students who say he expected them to have a deep understanding and appreciation of the beauty of physics. When it came time for exams, he would respond to questions about how to be prepared by saying "know everything."

In honoring Richard Raspet posthumously, the Acoustical Society of America recognizes his sustained effort in the field of Physical Acoustics and his valuable leadership in teaching and the broad field of acoustics.

JAMES M. SABATIER
GILLES A. DAIGLE

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Psychological and Physiological Acoustics

Andrew J. Oxenham

2024

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Lloyd A. Jeffress	1977	Brian C. J. Moore	2002
Ernest Glen Wever	1981	H. Steven Colburn	2004
Eberhard Zwicker	1987	William A. Yost	2006
David M. Green	1990	Roy D. Patterson	2015
Nathaniel I. Durlach	1994	Ruth Y. Litovsky	2021
Neal F. Viemeister	2001		

von Békésy Medal

The von Békésy Medal is presented to an individual, irrespective of nationality, age, or society affi liation, who has made an outstanding contribution to the science of psychological and physiological acoustics, as evidenced by publication of research results in professional journals or by other accomplishments in the field.

PREVIOUS RECIPIENTS

Jozef J. Zwislocki	1985
Peter Dallos	1995
Murray B. Sachs	1998
William S. Rhode	2010
M. Charles Liberman	2012

ENCOMIUM FOR ANDREW J. OXENHAM

…for contributions to understanding the perception of pitch, auditory stream formation, context effects, and masking

NEW ORLEANS, LOUISIANA • 21 MAY 2025

Andrew Oxenham was born in Boston, MA, but left (with his parents!) when he was about 18 months old. He spent two years in Ankara, Turkey, before moving to Brighton, UK, where he completed all of his schooling. He excelled at mathematics and physics, but most enjoyed music (playing classical piano, church organ, and bass guitar in a band), which led to him take a "Tonmeister" degree in Music and Sound Recording at the University of Surrey, UK. This course included a year working for West German Broadcasting (WDR) in Cologne, where he met Sylvia, later to be his wife. His interest in hearing started when he did a final-year undergraduate research project at the British Broadcasting Corporation (BBC) Research Department, running perceptual evaluations of low bit-rate audio coding, which formed the basis of the MP3 standard.

Andrew then applied to the Engineering Department at Cambridge to do a Master's degree in signal processing, but, fortuitously, I was a member of the panel that interviewed him for admission. I quickly realised that he had exceptional abilities, and after the interview I told him that he would be much better off doing a Ph.D. under my supervision.

Upon finishing his Ph.D., Andrew married Sylvia in 1995 and started a two-year postdoc supported by a Wellcome Trust International Prize Fellowship to work with Armin Kohlrausch at the Institute for Perception Research, Eindhoven, The Netherlands. He moved to Boston at the end of 1997 and spent 18 months at Northeastern University with Soren Buus and Mary Florentine. In 1999 he started his own research group within MIT's Research Lab of Electronics. Andrew moved to Minnesota in 2006, where he is currently a Distinguished McKnight University Professor, Director of the Center for Applied and Translational Sensory Science (CATSS), and Associate Chair for Research for the Department of Psychology.

Andrew's Ph.D. work was concerned with auditory forward and backward masking (masking occurring when the signal follows or precedes the masker), and especially how the two combine. He showed how differences in the additivity of forward and backward masking between hearing-impaired listeners and normal-hearing listeners could be explained in terms of differences in compression in the peripheral auditory system. The work on cochlear compression was continued in a collaboration with Chris Plack that started during Andrew's Ph.D. viva, and that resulted in several influential papers on methods for estimating cochlear compression. Andrew also conducted important research on how cochlear compression influences the intelligibility of speech in steady and fluctuating background sounds.

A second area in which Andrew has made very significant contributions is the perception of pitch. It was believed for several decades that the pitch of complex tones is dominated by low harmonics because those harmonics are resolved ("heard out") in the auditory system. However, Andrew, in work together with his then Ph.D. student Joshua Bernstein, showed that the key factor appeared to be harmonic number rather than resolvability. This was demonstrated via the ingenious use of dichotic presentation, with even harmonics presented to one ear and odd harmonics to the other. In another important contribution, Andrew showed that the harmonics of a complex tone need to be presented at the "correct" place in the cochlea for a pitch corresponding to the missing fundamental to be achieved. This has important implications for the perception of pitch by people with cochlear implants. In a series of novel studies, Andrew and co-workers showed that, contrary to prevailing beliefs, a pitch corresponding to the "missing fundamental" could be perceived when all harmonics fell above 6 kHz. This has important implications for theories of pitch perception. Andrew has also published important papers on the interaction of pitch and timbre, the perception of consonance and dissonance, the perception of and mechanisms underlying the detection of frequency modulation and the characteristics of congenital amusia.

A third general area in which Andrew has made significant contributions is in the characterization of the frequency selectivity of the human auditory system, using both behavioral measures and measures based on otoacoustic emissions. He showed that tuning was sharper than had traditionally been believed, especially for low levels and high frequencies.

Last but not least, Andrew has made very significant contributions to the understanding of two different but related phenomena: auditory enhancement, which reflects the sensitivity of the auditory system to changes in spectrum over time; and auditory stream segregation, which reflects the ability of the auditory system to group together sound sequences emanating from a single source and to segregate them from sounds that emanate from other sources. He has performed many experiments to clarify the mechanisms that underlie auditory enhancement, for both normal-hearing listeners and listeners with impaired hearing or cochlear implants. He has clarified the factors underlying stream segregation, showing that segregation can occur for complex tones with unresolved harmonics. These results are important both for their theoretical implications and for their applications to improving signal processing for hearing aids and cochlear implants.

In addition to his own achievements, Andrew has been a great mentor to students and post-docs. To quote one such person: “He is absolutely brilliant, creative, and very hard working. He finds ingenious ways to overcome any obstacles to answer his ever-evolving research questions. He inspires students, postdocs, and other researchers around him through his infectious enthusiasm for research.”

Andrew has been a strong supporter of the ASA and has made significant contributions to the ASA. He served on the P&P Technical Committee (2001-2004), was an Associate Editor of the *Journal of the Acoustical Society of America* (JASA) (2004-2007), Chair of the P&P Technical Committee (2008-2011), member of the Executive Council of the ASA (2015-2018), and a member of the ASA Medals and Awards Committee (2018-2022). He is a regular attendee at ASA meetings. Andrew was elected a Fellow of the ASA in 2003, and has received many awards, including the R. Bruce Lindsay Award of the ASA (2003) and the Troland Research Award of the National Academy of Sciences (2009).

In summary, Andrew has made tremendous contributions to knowledge and education in psychological acoustics and he has given outstanding service to the ASA. I congratulate him most warmly on the award of the Silver Medal in Psychological and Physiological Acoustics of the Acoustical Society of America.

BRIAN C. J. MOORE

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Speech Communication

Jody E. Kreiman

2025

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Franklin S. Cooper	1975	Katherine S. Harris	2005
Gunnar Fant	1980	Ingo R. Titze	2007
Kenneth N. Stevens	1983	Winifred Strange	2008
Dennis H. Klatt	1987	David B. Pisoni	2010
Arthur S. House	1991	Sheila E. Blumstein	2014
Peter Ladefoged	1994	John J. Ohala	2015
Patricia K. Kuhl	1997	Anne Cutler	2020
		Joanne L. Miller	2024

ENCOMIUM FOR JODY KREIMAN

… for contributions on the perception and production of the human voice

21 MAY 2025 • NEW ORLEANS, LOUISIANA

Jody Kreiman mostly grew up in southern California; her father worked in film processing, eventually becoming president of DeLuxe Laboratories in Hollywood. Her mother was a homemaker and ardent volunteer. Jody's career shows the influence of both parents.

Jody received her BA in Linguistics from Brown University, where her interest in "what was said and how it was said" found a home in Brown's program in Semiotics. She received her PhD in Linguistics in 1987 from the University of Chicago, though after her Chicago adviser left the university, she wrote her dissertation in the UCLA Phonetics Lab, where she was unofficially mentored by Peter Ladefoged (ASA Silver Medal in Speech Communication 1994) and Diana Van Lancker Sidtis (now retired from NYU). Jody described it this way in 2006: *"Peter took me in, gave me an office and run of the facilities, supervised my thesis without credit, and never once asked me when I was going back to Chicago (although a few months after I finally finished my dissertation he did say, "Dear, don't you think it's time you looked for a job?"). (…) It still seems extraordinary to me that he saw merit in the naïve, unformed student who wandered in his door, but somehow he did, and that made all the difference to me."* Right after Peter's question, a serendipitous post-doc offer from Bruce Gerratt led Jody to move across campus to the Department of Head and Neck Surgery in the School of Medicine. Jody set up a voice research lab, the Bureau of Glottal Affairs—a name saluted as Research-Bureau-of-the-Month by the Mini Annals of Improbable Research. She has spent her career in that department and lab, and has recently been promoted to Distinguished Professor in Residence.

Her most noteworthy research product is her 2011 book with Diana Sidtis, *Foundations of Voice Studies*. This mammoth compendium and synthesis of knowledge about the human voice remains an extremely valuable reference work, impressively cited, and the winner of an award for best book in Language and Linguistics.

Perhaps her most significant recent contribution to the study of the human voice has been her leadership (with several collaborators) in providing a tractable, low-dimensional model of the acoustic spectrum of the human voice. The harmonic and noise structure of the human voice source is complex, and varies greatly across individual speakers. Over twenty years ago, Jody's staff developed an acoustic speech synthesis tool specifically for the human voice source (the UCLA Voice Synthesizer). She then (with Bruce Gerratt) demonstrated that a voice sample could be usefully characterized by finding the synthesis parameter values that result in a copy of the voice perceptually indistinguishable from the original. That is, the settings needed to synthesize the copy are a description of the voice quality. However, the large number of available parameters make an unwieldy description, and it was not clear which ones listeners, in fact, attend to. Data reduction of a large-scale acoustic analysis of many voice samples suggested that as few as four harmonic parameters could characterize different voices, and a series of perception studies then showed that human listeners could rely on these to distinguish voice samples. Thus, we now have a model of the human voice source that is not only acoustic, but psychoacoustic—known to be perceptually important to human listeners in discriminating and recognizing voices.

What makes her research program especially valuable is not only its breadth, but its goal (with Zhaoyan Zhang) of bringing together all the different strands about voice—physiology, production, perception, measurement, and synthesis—so that the cause-and-effect connections from physiology to perception become computationally implemented and well-understood.

Jody has published 26 papers in *The Journal of the Acoustical Society of America* (JASA) and has given over 90 presentations at ASA meetings. She has had career-long research and training grants from NSF and/or NIH. She was made a fellow of the Society in 2006, and then of the American Speech Language and Hearing Association in 2015, indicative of her diverse interests and strong work in both the research and clinical applications of voice science. And there are also Hollywood applications! Jody has given invited presentations about voices at both PIXAR and Disney.

Jody has established herself, and her research group at UCLA, as pre-eminent in the study of the human voice. She also has connections to other disciplines and departments across the UCLA campus: in addition to teaching and supervising in her own department, she has been a popular mentor of graduate students and postdocs in other departments, and she has participated in several cross-department NIH training grants. She maintains long-term research collaborations on voice with Abeer Alwan in Electrical Engineering, Pat Keating in Linguistics, and Nina Eidsheim in Musicology. We and our students have been fortunate to benefit from having Jody as a colleague.

Jody's professional service has been extraordinary—like her mom, Jody is a volunteer. Her extensive service to the Acoustical Society includes activities for the Society's meetings (TPOMs, poster judging, chairing sessions) and publications (serving as an Associate Editor for the *Journal*, putting together an issue on Speech Communication for *Acoustics Today*). Most notably, she has been a member of several ASA committees: Vision 2010, the Strategic Leadership for the Future summit, the Committee to examine the financial stability of the ASA, the Nominating Committee, and the Finance Committee. Outside the Society, she also has significant editorial service to a range of journals, and is a frequent reviewer of grant proposals. At UCLA in 2019, she was the first recipient of a new university service award because of her longstanding and wide-ranging service on university committees.

We are delighted that the Acoustical Society recognizes and honors Professor Jody Kreiman's outstanding record of research accomplishments and service with the Society's Silver Medal in Speech Communication for contributions on the perception and production of the human voice. Congratulations, Jody!

PATRICIA KEATING

R. BRUCE LINDSAY AWARD

ASA
ACOUSTICAL SOCIETY
OF AMERICA

Andrea P. Arguelles

2025

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is no more than 10 years post terminal degree on 1 July at the time of Award acceptance and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942
Leo L. Beranek	1944
Vincent Salmon	1946
Isadore Rudnick	1948
J. C. R. Licklider	1950
Osman K. Mawardi	1952
Uno Ingard	1954
Ernest Yeager	1956
Ira J. Hirsh	1956
Bruce P. Bogert	1958
Ira Dyer	1960
Alan Powell	1962
Tony F. W. Embleton	1964
David M. Green	1966
Emmanuel P. Papadakis	1968
Logan E. Hargrove	1970
Robert D. Finch	1972
Lawrence R. Rabiner	1974
Robert E. Apfel	1976
Henry E. Bass	1978
Peter H. Rogers	1980
Ralph N. Baer	1982
Peter N. Mikhalevsky	1984
William E. Cooper	1986
Ilene J. Busch-Vishniac	1987
Gilles A. Daigle	1988
Mark F. Hamilton	1989
Thomas J. Hofler	1990
Yves H. Berthelot	1991
Joseph M. Cuschieri	1991
Anthony A. Atchley	1992
Michael D. Collins	1993
Robert P. Carlyon	1994
Beverly A. Wright	1995
Victor W. Sparrow	1996
D. Keith Wilson	1997
Robert L. Clark	1998
Paul E. Barbone	1999
Robin O. Cleveland	2000
Andrew J. Oxenham	2001
James J. Finneran	2002
Thomas J. Royston	2002
Dani Byrd	2003
Michael R. Bailey	2004
Lily M. Wang	2005
Purnima Ratilal	2006
Dorian S. Houser	2007
Tyrone M. Porter	2008
Kelly J. Benoit-Bird	2009
Kent L. Gee	2010
Karim G. Sabra	2011
Constantin-C. Coussios	2012
Eleanor P. J. Stride	2013
Matthew J. Goupell	2014
Matthew W. Urban	2015
Megan S. Ballard	2016
Bradley E. Treeby	2017
Yun Jing	2018
Adam Maxwell	2019
Julien Bonnel	2020
Likun Zhang	2021
Meaghan A. O'Reilly	2022
Julianna C. Simon	2023
Christopher Kube	2024

ENCOMIUM FOR ANDREA P. ARGÜELLES

… for contributions to the understanding of ultrasonic interactions with heterogeneous materials

21 MAY 2025 • NEW ORLEANS, LOUISIANA

Andrea Paola Argüelles grew up in Venezuela, one of six children, and moved with her family to the U.S. in 2007. With a high aptitude for mathematics and physics, and parents who were both engineers, Andrea was drawn to the Mechanical Engineering program at the University of Texas Pan American (currently the University of Texas Rio Grande Valley [UTRGV]), where she enrolled at the age of 16. There she thrived in the challenging environment and her hard work and excellent grades caught the attention of Professor Constantine Tarawneh in only her first semester. He nudged Andrea toward undergraduate research and she encountered her first role model, mentor, and research leader Professor Karen Lozano. This relationship gave Andrea the motivation to pursue an academic career.

In the last semester of Andrea's BS degree, Professor Tarawneh convinced her to join his growing research group which focused on railroad bearing safety which had a clear societal impact. She eagerly chose to continue her education at the graduate level. Andrea's MS research was focused on the design and testing of a low-cost device to measure the temperature and vibrations of railroad bearings with potential for condition monitoring. This research involved extensive design, experimental measurements, and validation using other sensors. This project also allowed Andrea to be a part of a large team with each member contributing to the overall goal.

Near the end of her MS, Andrea decided to continue toward her academic career goal and pursue her PhD. She made a campus visit to the University of Nebraska-Lincoln (UNL), Professor Tarawneh's alma mater. There, Andrea was introduced to the research group of Professor Joseph Turner who studied ultrasonic scattering within polycrystalline media and other nondestructive evaluation (NDE) topics, including railroad applications. She accepted a research assistantship for an industry project focused on ultrasonic inspection for steel quality of railroad bearings. For that project, Andrea and other students travelled to the bearing manufacturer to scan hundreds of bearing components ultrasonically. These results would help establish inspection guidelines for the manufacturing process that are used today to qualify steel suppliers. That experience helped Andrea see the important role that ultrasonic inspections could play for manufacturers.

During her PhD studies, Andrea was highly motivated and driven to produce quality results as she learned more about wave propagation through heterogeneous media and the associated scattering which led Andrea to expand ultrasonic scattering models for polycrystalline media. Her most impactful article, published in the *Journal of the Acoustical Society of America* (*JASA*) in 2017, studied the influence of the grain size distribution on the ultrasonic response. Prior models focused solely on the mean grain size, but Andrea showed that the distribution width was a critically important aspect that was often overlooked. Her model is now used by researchers around the globe to improve the interpretation of experimental data. Andrea also extended ultrasonic scattering models to include mode conversion, grain elongation, and address measurements with arbitrary configurations of transducers. At UNL, Andrea was a very positive influence on the entire research group with her drive for success and her optimistic outlook. Her leadership skills allowed her to direct research topics herself and easily collaborate with others – a remarkable mindset for a PhD student. Andrea's time at UNL was also personally life changing. There she met Christopher Kube, a fellow student in Turner's group, whom she would marry in 2018. Not only her husband, Kube is now her collaborator and department colleague at the Pennsylvania State University, where they welcomed son Jack in December 2023.

In 2018, Andrea joined the Engineering Science and Mechanics department at Penn State, excelling at the missions of teaching and research. Andrea has received numerous accolades from the American Society of Engineering Education and the American Society for Nondestructive Testing. Andrea's excellence in student mentorship is also evident from the various awards to her students, including many national graduate fellowships, undergraduate scholarships, and numerous awards for presentations and posters at international conferences. Andrea has been recognized through several awards at Penn State for her teaching, research and student advocacy.

Andrea is now an internationally recognized leader in ultrasonic interactions with complex materials and structures with an emphasis on characterization. Her expertise spans several manufacturing processes including additive manufacturing, binder jetting, and cold sintering, all of which can create particularly challenging material systems especially from a quality control standpoint. Andrea has a keen ability to address industry challenges using sophisticated models that are tailored to specific problems of interest. One renowned NDE researcher noted that Andrea is "an exceptional young scholar and a rising star in the field of ultrasonics and acoustics." Andrea's great ideas, hard work, and dedication to her students led to an NSF CAREER award for which she will improve ceramic processing science through acoustic characterization. As part of this grant, Andrea will create Girls Learn About Sound in Solids (GLASS), a program for middle and high school girls, through a partnership with Girls Code the World, in which students will learn the principles of acoustics and engineering applications.

Andrea's involvement with the ASA, her primary professional community, is outstanding. She is a member of the Physical Acoustics Technical Committee, the Structural Acoustics and Vibration Technical Committee and has reviewed for *JASA*, *JASA-Express Letters* as well as many other journals related to NDE, materials, and ultrasonics. Four of her undergraduate students received the ASA Robert W. Young Award. Andrea was an instructor at the 2024 ASA School, she has presented invited lectures at ASA meetings, and she regularly organizes and chairs special sessions. Andrea's strong reputation as an advocate for inclusivity led to her appointment as chair of the ASA Committee to Improve Racial Diversity and Inclusivity (CIRDI) in 2023 – a remarkable accomplishment for someone so early in her career. One prominent ASA fellow has been impressed with Andrea's "drive to make the ASA an organization that is welcoming of all people as a means to simultaneously improve the quality of the organization and advance scientific research." It is an honor to present Andrea the 2025 R. Bruce Lindsay Award.

JOSEPH A. TURNER

Helmholtz-Rayleigh Interdisciplinary Silver Medal
in
Underwater Acoustics, Signal Processing in Acoustics, and Acoustical Oceanography

ASA
ACOUSTICAL SOCIETY
OF AMERICA

N. Ross Chapman

2025

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997	James E. Barger	2011
David E. Weston	1998	Timothy J. Leighton	2013
Jens P. Blauert	1999	Mark F. Hamilton	2014
Lawrence A. Crum	2000	Henry Cox	2015
William M. Hartmann	2001	Armen Sarvazyan	2016
Arthur B. Baggeroer	2002	Blake S. Wilson	2017
David Lubman	2004	Kenneth S. Suslick	2018
Gilles A. Daigle	2005	Barbara G. Shinn-Cunningham	2019
Mathias Fink	2006	Michael R. Moldover	2021
Edwin L. Carstensen	2007	George L. Augspurger	2022
James V. Candy	2008	Vera A. Khokhlova	2023
Ronald A. Roy	2010	D. Keith Wilson	2024

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993

ENCOMIUM FOR N. ROSS CHAPMAN

…for establishing the field of geoacoustic inversion

21 MAY 2025 • NEW ORLEANS, LOUISIANA

Ross Chapman always loved physics. He pursued his passion first at McMaster University where he received his BSc in Physics in 1968 and then at the University of British Columbia where he received a PhD, also in Physics, in 1975. He served as a Research Scientist for the Defence Research Establishment Pacific, Canada from 1976 to 1995 after which he joined the School of Earth and Ocean Sciences, at the University of Victoria, where he served as a Professor with distinction until 2011 and is currently Emeritus Professor

Ross has made extraordinary contributions to science throughout his career through a substantial body of innovative, cross-disciplinary, and thoughtful research and experimental work in areas including underwater acoustics, signal processing, and acoustical oceanography. His numerous publications and citations, advising of research scientists and students alike, and leadership positions in administration related to scientific discoveries are a testament to those contributions.

When Matched Field Processing (MFP) was developed and adopted in the 1980s and early 1990s as an extremely useful acoustical signal processing tool for source localization in the ocean, Ross recognized its potential for addressing critical problems in acoustical oceanography. He realized ahead of many working in the field that, in addition to providing information about the location of a transmitting source, MFP can be used for geoacoustic inversion as well as characterization of the water column.

Specifically, Ross was one of the first scientists who used global optimization with Monte Carlo methods to solve the problem of Matched Field Inversion (MFI) in multidimensional spaces with a plethora of environmental unknowns, making a seminal contribution to Acoustical Signal Processing as it relates to Underwater Acoustics and Acoustical Oceanography. This work was critical for the advancement of inversion endeavors, leading to numerous related methodologies for seabed property extraction. His research formed the basis for recent critical developments in which he still plays a role, such as transdimensional inversion (with Stan Dosso) and warping for modal identification and Bayesian inversion (with Julien Bonnel).

Machine learning has recently gained traction in acoustical signal processing as it applies to ocean acoustics. In addition to introducing Monte Carlo methods to MFP and MFI, Ross was among the first scientists who employed machine learning towards solving the inverse problem in ocean acoustics, work that was presented in papers in the 90s. The recognition these publications are currently gaining is a testament to their importance.

Ross promoted networking by organizing workshops in ocean acoustics, participating in conferences, and engaging in conversations with researchers from all over the world. He has organized several workshops and special sessions on MFP/MFI in collaboration with distinguished colleagues.

Ross participated in many experiments, several times as a Chief Scientist. His participation in the Shallow Water 2006 experiment was remarkable, with many insights gained from the data and work by multiple researchers. Ross was a co-author in a seminal paper in 2020 on benchmarking geoacoustic inversion methods applied to this important data set.

In addition to his contributions to geoacoustic inversion combining ocean acoustics and signal processing, Ross has made significant strides in the characterization of gas hydrate deposits in the seabed (a significant source of energy), yet another critical issue in the study of the oceans. He was a pioneer in methane hydrate research (a crucial topic related to climate change) and his work culminated in two papers in *Nature*.

Ross has been a mentor to many students and young professionals, who have become distinguished scientists as evidenced by the positions they currently hold and the recognition that they have received. He has mentored numerous graduate students and postdoctoral scholars and has worked with many undergraduate students, generously dedicating his time, and instilling in them his passion for acoustics. His broad and far-reaching efforts demonstrate his deep dedication to educating budding scientists at all levels. Ross is a true educator who propagates scientific discovery in the best possible way. What stands out about Ross is his eagerness to share his experimental data sets and encourage others, especially junior colleagues, to use them; throughout, he offers valuable advice and guidance. Ross does not just make available his data when asked; he offers them broadly and unconditionally.

In 2024, Ross was awarded the prestigious Munk Medal of the Oceanography Society for his seminal contributions to the field. Receiving the award in Ottawa in May 2024, Ross gave a lecture summarizing in an enlightening way the past, present, and future of geoacoustic inversion, the field that he has established. The talk went in depth into the why and how but was also entertaining and engaging in a way that only Ross can achieve.

Ross's travels have taken him to all corners of the globe, attending scientific conferences and visiting institutions, where colleagues from around the world have appreciated his presence, humor, and friendship. Today, Ross enjoys life on his beloved Gabriola Island, devoting time to his family, particularly his three young grandchildren.

This encomium is but a brief description of Ross's service in the fields of Acoustical Oceanography, Underwater Acoustics, and Acoustical Signal Processing and science at large. The Society is proud to honor Ross Chapman for his invaluable accomplishments in acoustics with the Helmholtz-Rayleigh Interdisciplinary Silver Medal in Underwater Acoustics, Signal Processing in Acoustics, and Acoustical Oceanography.

ZOI-HELENI MICHALOPOULOU

Gold Medal

ACOUSTICAL SOCIETY OF AMERICA

Arthur N. Popper
2025

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954
Floyd A. Firestone	1955
Harvey Fletcher	1957
Edward C. Wente	1959
Georg von Békésy	1961
R. Bruce Lindsay	1963
Hallowell Davis	1965
Vern O. Knudsen	1967
Frederick V. Hunt	1969
Warren P. Mason	1971
Philip M. Morse	1973
Leo L. Beranek	1975
Raymond W. B. Stephens	1977
Richard H. Bolt	1979
Harry F. Olson	1981
Isadore Rudnick	1982
Martin Greenspan	1983
Robert T. Beyer	1984
Laurence Batchelder	1985
James L. Flanagan	1986
Cyril M. Harris	1987
Arthur H. Benade	1988
Richard K. Cook	1988
Lothar W. Cremer	1989
Eugen J. Skudrzyk	1990
Manfred R. Schroeder	1991
Ira J. Hirsh	1992
David T. Blackstock	1993
David M. Green	1994
Kenneth N. Stevens	1995
Ira Dyer	1996
K. Uno Ingard	1997
Floyd Dunn	1998
Henning E. von Gierke	1999
Murray Strasberg	2000
Herman Medwin	2001
Robert E. Apfel	2002
Tony F. W. Embleton	2002
Richard H. Lyon	2003
Chester M. McKinney	2004
Allan D. Pierce	2005
James E. West	2006
Katherine S. Harris	2007
Patricia K. Kuhl	2008
Thomas D. Rossing	2009
Jiri Tichy	2010
Eric E. Ungar	2011
William A. Kuperman	2012
Lawrence A. Crum	2013
Brian C. J. Moore	2014
Gerhard M. Sessler	2015
Whitlow W. L. Au	2016
William M. Hartmann	2017
William A. Yost	2018
William J. Cavanaugh	2019
Judy R. Dubno	2020
James F. Lynch	2021
Michael J. Buckingham	2022
Mark F. Hamilton	2023
Ingo R. Titze	2024

ENCOMIUM FOR ARTHUR N. POPPER

…for contributions to understanding fish hearing and the effects of anthropogenic noise on aquatic life and for leadership and service to the Society.

21 MAY 2025 • NEW ORLEANS, LOUISIANA

The Acoustical Society of America (ASA) is privileged to honor Dr. Arthur "Art" N. Popper with the Gold Medal for his distinguished scientific achievements and his unwavering service to the Society.

Over the past six decades, Art has made significant and sustained contributions to our understanding of fish hearing and the effects of anthropogenic (human-generated) sound on aquatic life. His pioneering work has fundamentally shaped the field of animal bioacoustics, earning him global recognition as one of its foremost authorities on fish hearing and bioacoustics.

Art's academic journey began with an early fascination with fish auditory systems, which sparked a distinguished career. Art made enduring contributions to the field during nine years each at the University of Hawai'i and Georgetown University and 38 years at the University of Maryland, College Park, where he served as professor and at various times as chair and associate dean. His prolific scientific endeavors have resulted in over 300 peer-reviewed publications, each exploring diverse aspects of hearing and significantly advancing our understanding of fish auditory systems. His seminal discoveries include a groundbreaking paper in *Science* that provided the first detailed ultrastructural examination of the auditory regions in the inner ear of a fish species, offering critical insights into how hair cell morphology serves as the basis for sound source localization in fishes and sharks. Equally impactful is his work published in *Nature* on clupeid fishes (e.g., herrings, sardines, and shad), which revealed that certain species of shad can detect and respond to ultrasound to well over 100 kHz, an extraordinary ability believed to be an evolutionary adaptation for evading echolocating predators such as dolphins.

Art's investigations into the morphology and function of fish auditory systems have enhanced our understanding of the evolution of vertebrate hearing. His research on a wide range of species has provided insights into the structural and functional diversity of hearing capabilities across taxa.

Beyond academia, Art has addressed pressing environmental challenges through his pioneering studies on the impact of anthropogenic sounds, such as pile driving and seismic air guns, on aquatic life. These efforts have not only informed international guidelines but have also heightened global awareness of the ecological impact of underwater noise pollution.

In addition to his scientific contributions, Art has played a pivotal role in advancing auditory research through his editorial leadership. Notably, Art has served as associate editor for several journals including *The Journal of the Acoustical Society of America* (JASA), and as editor of *Acoustics Today* (2014–2024).

One of Art's major contributions has been as editor of 87 books on various aspects of hearing. Most notably his editing of the *Springer Handbook of Auditory Research* (SHAR) (1989–2024), a series of books he co-founded in 1992 with his close friend and collaborator Richard R. Fay. The series launched with the landmark volumes *The Mammalian Auditory Pathway: Neuroanatomy* (volume 1) and *The Mammalian Auditory Pathway: Neurophysiology* (volume 2). Over the decades, the series has grown to encompass 77 volumes, with millions of downloads worldwide. This extraordinary reach has established SHAR as a cornerstone resource for auditory research, widely utilized by scientists and practitioners alike.

Additionally, Art serves as senior editor for the influential Springer books on the *Effects of Noise on Aquatic Life*. These volumes have not only been groundbreaking in their content but also immensely popular, with the first two volumes amassing over 1.1-million chapter downloads and views (to date!). Art's editorial efforts have significantly shaped the dissemination of auditory and bioacoustics knowledge, ensuring that cutting-edge research is accessible to a global audience.

Art's influence is not limited to the printed word. He has trained and mentored a generation of scientists who continue to advance bioacoustics research. Over the course of his career, he has mentored and inspired more than 40 Ph.D. students and postdoctoral fellows, many of whom have made contributions to the field of animal bioacoustics. Through his guidance, these individuals have not only advanced bioacoustics research but have also become leaders in academia, industry, and beyond. Art's mentorship has fostered a thriving community of scholars whose collective work continues to redefine the field. His commitment to nurturing the intellectual growth of his mentees goes beyond teaching technical expertise; he instills a passion for discovery, a drive for innovation, and a deep sense of responsibility for scientific rigor. Through his support and encouragement, Art has created a legacy that transcends his own achievements, embodied in the successes of those he has guided.

Beyond his scientific accomplishments, Art has also been a steadfast leader within ASA. A valued member of the Society for over 50 years, Art was elected as a Fellow of ASA in 1995 in recognition of his exceptional "contributions to understanding hearing in aquatic animals." Since 2014, he has served as editor of *Acoustics Today*, shaping the publication into a platform for disseminating knowledge across diverse areas of acoustics and to a broader lay audience. In addition, from 2016 to 2024, Art served as the coordinating editor for the Animal Bioacoustics section of *JASA*, a position he held alongside his role as associate editor from 2013 to 2024. His editorial leadership has been instrumental in maintaining the journal's reputation for academic rigor and relevance, particularly in the field of animal bioacoustics.

Art's service to ASA extends far beyond his editorial roles. He has been an active member of the Society's Animal Bioacoustics Technical Committee, serving three terms (1991–1998, 2006-2009 and 2012–2024), during which he provided invaluable guidance on advancing the study of animal bioacoustics. Through his leadership, dedication, and vision, Art has significantly enriched ASA, ensuring its continued prominence as a leading professional society in acoustics. His sustained contributions have left an indelible mark on the organization and the field at large.

Despite his achievements, Art is renowned for his humility, humor, and kindness. Colleagues speak warmly of his ability to make even the most complex scientific concepts accessible and engaging. His infectious enthusiasm for discovery has inspired countless collaborators and enriched the scientific community.

In celebrating Dr. Arthur N. Popper, the ASA not only honors a remarkable scientist but also a cherished mentor, leader, and friend. His contributions to our understanding of underwater bioacoustics have shaped the field for generations, ensuring his enduring legacy.

JOSEPH SISNEROS

Session 4aAAa

Architectural Acoustics, Signal Processing in Acoustics and Computational Acoustics: Data-Driven Room Acoustics I

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

Xenofon Karakonstantis, Cochair
Electrical Engineering, Technical University of Denmark, Ørsteds Plads, Building 352, Kgs. Lyngby 2800, Denmark

Chair's Introduction—7:00

Contributed Papers

7:05

4aAAa1. Preferred values for concert hall acoustic parameters for different musical styles based on subjective preference. Fernando M. del Solar Dorrego (The Pennsylvania State Univ., 445 Waupelani Dr., Apt B10, State College, PA 16801, fsolar@gmail.com) and Michelle C. Vigeant (The Pennsylvania State Univ., University Park, PA)

In the study of concert hall acoustics, acoustical parameters have been proposed for the objective description of the acoustics of a performance venue. The purpose of the present study was to obtain preferred values for concert hall acoustic parameters for different musical styles. A subjective study was undertaken in which subjects had to rate a set of stimuli in terms of overall preference using a paired comparison rating technique. The stimuli were obtained through the convolution of seven spatial room impulse responses (SRIRs) from seven different halls, with three anechoic musical excerpts, which are representative of the Baroque, Classical, and Romantic musical styles, respectively. Once the preference ratings were obtained, a rank ordering of the seven concert halls was determined for each musical style. The values of the acoustical parameters of the most preferred halls are proposed as recommended values for these measures. Preferred values are presented both for the parameters described in Annex A of the ISO 3382-1 standard, and for two new metrics proposed by the authors, which can predict perceived reverberance and musical clarity with increased accuracy from the octave-band values of early decay time (EDT) and clarity index (C80).

7:25

4aAAa2. Acoustical design and historical landmark: Places for the performing arts. Jose A. Nepomuceno (Design, Acústica & Sônica, Rua Girassol, 139, Cjt 12, Sao Paulo, Sao Paulo 05433-000, Brazil, nepo.acustica@gmail.com)

Renovating historic theaters and converting historic spaces into places dedicated to the performing arts present unique challenges. On the one hand, preserving the building's identity is necessary—an imprecise concept. On the other hand, there are requirements to update the infrastructure, including comfort, safety standards, acoustics, stage, and technology. It is worth considering that a building's identity also includes its acoustical characteristics. Preserving these is particularly important in cases of exemplary acoustic signatures. However, one aspect that often drives the renovation of historic theaters is the solution to acoustic issues. In other cases, changes in the architecture driven by the need for restoration can alter the acoustic response. This article presents different cases of interventions in historic spaces: an opera house 100 years old with specific actions to meet architectural updates; a recital hall to correct the acoustic response between 63 and 250 Hz; the lounge of a railway station converted into a concert hall; and finally, the development of a recital hall on the site of a theater that burned down, of which only the Foyer remained. Acoustical parameters for the halls are presented.

Invited Paper

7:45

4aAAa3. Acoustic equity in performance spaces: Measurement results at wheelchair accessible seating positions. Steph Ahrens (DLR Group, 6457 Frances St., Ste. 200, Omaha, NE 68106, sahrens@dlrgroup.com), David Manley (DLR Group, Omaha, NE), and Kenton Hummel (Idaho National Lab., Omaha, NE)

This presentation is a preliminary review of research on acoustic equity in performance spaces. The focus of the research study is to evaluate measured acoustic parameters in wheelchair spaces and general patron seats with the goal to identify differences, and their relations to the architectural design. The goal is not to identify acoustic deficiencies but rather to identify trends in the built environment for equitable acoustic experiences in this new area of research. In this presentation, we will give a preview of the results while focusing on the "why" and "how" of the study. We will discuss methodologies, challenges during data collection and analysis, interpreting the ISO 3382 measurement results, and how the study evolved once the data was analyzed. This presentation will close with a discussion of future areas of research in the intersection of equity and room acoustics.

Contributed Papers

8:05

4aAAa4. Refurbishment of a choir rehearsal room. Stephen Dance (School of the Built Environment and Architecture, London South Bank Univ., Borough Rd., Torquay Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk) and Vincent Tham Jee Sheng (School of the Built Environment and Architecture, London South Bank Univ., London, United Kingdom)

A choir rehearsal room has been recently refurbished due to the original room having insufficient reverberation according to the newly introduced ISO23591:2021. Feedback from choir singers was obtained to study the effect of room acoustics on their perception with regard to singing effort and ability to hear oneself and others in the refurbished room. Field measurements taken to ISO3382-1:2009 were compared to the original configuration of the choral rehearsal room. These included room acoustics parameters: T_{mf}, Bass Ratio, Clarity, Sound Strength, and Critical Distance. The choir was surveyed as to their preference to critique the suitability of ISO23591:2021, including a comparison to NS8178:2014. In light of the answers, a calibrated computer simulation was produced in CATT-Acoustics to predict the necessary changes to the newly refurbished room to improve the acoustic.

8:25

4aAAa5. Acoustics measurement and analysis methodology for archaeoacoustics studies and auralization. Luna Valentin (Music, Stanford, 660 Lomita Ct, Stanford, CA 94305, luna@ccrma.stanford.edu), Peter Svensson, Sara Martin (Dept. of Electron. Systems, NTNU, Trondheim, Norway), Jonathan Berger, Jonathan S. Abel (Music, Stanford, Stanford, CA), and Romain Michon (Team Emeraude, INRIA, Lyon, France)

We describe a framework and methodology for characterizing the acoustics of archaeological sites. Chauvet Cave, a protected space inaccessible to the public in France, serves as a case study where acoustic parameters are derived from impulse responses obtained by data analysis of *in situ* measurements. Chauvet's exquisite paintings of animals and other artifacts provide a unique glimpse into aspects of Upper Palaeolithic life. Absent from these clues are hints of sonic practices of Chauvet's visitors—evidence that could provide hypotheses of how and why the caves were used. Acoustic parameters such as reverberation time, strength, sound clarity, spectral balance, echo density, and envelopment are extracted from the measured impulse responses and integrated into GIS-based spatial maps of the cave. This integration enables statistical comparisons with archaeological data, facilitating hypothesis testing regarding site usage and cultural practices. Our approach adheres to strict conservation protocols and room acoustic measurement practices, ensuring the collected data is suitable for room acoustics analysis and auralization applications. Quality control processes and assessments of measurement reliability are key. Establishing a standard data-driven protocol for impulse response measurement to afford reproducible and reliable archaeoacoustic research can provide novel insights into the role of sound in ancient civilizations, highlighting the complementary role of acoustics in enhancing traditional archaeological studies.

8:45

4aAAa6. Scaled models application for the measurements and adaptation planning in small room acoustic. Bartlomiej Chojnacki (Mech. and Vibroacoustics, AGH Univ. of Krakow, Mickiewicza 30, Cracow 30-059, Poland, bchojnacki@agh.edu.pl), Paulina Pik, and Maria Brzoska (Mech. and Vibroacoustics, AGH Univ. of Krakow, Cracow, Poland)

The continuous advancement of acoustic scale modeling methods clears the way for new applications of reduced-scale models. Small room acoustics is an area where scale models have yet to be fully explored, presenting the potential for studying acoustic treatment methods and testing innovative sound-absorbing and diffusing panels *in situ*. This presentation highlights recent developments in scaled modeling at AGH University of Krakow, showcasing a new measurement setup designed for small-room acoustic testing at a 1:8 scale. To validate the proposed methods, measurements from the scaled model will be compared with those from a full-size reference room. Additionally, a novel type of electroacoustic source will be introduced, focusing on aspects such as frequency response requirements and sound source directivity. Modifications to analysis methods—such as decay curve analysis, waterfall plots, and STFT for room resonance analysis—will also be discussed. Experimental results will be presented, including studies on first reflection point treatments, reverberation adjustments, room mode reduction with bass traps, and sound field diffuseness using scaled models.

9:05–9:25 Break

9:25

4aAAa7. Investigation on sound reverberation characteristics of a coupled volume system. Mengyi Yang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, yangm11@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

The characteristics of sound reverberation are fundamental and significant aspects of room acoustics. This work investigates spatial and temporal reverberation characteristics in a coupled volume system. In coupled volume systems, reverberant energy often manifests as a multiple-slope decay, which is relevant to both physical and perceptual behaviors. This feature also brings a localizable reverberance from the secondary volume, invoking interest in spatial analysis of the reverberation. The intention is to leverage clarity and reverberance, being both competing yet desirable perceptual attributes. Using experimentally measured binaural impulse responses, the advanced Bayesian method and the interaural decorrelation analysis facilitate a systematic analysis of both temporal and spatial characteristics of sound reverberation in the coupled volume system. It will also guide the investigations into a subjective evaluation. Driven by experimental data measured in the coupled volume system this talk discusses these analysis results.

Invited Paper

9:45

4aAAa8. Effectiveness of the ensemble averaging technique in measurements for sound absorption characteristics of materials. Toru Otsuru (Sci. and Technol., Oita Univ., 700 Dannoharu, Oita 870-1192, Japan, otsuru@oita-u.ac.jp), Reiji Tomiku, Noriko Okamoto (Sci. and Technol., Oita Univ., Oita, Japan), and Yoshitaka Imaoka (Graduate School of Oita Univ., Oita, Japan)

The ensemble averaging is one of the most important concepts for scientific measurements, especially for sound absorption measurements of materials. Applying a technique based on ensemble averaging to the sound field where a measurement is undertaken, the authors have proposed a novel measurement method for sound absorption characteristics, surface normal impedance and corresponding absorption coefficient, of materials. We named it EA method and we presented the method's excellent robustness and good reproducibility. Herein, a theoretical background of the method is given, briefly. Then, the outline of an EA-method measurement conducted in a reverberation room using pseudo-random incident noises is described and measured values of both ensemble averaged surface normal impedance and corresponding absorption coefficient are shown to examine the effectiveness of the averaging. In the EA-method

measurement, two types of sensors are utilized: pu-sensor and two-microphone. Finally, the applicability of the EA method to *in situ* measurements at various field environments is exhibited and the importance of on-site calibration is confirmed.

Contributed Papers

10:05

4aAAa9. Solid-angle based reverberation time estimation for rooms with substantial "early absorption". Cameron Hough (Marshall Day Acoust., 10/50 Gipps St., Collingwood, Victoria 3067, Australia, chough@marshallday.com) and Peter Exton (Marshall Day Acoust., Melbourne, Victoria, Australia)

Parametric design tools allow for rapid design evaluation of geometric options for an auditorium early in the design process. A key design parameter is the required room volume to obtain a design reverberation time. While initial volumetric requirements can be obtained using existing methods (e.g., Sabine equation), the accuracy is reduced for "early absorptive" rooms where a significant proportion of the incident sound goes onto absorptive surfaces either directly or after a strong early reflection, e.g., auditoria with steep seating rakes and "directed reflection sequence" rooms with strong early reflections. In these rooms, the solid angle over which sound is incident on absorptive surfaces is large, and the volumetric requirement is often underestimated. Often, it is only when the room is modeled computationally that a shortfall in the room volume is identified. A more reliable method is needed, using geometrical parameters such as solid angles that can be obtained from parametric tools. A new formula for estimating reverberation time considering this "early absorptive" solid angle is proposed. The results are compared to traditional statistical formulae and Odeon simulations. The new formula provides an improvement to existing formulae for "early absorptive" rooms. Potential modifications to incorporate source directivity and scattering are discussed.

10:25

4aAAa10. Evaluating a scale model reverberation room for sound absorption coefficient measurements: A cost-effective alternative. Jacques Martell-Villalpando, Alejandro Martinez-Borquez (Tecnologico de Monterrey, Monterrey, Mexico), and David I. Ibarra-Zarate (Tecnologico de Monterrey, Eugenio Garza sada, Monterrey, Nuevo Leon 64700, Mexico, david.ibarra@tec.mx)

The sound absorption coefficient (SAC) is a key parameter in acoustic simulation and treatment of enclosures. Among SAC measurement methods, reverberation rooms are a preferred method for their versatility, but require significant volumes, specimen sizes, and investment. This study assessed the capability of a 1:6 scale model reverberation room (SMRR), based on the full-scale reverberation room (FSRR) at the Universidad Nacional Autónoma de México (UNAM), to perform SAC measurements and reduce dimensions and costs. A hybrid methodology based on the impulse response (IR) method was developed in compliance with ISO 354, ASTM C423, and SAE J2883 standards. IRs were measured within the FSRR and the SMRR using maximum-length sequence (MLS) and logarithmic sine sweep (LSS) signals with two different test materials. SACs extracted from both IRs were compared using non-parametric tests on matching measurement pairs in both enclosures. No statistically significant differences were found for MLS comparisons, and the correlation coefficient indicated a strong monotonic association between both datasets. These findings suggest the potential of SMRRs for SAC measurements as a cost-effective alternative. However, further refinement of the scale model and studies on IR acquisition methods are needed to enhance its reliability.

10:45

4aAAa11. Referenced measurements of diffuser reflections using an acoustic goniometer. Colin M. Gaines (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 1520 6th Ave., Apt. 312-D, Troy, NY 12180, cmgbazinga@gmail.com), Ziqi Chen, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This research embarks on high-resolution measurements for diffuser reflection coefficients in room acoustic and noise control applications. This work implements the practical goniometer measurements with a 16-channel microphone carriage. With the diffuser surrounded by the acoustical goniometer, scattered reflections are captured at varying angles through a correlation measuring technique emitted from a speaker. A windowing method is employed to separate incident and reflected components. The impulse responses facilitate the measurement of scattered reflectance at 144 angles, offering higher angular resolution. This work focuses on reference measurement in order to achieve scattered reflectance with desired accuracy. By adopting the goniometer-based approach, architectural acousticians can obtain a higher level of detailed acoustic analysis on diffusing devices, addressing a critical gap in current measurement practices pertaining to scattered reflectance, specifically for oblique incidences. This paper discusses measurement results on refined references and demonstrates high-resolution measurement practice.

Session 4aAAb

Architectural Acoustics and Structural Acoustics and Vibration: The Intersection of the Acoustic and Structural Domains in Sound Transmission in Buildings

Evelyn Way, Cochair
Maxxon Corp, 920 Hamel Road, Hamel, MN 55340

Pablo Daroux, Cochair

Jorge Patricio, Cochair
Buildings, LNEC, Av. do Brasil, 101, 1700-066 Lisbon, Portugal

Invited Papers

7:00

4aAAb1. Structural wood floor system design and the possible relations to accoustic properties. Jeffrey Olson (Boise Cascade, P.O. Box 2400, White City, OR 97503, jeffolson@bc.com)

Wood construction is the most common type of floor framing system currently used in structures built in the United States. Stiffness of the lumber or engineered wood joists typically controls the floor system's structural design, due to minimum deflection building code requirements or more restrictive limits specified by the project's design professional of record. Structural stiffness also dictates the floor joist's fundamental natural frequency which quantifies the floor system's performance, that is the occupant's perception of how the floor feels. The corresponding paper presentation discusses both light-frame wood and mass timber floor stiffness design and its possible relation to the floor system's acoustic properties.

7:20

4aAAb2. A comparison of sound and impact transmission through structures with different natural frequencies. Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com) and Jeffrey Olson (Civil & Environ. Eng., Washington State Univ., White City, OR)

Transmission loss through double panel systems is typically understood to be correlated with the mass of the two panels and the distance between them. This does not take into account the rigidity of the structure supporting the panels or other characteristics of a floor-ceiling system that may affect sound transmission. While it has been anecdotally observed that longer-span structures have lower transmission loss than similar floor-ceiling assemblies with shorter spans in buildings, disentangling other important variables such as construction quality or flanking through adjacent building elements is difficult. While lab tests remove many of those variables, previous studies have been limited by the size of the test openings to only compare small changes in span length. An experiment was designed to simulate span length by varying the natural frequency of the structural system. From this study, we hope to show whether and how much the stiffness of the structure contributes to the sound and impact transmission loss of a floor-ceiling assembly.

7:40

4aAAb3. Low-frequency structural vibration transmission—Finite element predictions compared to building measurements. Jeffrey A. Zapfe (Acentech, 33 Moulton St., Cambridge, MA 02138-1118, jzapfe@acentech.com) and Ethan R. Brush (Acentech, Cambridge, MA)

Above-grade fitness centers can be an annoying source of vibration disturbance in buildings. Their potential impact is best evaluated at the design stage before people begin to use its treadmills, free weights, and exercise classes. Often the building does not yet exist and structural models provide the best way to evaluate potential impacts. The authors recently had the opportunity to compare finite element frequency response function (FRF) predictions to test data obtained in the field. The driving point accelerance predictions showed very good agreement with the test data up to 100 Hz. Cross-accelerance predictions from one point to another similarly showed good agreement with the test data. The test data also confirmed reciprocity when the drive and response points were interchanged. Test data in the field were obtained using both an instrumented force hammer and a drop weight with an accelerometer. Both methods produced consistent experimental FRFs. This study shows that a well-conceived finite element model can produce reasonable predictions of low-frequency structural response.

8:00

4aAAb4. Portuguese admissibility criteria for re-radiated noise from underground metro lines. Jorge Patricio (Buildings, LNEC, Av. do Brasil, 101, 1700-066 Lisbon, Portugal, Jpatricio@lnec.pt) and Sónia Antunes (Buildings, LNEC, Lisbon, Portugal)

The improvement of the quality of housing buildings, mainly in terms of their sound insulation, has led to consideration of other aspects than those currently taken into account, such as the effects of low-frequency noise. In urban areas, the effects of vibrations due to rail traffic have been increasing significantly throughout the years. Building occupants can effectively perceive vibrations as mechanical oscillations (for frequencies between 1 and 80 Hz) or indirectly as re-radiated noise (in the frequency range of 16–250 Hz), causing physical and psychological disturbances. When rail traffic circulates in tunnels, as happens with metro lines, the re-radiated noise can be particularly noticeable in houses with good sound insulation. Currently, there is no consensus on how to assess the human response to indoor vibrations since the methodology used in several countries is rather different, as can be seen from the national standards or published legal requirements. This paper presents the admissibility criteria proposed in Portugal for the assessment of re-radiated noise, its applicability and drawbacks. These aspects are discussed based on vibration measurements inside buildings (in this case, floors) and simulations associated with metro lines in tunnels.

Contributed Paper

8:20

4aAAb5. Analysis of partial venting in continuous and point isolated concrete floating floors. Wilson Byrick (Pliteq Inc., 131 Royal Group Cresent, Woodbridge, ON L4H 1X9, Canada, wbyrick@pliteq.com)

Concrete floating floors are frequently used to improve airborne and impact sound insulation in buildings. If trapped air between the structural slab and topping slab is not adequately vented, the natural frequency of the spring-mass system is significantly increased due to air stiffness. Unger (1975) showed how stiffness, transmissibility and limiting radii could all be theoretically established in vented and unvented floors. In this paper, we measure laboratory and field installations with variable venting conditions and compare empirical data to the previously established theory. The vented area is changed, and natural frequency response is measured in an effort to establish some trends in partially vented floors. These effects are observed for continuous elastic layer floors and compared to discrete isolation systems.

Invited Papers

8:40

4aAAb6. Low-frequency sound and vibration trasmission from heavy weight drops in adjacent rooms: Case studies on fitness floors. Molly Smallcomb (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, molly.smallcomb@gmail.com), Nicolaus T. Dulworth, Chris Springthorpe, Hans Michel, Robin Glosemeyer Petrone, and Scott Pfeiffer (Threshold Acoust., Chicago, IL)

Gyms with free weights at floors above grade are a major concern for noise and vibration transfer, especially where heavy weight drops are anticipated at upper floors. Isolated floor build-ups of weightlifting rooms are designed to attenuate impacts and prevent low-frequency sound and vibration from transferring elsewhere in the building. In this study, *in situ* fitness floor build-ups were reviewed in steel structure buildings using heavy weight drops (barbells, kettlebells, and shotputs). Vibration measurements were taken at the weight-dropping room and adjacent rooms, while sound levels were also captured in adjacent rooms. Sound and vibration observations were also reviewed at other spaces throughout the facilities to understand the perceptibility of noise and vibration at a variety of adjacencies. Results and processes from case studies will be discussed, exploring system performance, subjective observations, and the opportunities and limitations of evaluating these systems alongside end users.

9:00–9:20 Break

9:20

4aAAb7. Alternative vibroacoustic methods for measuring airborne sound insulation. David W. Dong (Paul S. Veneklasen Res. Foundation, 11623 Talaud St., Cypress, CA 90630, info@veneklasenresearchfoundation.org), John LoVerde, Sunit Girdhar (Paul S. Veneklasen Res. Foundation, Cypress, CA), and Benjamin M. Shafer (PABCO Gypsum, Tacoma, WA)

The uncertainty in airborne sound insulation testing is high, as is well documented by interlaboratory studies, despite decades of effort. Some of this uncertainty may be inherent in the reverberation room test method, and improved evaluation of the airborne sound insulation of assemblies may be attainable using vibration measurements. The authors have begun an investigation of using vibroacoustic test methods to evaluate airborne sound insulation of wall assemblies. The goal is to develop a measurement method with an improved measurement uncertainty that may complement existing test methods. Additional measurements and analysis are presented.

9:40

4aAAb8. Advancing acoustic performance in mass timber construction: A novel approach to Kij testing. Alfredo Rodrigues (CDM Stravitec, 100 Sunrise Ave., 202, Toronto, ON M4A1B3, Canada, a.rodrigues@cdm-stravitec.com), Matias C. Camus, James Bligh, and Marina Rodrigues (CDM Stravitec, Overijse, Belgium)

Mass timber buildings offer a sustainable alternative to traditional materials due to their low carbon footprint, reusability, and high stiffness/mass ratio. However, this last feature can lead to poor acoustic insulation, mainly through flanking sound transmission. To mitigate the issue, specific joint detailing with acoustical decoupling is required. Though well-studied and with international standards developed to account for flanking sound transmission, estimating the vibrational reduction across junctions remains challenging due to the significant variability in building elements, structural connections, and acoustical insulation solutions. CDM Stravitec and Buildwise are collaborating to develop a novel Kij test setup and carry out a comprehensive test campaign of vibrational reduction across CLT junctions. Buildwise facility, combined with the latest test setup, adds to the existing flexibility in testing by allowing the simulation of additional loading of up to five floors on any floor–wall junction. This results in a highly flexible testing platform that can test Kij for most types of CLT building wall-to-floor junctions, CLT or composite slabs over glulam beams, among other test types. The main objective of this experimental campaign is to attempt to address several open questions relative to Kij across wall–slab junctions, including testing methods and sensitivity of Kij to key parameters on the junction and sound insulation solution.

10:00

4aAAb9. Impact sound annoyance—Further evaluation of timber-framed, concrete, and cross-laminated timber floors. Sabrina Skoda (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A0R6, Canada, sabrina.skoda@nrc-cnrc.gc.ca), Markus Mueller-Trapet, Jeffrey Mahn, and Iara Batista da Cunha (National Res. Council Canada, Ottawa, ON, Canada)

Impact noise from neighbors, such as footfall noise or the impact of dropped objects on floors can evoke significant feelings of annoyance in people living in multi-unit residential buildings. The sound insulation performance of the direct and flanking paths between adjacent dwellings is therefore significant for the quality of living of residents. To support the inclusion of requirements for impact noise limits in a future edition of the National Building Code of Canada, the National Research Council of Canada has been investigating the perceived annoyance from different types of impact sounds for various timber floor-ceiling assemblies. The results from listening experiments indicated that the correlation between subjective ratings and objective single-number metrics for impact noise varies depending on the type of impact source. To validate previous results and to include a wider variety of floor-ceiling assembly types, another listening experiment was conducted. Annoyance ratings for ball drops, hammer impacts, and footsteps on different timber-framed, precast concrete, and cross-laminated timber floor-ceiling assemblies with different toppings are compared.

10:20

4aAAb10. Analysis of floor impact noise reduction and impact force by floor mats. A-Hyeon Jo (Acoust. Environment Ctr., Korea Conformity Labs., 73, Yangcheong 3-gil, Ochang-eup, Cheongwon-gu, Cheongju-si, Chungcheongbuk-do 28115, Korea, joah@kcl.re.kr), Won-Hak Lee, and Jinyun Chung (Acoust. Environment Ctr., Korea Conformity Labs., Cheongju-si, Chungbuk, Korea)

This study is a basic research on improving floor finishing materials technology to reduce floor impact noise in apartments. The floor impact noise reduction performance and impact force of floor mats were measured on a general standard floor structure in Korea (concrete slab 210 mm + EPS 30 mm + finishing mortar 50 mm). Six types of PE mats, three types of EVA mats, and three types of PU mats were targeted. Lightweight and heavyweight impact noise were measured and evaluated according to KS F ISO 16283-2 and KS F ISO 717-2, and the floor impact noise reduction value of the mat was evaluated by the difference in the level of impact noise according to the installation of the mat. In addition, the impact force of the mats was measured according to JIS A 6519. The reduction performance of lightweight impact noise was high in the order of PU, PE, and EVA. PE mats showed a reduction performance of up to 4 dB in heavyweight impact noise. In addition, the impact force of the PE mats was the smallest, and the impact force decreased as the thickness of the mat increased.

10:40

4aAAb11. Improvement of floor impact sound insulation using small specimens. Won-Hak Lee (Acoust. Environment Ctr., Korea Conformity Labs., 73, Yangcheong 3-gil, Cheongju-si, Chungbuk 28115, Korea, whlee@kcl.re.kr), A-Hyeon Jo (Acoust. Environment Ctr., Korea Conformity Labs., Cheongju-si, Chungcheongbuk-do, Korea), and Jinyun Chung (Acoust. Environment Ctr., Korea Conformity Labs., Cheongju-si, Chungbuk, Korea)

In South Korea, where multi-family housing is prevalent, floor impact sound has become a significant social issue. According to statistics, more than 20 000 complaints related to floor impact sound are reported annually. In August 2022, amendments to the Housing Act in South Korea introduced stricter performance standards for floor impact sound insulation. Consequently, various floor structure developments are actively underway in the country. To improve the performance of floor impact sound insulation, it is common to conduct performance evaluations by applying various resilient materials and mortars either at construction sites or in laboratory settings. However, these methods have the disadvantage of being time-consuming and costly. This study investigates the potential of using small specimens to evaluate the reduction performance of floor impact sound. Specifically, specimens measuring 200 mm × 200 mm were used to analyze the impact force reduction, and their results were compared with those obtained from larger specimens measuring 3000 mm × 4000 mm. The comparison aimed to determine whether the performance of small specimens could be predictive of floor impact sound reduction. Furthermore, the study compares the results of large specimens with performance evaluations conducted on-site in laboratory settings to propose the scope and limitations of using small specimens for analyzing floor impact sound reduction.

Session 4aAB

Animal Bioacoustics: Acoustic Ecology and Biological Soundscapes

Xavier Mouy, Chair
Applied Ocean Physics & Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Road, Woods Hole, MA 02543

Contributed Papers

7:20

4aAB1. A mixed-design passive acoustic array to support Gulf-wide marine mammal science and restoration. Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0205, kfrasier@ucsd.edu), Len Thomas (Univ. of St. Andrews, St. Andrews, United Kingdom), Arturo Serrano (Universidad Veracruzana, Veracruz, Mexico), Adolfo Gracia (National Autonomous Univ. of Mexico, Mexico City, Mexico), Itzel Perez Carballo (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mexico City, Mexico), Heloise Frouin-Mouy (Univ. of Miami/NOAA, Key Biscayne, FL), Alba Solsona Berga (Scripps Inst. of Oceanogr., UCSD, San Diego, CA), Lance Garrison (NOAA, Miami, FL), John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Melissa Soldevilla (NOAA, Miami, FL)

Limited understanding of connectivity across the deep Gulf of Mexico has impeded efforts to effectively manage offshore marine mammal populations. In 2020, an international team established a distributed passive acoustic sensor array across the deep water Gulf, combining eight long-term moored stations with four annually relocated short-term stations selected using a space-filling design. This design provides a balance between continuous temporal monitoring at fixed sites and expanded spatial coverage through short-term deployments. Weekly density estimates for eight marine mammal species were analyzed to identify spatial and temporal patterns of occurrence and potential activity hotspots. To evaluate the utility of this mixed monitoring strategy, we applied an information-theoretic approach, quantifying the incremental variance in species density explained by each short-term deployment when integrated into a baseline model using only long-term stations. Preliminary results suggest incorporating short-term stations captures key phenomena—such as rare species occurrences, relationships with transient oceanographic features, and anthropogenic noise impacts—that may be missed by relying on a limited number of long-term stations, particularly in regions influenced by mesoscale processes. Long-term stations provide an essential temporal context for seasonal and interannual variability. This mixed monitoring strategy appears effective for pelagic species in large, dynamic ecosystems like the Gulf of Mexico.

7:40

4aAB2. Spatiotemporal distribution of dolphins within the Gulf of Maine. Peyton Steffek (Marine Biology, Scripps Inst. of Oceanogr., 166 Water St., Woods Hole, MA 02543, peyton.steffek@noaa.gov), Sophie R. Ferguson (NMFS, NOAA, Woods Hole, MA), Rose Nolan (Brandeis Univ., Waltham, MA), Xavier Mouy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), Julien Delarue (Halifax, JASCO Appl. Sci., Halifax, NS, Canada), Sofie Van Parijs, and Amanda Holdman (NMFS, NOAA, Woods Hole, MA)

Dolphin presence in the Gulf of Maine (GOM) has been previously classified as uniform across space and time, with little specification beyond this general distinction. With wind energy development planned for the GOM, a deeper understanding of the spatiotemporal distribution of dolphins is imperative. From 2020 to 2023, acoustic recorders (AMARs, Soundtraps, and F-PODs) were deployed in 30 sites throughout the GOM. This study analyzed SoundTrap data for dolphin whistles on an hourly time scale, using the PAMGuard whistle and moan detector along with custom python-based detection-viewing software, Soundscope, for validation. SoundTrap data detections were then compared to dolphin detections from other passive acoustic recorders in the region. Dolphins were detected at all recording locations with varying presence based on region and time of year. Dolphins were present year-round in offshore GOM, while seasonally present in inshore GOM. For the inshore GOM, presence varied seasonally by location. Along the Southern coastal shelf, dolphin presence peaked in spring and fall, while along the northern coastal shelf, dolphin presence peaked from fall to winter. These results expand on current knowledge of dolphin habitat use and can be used to guide wind energy plans for the lowest impact.

8:00

4aAB3. Understanding beaked whale summer distributions off the northeast United States through passive acoustics. Madison R. Medina (Univ. of California, Santa Cruz, 703 Almar Ave., Santa Cruz, CA 95060, imadison.medina@gmail.com) and Annamaria DeAngelis (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA)

Beaked whales are a cryptic family with limited behavioral and distributional data. Their long, deep-diving behavior limits their surface availability for observations. Passive acoustics have proven effective in detecting species-specific clicks during these deep dives, enabling continuous monitoring even under adverse conditions. In a 2021 (June–August) visual cetacean abundance survey off the Northeast region of the United States, passive acoustic data were collected using a towed hydrophone array. Active acoustic data were collected using shipboard echosounders, alternated systematically, to assess their impact on beaked whale passive acoustic detection. Species-specific frequency-modulated upsweep pulses from beaked whales were detected, classified, and localized using PAMGuard (v2.02.10), with individuals annotated by grouping click trains on consistent bearing changes. Data analysis using R (v4.3.2) and ArcGIS mapped distribution patterns, revealing three North Atlantic beaked whale species: Goose-beaked (*Ziphius cavirostris*), Sowerby's (*Mesoplodon bidens*), and True's (*Mesoplodon mirus*) beaked whales. Preliminary results indicate more detections occurred when the echosounder was off, consistent with past research. Species-specific distribution data in 2021 will be compared with the 2013 and 2016 abundance surveys. These results will improve our understanding of these cryptic species, contributing to the National Marine Fisheries Service's stock assessments, and supporting conservation and management.

8:20

4aAB4. Vertical acoustic stratification of tropical marine ecosystems.
Xavier Raick (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 136 East State St., Apt. 4F, Ithaca, NY 14850, xavier.raick@cornell.edu), Melanie Vendramme, and Eric Parmentier (Univ. of Liège, Liege, Belgium)

Vertical stratification shapes the structure and function of tropical marine ecosystems, yet its acoustic dimension remains poorly understood, especially in deeper zones. Extending our knowledge of reef dynamics across depth gradients is crucial not only to uncover the functioning of these largely unexplored ecosystems but also to advance non-invasive monitoring methods. This study provides the first insights into the soundscape of the lower rariphotic zone, revealing vertical acoustic stratification from altiphotic coral reefs, upper and lower mesophotic coral ecosystems (MCEs), to the rariphotic zone. Fish vocalizations decreased with depth overall but showed the reverse pattern at sunset, with sound activity rising from 20 m, 60 to 120 m, and peaking at 300 m. This sunset-driven surge was primarily driven by one dominant sound type, a phenomenon previously observed in environments such as MCEs and temperate seagrass meadows. This study represents the first exploration of vertical acoustic stratification, offering valuable insights into a region that remains largely unknown.

8:40

4aAB5. Modeling for acoustical corridors in patchy reef habitats of the Gulf of Tribugá, Colombia.
Maria Paula Rey-Baquero (Pontificia Universidad Javeriana, Bogotá, Colombia), Kerri D. Seger (Integral Consulting Inc., 2639B NW 56th St., Seattle, WA 98107, kerriseger@re-1.net), Camilo Andres Correa Ayram (Pontificia Universidad Javeriana, Bogota, Colombia), Natalia Botero Acosta (Fundacion Macuaticos Colombia, Santa Cruz, CA), and Maria Angela Echeverry-Galvis (Pontificia Universidad Javeriana, Bogota, Colombia)

Ecological corridors are a commonly implemented terrestrial management strategy used to connect ecological areas. They are meant to allow faunal movement, enabling greater habitat space, and species and genetic diversity. What if marine "acoustical corridors" could serve the same purpose in patchy reef habitats? The Gulf of the Tribugá on Colombia's Pacific Coast has a very patchy reef environment along its coastline. Fishes were recorded in these reef patches to identify common calls to the region. Then, acoustic propagation models from each known reef patch were produced to determine whether "acoustical corridors" did, in fact, connect them. Models were then created to consider whether a passing boat running perpendicular to the acoustical corridor could produce noise levels sufficiently high to fracture the corridor. The main product of this proof-of-concept exploration was maps of disturbed versus undisturbed acoustical connectivity of a typical fish call between reef patches. The same maps were made for singing humpback whales, as Tribugá is a breeding ground for Stock G. This talk will discuss the methods for making acoustical connectivity models that incorporate resistance levels as a way to consider how acoustical corridors may be a useful way to envision marine protected areas.

9:00–9:20 Break

9:20

4aAB6. Field demonstration of enhanced coral larvae settlement using acoustic enrichment, mesoscale artificial structures, and engineered biofilms.
Océane Boulais (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0206, La Jolla, CA 92093, oceane@ucsd.edu), Aaron M. Thode, Natalie Levy, Daniel Wangpraseurt (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Joshua Levy, and Joshua Madin (The Hawai'i Inst. of Marine Biology, The Univ. of Hawai'i at Mānoa, Kaneohe, HI)

Broadcasting healthy reef sounds as acoustic enrichment (AE) has been shown to stimulate settlement behavior among coral larvae in captivity. In July 2024, a field test in Kāne'ohe Bay, O'ahu, Hawai'i, evaluated AE during a natural coral spawning event. Nineteen Coral Settlement Modules (CSMs), mimicking reef roughness, were deployed at distances of 1–42 m from an underwater speaker on a 5-m-deep sandy seafloor. Additionally, synthetic biofilm-coated microhabitats, with Bacteria Reef Ink ("Brink"), were placed along a transect and around the AE setup. Healthy reef sounds were broadcast nightly for 3 weeks, centered on the new moon. Settlement was highest within 20 m of the speaker, peaking near it and declining to ambient levels beyond 20 m. A zero-inflated Poisson regression showed that Brink-coated microhabitats and speaker proximity significantly increased settlement densities. Distance effects were consistent across surfaces, with no significant relationship between algal cover and settlement. These findings highlight the potential of combining AE and synthetic biofilms to boost coral recruitment and accelerate reef restoration.

9:40

4aAB7. An illustration of the importance of baseline data: Blue whale behavioral response to a seismic survey.
Regina A. Guazzo (NIWC Pacific, 53560 Hull St., San Diego, CA 92152, regina.a.guazzo.civ@us.navy.mil), Dorene L. Stevenson, George J. Gagnon, Michael K. Edell (Marine Acoust., Inc., Arlington, VA), and Tyler Helble (NIWC Pacific, San Diego, CA)

Determining whether and how marine mammals respond to anthropogenic disturbance is challenging in part due to the need for adequate and appropriate baseline data. In this study, we tested the hypothesis that blue whale behavior changes during seismic surveys by analyzing vocalizing whale kinematic behavior during periods of seismic survey activity and comparing it to baseline behavior. A commercial seismic survey was monitored for about a month with US Navy arrays. Vocalizing blue whales in the area were tracked during the survey as well as during the same month for 3 years surrounding the survey. The blue whale tracks were closer to the area of the survey and were faster during baseline years. We used hidden Markov models to determine how swimming speed was related to the independent variables of hour, ship distance and heading, and survey phase. Blue whales were more likely to be in a fast state when they were closer to the seismic survey vessel during both the active and inactive phases. But when we analyzed tracks from the baseline periods, we also saw a higher probability of a whale being in a fast state as a function of distance to the "phantom" ship. Therefore, we could not conclude that the change in blue whale behavioral state with distance was a result of the seismic survey. Adequate and appropriate baseline information is needed to understand normal, undisturbed behavior, otherwise, incorrect conclusions could be drawn about the impact of disturbance. [Funding: SMART SEED, US Navy's LMR Program, OPNAVN974.]

10:00

4aAB8. The use of masking metrics to understand acoustic disturbance.
Rianna Burnham (Dept. of Fisheries and Oceans, Inst. of Ocean Sci., Sidney, BC V8L 5T5, Canada, rianna.burnham@dfo-mpo.gc.ca), Svein Vagle (Dept. of Fisheries and Oceans, Sidney, BC, Canada), and Maximilian Lauch (Dept. of Fisheries and Oceans, Victoria, BC, Canada)

We present the use of acoustic masking metrics, quantifying the reduced range of effective communication and echolocation signals used by marine mammals in various soundscapes and under differing underwater noise conditions resulting from anthropogenic activity. We show how this may be an effective and useful way to consider acoustic impacts, while taking a more species-centric approach in the calculations. This approach also allows comparison over time and in space, helping consider the impact on specific population groups or important life-history events, for example, if the noise emissions co-occur with species presence or certain behaviors. Using modeled vessel noise scenarios and *in situ* recordings we consider the potential disturbance of increased numbers and sizes of commercial ships on southern resident killer whales (SRKW, *Orcinus orca*) in coastal waters of British Columbia, as well as potential impacts of measures that could be put in place to mitigate this noise. Focus is given to SRKW's use of acoustics to forage. We will show how these metrics could be used as part of adaptive management processes, and their potential contribution when considering the use of noise thresholds or limits as a management action to prevent harm to at-risk species.

10:20

4aAB9. Underwater Internet of Things: Industrial boon, biological headache. Michael Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@ocr.org)

Technologies advancing the industrialization of the ocean are increasingly requiring autonomous operations, multi-nodal equipment interactions, and coordinated and accurate time and positioning data requirements within these interactions. Many of these interactions fall under the rubric of the "Underwater Internet of Things" (UIoT). While certain non-acoustic communication channels are available—wired and "Blue Light Laser," due to the constraints of the underwater environments, the most practical communication channels will be acoustic. Unfortunately, this feature was not lost through the evolution of underwater biological communication, so the preponderance of acoustical UIoT signals will be concentrated in frequency bands most heavily used by marine animals. In "dry" computer networks, transmission channels could be biologically isolated by wire and electromagnetic energy, minimizing biological concerns. As in the "dry" communication channels, these acoustical anthropogenic communication channels will require a useful architecture, assuring that communication signals are not ambiguated by noise—either from ambient industrial or natural sounds or by concurrent sound transmission in colliding frequency bands. Due to the importance of bioacoustic communication in underwater habitats, care must be taken to minimize biological exposure and interference.

10:40

4aAB10. Sperm whale and dolphin presence in the offshore Gulf of Maine. Rose Nolan (Brandeis Univ., 415, Waltham, MA 02453, arosenolan@brandeis.edu), Sophie Ferguson (Protected Species Div., National Oceanic and Atmospheric Administration, Azura Consulting LLC, under contract to the Northeast Fisheries Sci. Ctr., Woods Hole, MA), Amanda Holdman, Sofie Van Parijs (Protected Species Div., Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA), and Annabel Westell (Protected Species Div., National Oceanic and Atmospheric Administration, Azura Consulting LLC, under contract to the Northeast Fisheries Sci. Ctr., Woods Hole, MA)

Offshore wind energy development is planned for the Outer Continental Shelf (OCS) in the U.S. Gulf of Maine (GOM). The GOM is an important habitat for many marine animals including sperm whales and dolphins. In 2019 the Northeast Fisheries Science Center started deploying bottom-mounted passive acoustic recorders in the GOM. Here, the data collected between 2022 and 2023 from one site located in Georges Basin, west of the Northeast Channel, were analyzed for the hourly presence of sperm whales and dolphins using PAMGuard and SoundScope. Sperm whales were detected year-round but presence varied by season. From June 2023 to November 2023 sperm whales had a mean percent presence of 87%. In contrast, from December 2022 to May 2023 sperm whales had a mean percent presence of 8.6%. Dolphins were detected consistently year round and at all times of day. These results indicate this area in Georges Basin may be an important habitat for sperm whales and dolphins and provide a baseline that will be used to assess any future changes. This study supports the idea that sperm whales, classified as Endangered in US waters, and dolphins should be considered in mitigation plans and permitting efforts for offshore wind energy.

4a THU. AM

Session 4aBAa

Biomedical Acoustics: Bubbles and Ultrasound—Physiological Considerations I

Virginie Papadopoulou, Cochair
Biomedical Engineering, The University of North Carolina at Chapel Hill, 116 Manning Drive, 9004 Mary Ellen Jones Building, CB 7575, Chapel Hill, NC 27599-7575

Nicholas Ovenden, Cochair
Department of Mathematics, University College London, London WC1E 6BT, United Kingdom

Athanasios Athanassiadis, Cochair
Heidelberg University, Im Neuenheimer Feld 225, IMSEAM—AG Fischer, Heidelberg 69120, Germany

Chair's Introduction—7:55

Invited Paper

8:00

4aBAa1. Modulating the (sympathetic) nervous system with anesthetic-loaded nanodroplets. Harriet Lea-Banks (Physical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, harriet.lea-banks@sri.utoronto.ca), Neha Chauhan (Physical Sci., Sunnybrook Res. Inst., Toronto, ON, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Neurotransmitters are used by the sympathetic nervous system to signal danger and regulate the body. Exploiting these neural and hormonal responses may hold promise for new treatments for hypertension, anxiety, and chronic stress. We have developed a platform using transcranial focused ultrasound, bubble activity from vaporizing nanodroplets, and locally released neuroactive substances to perturb the sympathetic nervous system. By sonicating the periaqueductal grey region (540 kHz, 10 ms pulse length, 1 Hz PRF, 0.275–1.65 MPa), with and without systemically injected nanodroplets, local brain activity is altered, neurotransmitters and hormones are detected in the bloodstream, and systolic and diastolic blood pressure is reduced. By mapping the suppression and activation of neurons, and identifying the specific neuron subtype, we compare midbrain and frontal cortex stimulation. We compare healthy and hypertensive rodents and assess differences in cavitation activity across sex, brain region, and disease state. We investigate the influence of inhaled isoflurane level, which was found to mask the hypotensive effect of FUS during sedation compared to the conscious state. Understanding how transcranial focused ultrasound and cavitation activity, in combination with locally delivered anesthetics, affects the sympathetic nervous system, could have a significant impact in treating hypertension, anxiety, and chronic stress.

Contributed Papers

8:20

4aBAa2. Targeted contrast enhanced ultrasound imaging to detect molecular changes in the preeclamptic rat placenta. Lili Shi (Biomedical Eng., Tulane Univ., 1324 Tulane Ave., New Orleans, LA 70112, lshi6@tulane.edu), Allan Alencar (Biomedical Eng., Tulane Univ., New Orleans, LA), Kenneth Swan, Dylan Lawrence, Gabriella Pridjian (Tulane Univ., New Orleans, LA), and Carolyn Bayer (Biomedical Eng., Tulane Univ., New Orleans, LA)

Preeclampsia (PE), affecting 6%–8% of pregnancies worldwide, is characterized by hypertension and proteinuria in late pregnancy stages. The primary cause of PE is insufficient placental vascular remodeling, resulting in placental ischemia. This study analyzes targeted contrast-enhanced ultrasound (T-CEUS) images using a bicompartmental (BCM) model to quantify changes in molecular expression in placentas affected by preeclampsia. T-CEUS employs targeted contrast agents (TCA) to enhance ultrasound signals. To enhance the detection of molecular expression in PE placentas, we modified the surface of targeted TCA microbubbles to bind to $\alpha_v\beta_3$ integrin, a biomarker associated with angiogenesis. T-CEUS imaging was performed on 12 timed-pregnant Sprague-Dawley rats. On gestational day (GD) 14, 6 rats underwent reduced uterine perfusion pressure (RUPP) surgery to induce PE. On GD 18, CEUS image data was collected following the injection of TCA. The time-intensity curve (TIC) of each pixel was fit to the BCM model to obtain a parameter map of the binding constant (K_b) within the placenta. RUPP placentas had a lower K_b, compared to normal placentas, which is consistent with the results of two traditional semi-quantitative methods: differential target enhancement (dTE) and late enhancement (LE). Compared to dTE and LE, BCM images can distinguish more intrinsic anatomical structures.

8:40

4aBAa3. Histotripsy has an abscopal effect and induces an adaptive immune response in a neuroblastoma syngeneic model. Natalia L. Antonides-Jensen, Fernando Flores-Guzman (Surgery, Univ. of Chicago, Chicago, IL), Muskan Singh (Radiology, Univ. of Chicago, Chicago, IL), Jacky L. Gomez-Villa, Lydia Wu (Surgery, Univ. of Chicago, Chicago, IL), Daniela Olivera-Velarde, Erik Saucedo (Radiology, Univ. of Chicago, Chicago, IL), Timothy L. Hall (Univ. of Michigan, Ann Arbor, MI), Kenneth B. Bader (Univ. of Chicago, Chicago, IL), and Sonia L. Hernandez (Surgery, Univ. of Chicago, 5841 S. Maryland Ave., Ste. AB532, MC 4062, Chicago, IL 60637, soniah@uchicago.edu)

Background: Over 50% of high-risk neuroblastoma (NB) patients fail to respond to treatment. Histotripsy, a focused ultrasound therapy under development for nonthermal-tissue ablation via bubble activity, can ablate NB xenograft tumors and induce apoptosis. Reports suggest histotripsy may turn cold tumors to be immune responsive: we hypothesized that histotripsy can activate an immune response and an abscopal effect (distal tumor targeting). Methods: NB neuro-2a cells were bilaterally injected subcutaneously on each flank of immunocompetent mice. Histotripsy was applied to approximately 80% of one tumor with a custom system. Untreated controls received no treatment. Tumor volumes were calculated using the ellipsoidal formula from caliper measurements. After 6 days, single-cell suspensions were fixed and interrogated via flow cytometry using immune cell markers. $n = 4–6$ per group. Results: By day 5, untreated controls were twofold larger than histotripsy-treated tumors ($p = 0.03$) as well as their contralateral tumors. Flow cytometry revealed that immune cell markers such as CD3 + CD4 + T cells, F4/80 + macrophages, and CD11b + granulocytes were higher in both histotripsy and contralateral tumors compared to untreated controls ($p < 0.05$). Conclusions: Histotripsy delays tumor growth and has an abscopal effect on this NB model and activates an immune response.

9:00

4aBAa4. Assessment of an image-guided histotripsy system for renal cell carcinoma. Muskan Singh (Radiology, Univ. of Chicago, 5841 S. Maryland Ave., Chicago, IL 60637, Muskan.Singh@bsd.uchicago.edu), George R. Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), Vishwas Trivedi, Abhinav Kumar, Himanshu Shekhar (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), Adam D. Maxwell (Dept. of Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Renal cell carcinoma cancer (RCC) is the sixth most common cancer in the United States. Although surgery remains the primary treatment approach, the number of patients who qualify decreases annually due to an aging population with multiple comorbidities. An alternative to surgery is histotripsy, a non-thermal focused ultrasound technique that disrupts tissue through the generation of bubble clouds. In this study, an image-guided histotripsy system for targeting RCC was built and characterized. A focused transducer was developed with an elliptical geometry (12/9.6-cm major/minor dimensions), 8-cm focal length, and 1-MHz fundamental frequency. The transducer was capable of generating bubbles in scattering phantoms and *in vivo* porcine kidneys with pulses 10 cycles in duration for pulse peak negative pressures greater than 20 MPa. The transducer was equipped with a port for coaxial treatment monitoring with a curvilinear imaging probe. An image processing pipeline was developed that achieved a bubble-to-tissue ratio greater than 70 dB for ultrafast active imaging and passive detection of bubble-based emissions with a signal-to-interference ratio of 35 dB. Overall, this system has the capacity to serve as a tool for pre-clinical investigations to monitor the histotripsy treatment of RCC.

9:20–9:40 Break

Invited Paper

9:40

4aBAa5. Efficient computation of high-amplitude acoustics in the body: From ultrasound to noise. Spencer H. Bryngelson (Georgia Tech, 888 Juniper St. NE, Apt. 1516, Atlanta, GA 30309, shb@gatech.edu)

We present computational and experimental results for appreciating the biological effects of high-amplitude acoustics. This talk focuses on two problems with shared motivation. We represent the damage of noise to ear drums via vorticity production from broadband acoustic fields. Scale-resolved simulations identify the mechanisms that introduce harmful noise levels in purportedly well-fit ear plugs. Similarly, we present a strategy for optimal experimentation that identifies the material properties of soft, biological materials at high strains. A Bayesian strategy is buttressed by data assimilation and computer simulation. With these tools, we parameterize models for biological matter that otherwise require expensive experimentation via only a few simulations. Equipped with this information, we reliably distinguish between good and poor models and validate against simulation-unseen established experiments. A method for classifying the otherwise unknown biomaterial response is presented. Simulation-aided ablation treatment of biologically formed stones and malignant tissues via non-invasive biomedical ultrasound is also discussed.

Contributed Papers

10:00

4aBAa6. Vaporization dynamics of superheated phase-change contrast agents—Simulations versus experiments. Nicholas Ovenden (Dept. of Mathematics, Univ. College London, London WC1E 6BT, United Kingdom, n.ovenden@ucl.ac.uk), Laura Taylor (Dept. of Bioengineering, Imperial College London, London, United Kingdom), Qiang Wu (Dept. of Eng. Sciense, Univ. of Oxford, Oxford, United Kingdom), Luca Bau (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Meng-Xing Tang (Dept of Bioengineering, Imperial College London, London, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom)

Superheated perfluorocarbon droplets and other similar phase change contrast agents offer exciting new opportunities for ultrasound imaging and therapy. Their small size allows them to more easily perfuse the microvasculature of tumors than commonly used microbubble contrast agents; their typically longer circulation half-lives and greater surface-to-volume ratios are also highly advantageous for clinical use. These droplets can potentially be "activated" (vaporized) by an externally applied ultrasound pulse. Controlling the timing and location of droplet activation, while minimizing the risk of spontaneous vaporization, is vitally important to avoid potential safety concerns *in vivo*, such as vascular occlusion. The activating pulse, however, must fall within a range of pressures that can be safely used on patients. The aim of this study is to simulate the radial dynamics and acoustic signature of an activated nanodroplet and compare these to recent experimental temporal measurements of individual vaporizing nanodroplets. The accuracy of the ideal gas approximation for these droplets will be discussed, as well as the impact of

shell properties on the radial dynamics and what insight can be gained from the simulations in terms of control of the activation process.

10:20

4aBAa7. Analyzing size and shell stability of lipid-coated microbubbles over a fixed time span. Saikat Halder (Mech. and Aerosp. Eng., George Washington Univ., 800 22nd St. NW, Ste. 3000, Sci. and Eng. Hall, Washington, DC 20052, saikat.halder@gwmail.gwu.edu), Mehmet Yapar, and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC)

Microbubbles are gas-filled contrast increasing agents used in ultrasound imaging, typically ranging from 1 to 10 μm in size. They are encapsulated by a protective shell to improve stability and prevent coalescence. Ensuring a stable size distribution and reliable performance over time is becoming increasingly important for their effectiveness. In this study, we explore the long-term stability and acoustic performance of our homemade microbubbles. These microbubbles are created using a mechanical agitation method, with a gas core of perfluorobutane (C_4F_{10}) and a shell made from a lipid mixture of DPPC and DPPE-PEG-2000 in a 9:1 ratio. We measure the size distribution, scattering, and attenuation of these microbubbles at regular intervals over 4 weeks. Additionally, we analyze the bubble shell properties by using the size and attenuation data. This research provides insights into the lifespan and stability of polydisperse microbubbles over expanded periods, highlighting their potential as efficient ultrasound contrast agents.

10:40

4aBAa8. Mechanics of microbubble-induced cavitation in dense fibrin networks. Aarushi Bhargava (Biomedical Eng., Univ. of Wisconsin-Madison, 1550 Eng. Dr., Fitchburg, WI 53706, aarushi.bhargava@wisc.edu) and Gaurav Gardi (Physical Intelligence, Max Planck Inst. for Intelligent Systems, Stuttgart, Germany)

Chronic blood clots are highly retracted tissue-like structures with a dense and stiff fibrin fiber mesh. These fibers exhibit extraordinary extensibility and strain-stiffening behaviors that enable the mesh to withstand high forces, provide structural stability to the clot, and prevent drug penetration for clot lysis. We show acoustic stable cavitation informed by fibrin mechanics as a potential strategy to disrupt the fiber mesh and enhance drug delivery. Fibrin gels mimicking the stiffness and porosity of the fiber mesh in chronic clots were fabricated. Using high-speed imaging and fluorescence microscopy, we characterize the interaction of the microbubbles with fibrin fibers and their penetration ability into the gels. Using atomic force microscopy, we identify the microscale damage mechanism of fibrin fibers under forces equivalent to that induced by stable bubble cavitation. This information is used to optimize the cavitation conditions to maximize penetration ability. Cyclic forcing from the radial oscillation of bubbles is observed to induce damage in fibers that accumulate over time, leading to their rupture. We combine results from optical and atomic force microscopy to determine the relation of this damage with fibrin stiffness, porosity, and acoustic pulsing conditions.

THURSDAY MORNING, 22 MAY 2025 — BALCONY L, 7:55 A.M. TO 11:00 A.M.

Session 4aBAb

Biomedical Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Wave Propagation and Aberration in Complex Media: From Theory to Applications I

Vera A. Khokhlova, Cochair
University of Washington/Moscow State University, Physics Faculty, Moscow 119991, Russian Federation

T. Douglas Mast, Cochair
Biomedical Engineering, University of Cincinnati, 3938 Cardiovascular Research Center, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Chair's Introduction—7:55

Invited Papers

8:00

4aBAb1. Distributed aberration correction via differentiable beamforming. Jeremy Dahl (Radiology, Stanford Univ., 3155 Porter Dr., MC 5483, Palo Alto, CA 94304, jjdahl@stanford.edu), Louise L. Zhuang (Elec. Eng., Stanford Univ., Stanford, CA), Benjamin Frey (Appl. Phys., Stanford Univ., Stanford, CA), Dongwoon Hyun (Siemens Healthineers, Palo Alto, CA), and Walter A. Simson (NVIDIA, Palo Alto, CA)

Distributed aberration correction has long been a goal in adaptive imaging. While early models of distributed aberration and simulations of wave propagation could adequately capture the distortion imposed by a heterogeneous medium, the ability to correct for distributed aberration in ultrasound imaging remained challenging. In recent years, advances in sound speed estimation techniques coupled with synthetic aperture focusing have allowed for the development of a wide variety of distributed aberration correction techniques that

incorporate wave propagation models. In this talk, we describe distributed aberration correction using the differentiable beamforming model. This model iteratively beamforms signals from a synthetic aperture sequence using an *a priori* sound speed map, calculates a phase error loss from signals sharing a common midpoint, and backpropagates the loss gradient to update the sound speed map used for calculating the beamforming time delays. Iterations are continued until the phase error converges. Beamforming time delays depend on the wave propagation model used, which includes straight ray, bent ray path, or numerical implementations of the wave equation. We show examples of distributed aberration correction with the differentiable beamformer in both heterogeneous tissue-mimicking phantoms and *in vivo* tissue, demonstrating correction of both local and global image distortion.

8:20

4aBAb2. Aberration by soft tissues in therapeutic ultrasound: Current noninvasive correction methods and their challenges. Gilles P. Thomas (APL-CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, gthom@uw.edu), Pavel B. Rosnitskiy (Dept. of Medicine, Univ. of Washington, Seattle, WA), Vera A. Khokhlova (Dept. of Medicine, Univ. of Washington, Moscow), Oleg A. Sapozhnikov (Univ. of Washington/Moscow State Univ., Seattle, WA), and Vera A. Khokhlova (Univ. of Washington/Moscow State Univ., Moscow)

One of the challenges of transcutaneous high-intensity focused ultrasound (HIFU) therapies is the deterioration of focus due to aberration by inhomogeneous soft tissues. This effect is of particular importance in applications where the acoustic path is through inhomogenously distributed layers of fat such as kidney and breast tumor ablation. Deformation and respiratory motion of the soft tissues result in additional complexity of the problem. Using multi-element HIFU arrays, aberrations can be corrected by implementing time delays on each element of the array to compensate for differences in travel time caused by variations in the thickness and sound speed of the tissue layers in the beam path. Various solutions have been developed to determine these time delays. In particular, noninvasive methods relying on harmonic pulse-echo sensing with the HIFU array combined with CT- or 3-D ultrasound-based simulations have shown promise. This presentation will focus on these methods and their limitations, including the isoplanatic patch — the volume where the aberration correction made at one point remains applicable. A way of measuring the isoplanatic patch size of porcine body wall will be presented, and methods to reduce the number of corrections needed to treat a large volume will be discussed.

8:40

4aBAb3. Aberration correction for trans-skull histotripsy. Timothy L. Hall (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd, 1107 Gerstacker Bldg., Ann Arbor, MI 48109, hallt@umich.edu), Jonathan R. Sukovich (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Ning Lu (Stanford Univ., Ann Arbor, MI), Ellen Yeats, and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

The skull represents a substantial barrier to performing histotripsy therapy within the brain causing focal pressure loss and shifts in focal location. At sub MHz frequencies, both effects are mostly due to wavefront aberration from localized variations in skull thickness and sound speed. These aberrations can be corrected using a phased array by varying element transmit timings to realign wavefronts at the focus. This talk presents results using propagation modeling and acoustic emissions from cavitation (as well as a combination of both) for determining appropriate element timings for aberration correction. Propagation models tested included forward and backward ray-tracing methods and full-wave simulations (k-wave). Acoustic emissions associated with the expansion and collapse of bubble clouds as well as the re-excitation of previously generated bubbles were considered for time-reversal-based corrections. Propagation modeling was found to be very effective at correcting focal shift errors, but generally provided only marginal increases in focal pressure amplitude. Aberration correction using acoustic emissions was found to significantly increase focal pressure amplitude, particularly when combined with propagation modeling, achieving roughly 90% of the pressure obtained using a hydrophone. Full-wave modeling did not show significant improvement over ray-tracing methods perhaps because of errors in sound speed estimation from CT data.

9:00

4aBAb4. Aberration in complex media: The case of shear wave elastography. Stefan Catheline (INSERM U1032, 151 cours albert thomas, Lyon 69003, France, stefan.catheline@inserm.fr) and Bruno Giammarinaro (INSERM U1032, Lyon, France)

Shear wave elastography depends on the type of medical imaging used to track shear waves in soft biological tissues. This aberration correction study in elastography will nevertheless not cover the study of ultrasound, MRI, or optical imaging. It will rather investigate the impact of complexity medium on the shear wave propagation. Starting on the aberration correction techniques developed for ultrasound imaging, it will be shown how shear wave imaging, thanks to the local nature of its measurement, avoids the main pitfalls encountered by ultrasound. The singularity of various causes of aberration in elastography will then be presented.

9:20–9:40 Break

9:40

4aBAb5. Mapping local anisotropy for aberration correction in ultrasonic imaging for non-destructive evaluation applications. Katy Tant (James Watt School of Eng., Univ. of Glasgow, James Watt South Bldg., University of Glasgow, Glasgow, Scotland G128 QQ, United Kingdom, katy.tant@glasgow.ac.uk), James Ludlam (Mathematics and Statistics, Univ. of Strathclyde, Glasgow, United Kingdom), Andrew Curtis (Univ. of Edinburgh, Edinburgh, United Kingdom), and Victorita Dolean-Maini (TU Eindhoven, Eindhoven, Netherlands)

In ultrasonic imaging, to correctly focus scattered wave energy in the imaging domain, we require good knowledge of the underlying spatial distribution of material properties of the object of interest, as these can impact the speed and direction of the propagating waves. Travel-time tomography methods can be used to invert the fastest time of arrivals between pairs of transmitters and sensors to construct a map of some material property that varies in space. Given the usually high-dimensional and non-linear nature of these problems, much of the related literature has focused on driving these tomography approaches with Markov Chain Monte Carlo (MCMC) methods, which are of course computationally expensive. Variational Bayesian Inversion however offers an alternative, more efficient approach. These

methods pose the problem as a deterministic optimization one instead of a random sampling one, promoting faster convergence whist retaining probabilistic estimates and facilitating uncertainty quantification. In this work, we present the Stein Variational Gradient Descent (SVGD) as a means to invert for some parameterization of the spatially varying anisotropy in complex media. We subsequently demonstrate that the generated images can be used to refocus ultrasonic wave energy correctly in the imaging domain to produce more reliable defect characterization capabilities.

10:00

4aBAb6. Guided wave tomography of pipe bends for wall thinning detection using full waveform inversion. Madis Ratassepp (Dept. of Civil Eng. and Architecture, Tallinn Univ. of Technol., Ehitajate tee 5, Tallinn 19086, Estonia, madis.ratassepp@taltech.ee)

Pipe bends are critical areas prone to wall thinning due to flow-accelerated corrosion. Traditional monitoring methods rely on localized ultrasonic measurements, which are slow and costly. Guided wave tomography (GWT) offers an efficient alternative by utilizing permanently installed transducers to monitor an entire bend. However, an accurate forward model is crucial for GWT, especially for complex geometries like pipe bends. In this study, we propose a GWT approach based on full waveform inversion (FWI) for high-resolution thickness reconstruction of steel pipe bends. To improve the integration of the forward model with the FWI algorithm, we employ Thomsen parameters in a two-dimensional (2-D) domain to simulate the anisotropic nature of wave propagation in the bend. Additionally, we introduce a helical path separation technique, based on geodesic equations, to compute the propagation distance in the pipe bend, preserving wavefront information while accounting for the cyclic nature of the geometry. This method enables accurate separation of wavefronts, including focusing and scattered wave effects. The results were validated experimentally on a pipe with a 220-mm diameter and 1.5-d bend radius, where an artificial defect was introduced. Our findings demonstrate that FWI, combined with the helical path separation, effectively reconstructs the thickness map of defects, particularly those located near the extrados, confirming the feasibility of this approach for large-area, real-time assessments of structural integrity.

10:20

4aBAb7. Matrix imaging as a tool for high-resolution monitoring of deep volcanic plumbing systems and fault areas. Elsa Giraudat, Rita Touma (Institut Langevin, CNRS, ESPCI, PSL Univ., Paris, France), Arnaud Burtin (Institut de Physique du Globe, Université Paris Cité, Paris, France), Mathias Fink (Institut Langevin, CNRS, ESPCI, PSL Univ., Paris, France), Jean-Christophe Komorowski (ISTERRE, Université Greboble Alpes, Paris, France), Michel Campillo (ISTERRE, Université Greboble Alpes, Grenoble, France), and Alexandre Aubry (Institut Langevin, CNRS, ESPCI, PSL Univ., 1 rue Jussieu, Paris 75005, France, alexandre.aubry@espci.fr)

Volcanic eruptions necessitate precise monitoring of magma pressure and inflation for improved forecasting. Imaging the structure of major fault zones is essential for our understanding of crustal deformations and their implications on seismic hazards. Yet imaging those systems is challenging due to complex heterogeneities that disrupt standard seismic migration techniques. Here we map the magmatic system of the La Soufrière volcano in Guadeloupe [1] and the subsurface structure of the San Jacinto Fault zone (SJFZ) in California [2] by analyzing seismic noise data from a geophone array under a matrix formalism. Seismic noise interferometry provides a reflection matrix containing the signature of echoes from deep heterogeneities. Using wave correlations resistant to disorder, matrix imaging successfully unscrambles wave distortions, revealing La Soufrière and SJFZ's internal structure down to 10 km with 100 m resolution. This method surpasses the diffraction limit imposed by the geophone array aperture, providing crucial data for modeling and high-resolution monitoring. We see matrix imaging as a revolutionary tool for understanding the dynamics at work in volcanic systems and fault areas as well as for forecasting eruptions and earthquakes. [1] E. Giraudat *et al*., Commun. Earth Environ. 5, 509 (2024). [2] R. Touma *et al*., Geophys. J. Int., 2021, 780–794 (2021).

Contributed Paper

10:40

4aBAb8. Reflection matrix approach for quantitative imaging of complex media. Flavien Bureau, Antton Goicoechea (Institut Langevin, CNRS, ESPCI, PSL Univ., Paris, France), Cécile Brütt (Safran Tech, Magny-les-Hameaux, France), William Lambert (SuperSonic Imagine, Paris, France), Mathias Fink (Langevin Inst., ESPCI Paris, Paris, France), Arnaud Derode, Claire Prada (Institut Langevin, CNRS, ESPCI, PSL Univ., Paris, France), and Alexandre Aubry (Institut Langevin, CNRS, ESPCI, PSL Univ., 1 rue Jussieu, Paris 75005, France, alexandre.aubry@espci.fr)

We present a physically intuitive matrix approach for quantitative imaging of complex media. While standard reflection imaging methods generally rely on confocal focusing operations, matrix imaging consists of decoupling the location of the incident and received focal spots [1]. Following this principle, a self-portrait of the focusing process can be obtained around each point of the medium. The Gouy phase shift exhibited by each focal spot can be leveraged to finely monitor the wave velocity distribution inside the medium [2]. A local multiple scattering rate can also be evaluated and its depth evolution can lead to a local measurement of the scattering mean free path, independently from absorption losses [3]. The approach is here demonstrated with ultrasound for a controllable phantom system before being applied *in vivo* to liver. The wave velocity and the scattering mean free path are quantitative markers for biomedical diagnosis but they are also important monitoring parameters for non-destructive testing and geophysical applications. This work thus opens important perspectives for quantitative imaging of heterogeneous media in all fields of acoustics. [1] W. Lambert *et al*., Phys. Rev. X 10, 021048 (2020). [2] F. Bureau *et al*., arXiv:2409.13901 (2024). [3] A. Goicoechea *et al*., Phys. Rev. Lett. 133, 176301 (2024).

Session 4aEA

Engineering Acoustics: General Topics in Engineering Acoustics

Joseph Vignola, Chair
Mechanical Engineering, The Catholic University of America, 620 Michigan Avenue, NE, Washington, DC 20640

Contributed Papers

7:40

4aEA1. Musical field visualization using XR technology. Wataru Teraoka (Waseda Univ., 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, terawata1129@fuji.waseda.jp), Atsuto Inoue (Waseda Univ., Shinjuku-ku, Tokyo, Japan), Yasuhiro Oikawa (Waseda Univ., Tokyo, Japan), Takahiro Satou, and Masahito Kobayashi (Tobishima Corp., Noda-shi, Chiba, Japan)

Sound field visualization is effective for intuitive evaluation of the field. To visualize the sound field, various methods and systems have been studied and proposed such as acoustic holography, optical methods, and sound intensity visualization. Especially, the musical field visualization is effective for evaluating musical performance, supporting hearing-impaired persons, and inspiring artistic expression. In this study, we use the sound field visualization system "OTOMIRU®." The system consists of a 16ch microphone array, measurement system on PC, and visualization system on XR (extended reality) devices. The measurement system analyses sound sources recorded from the microphone array using the MVDR (minimum variance distortionless response) beamforming method. The analysis results are sent to the visualization system by TCP (transmission control protocol). Finally, the results are visualized as colormap-like thermography and superimposed on the real world on XR devices. The system is mainly used for noise visualization, detection, and countermeasure of construction sites or indoor spaces. To visualize the musical field, we applicate the system to visualize not only sound pressure levels but also other acoustical parameters such as frequencies. To visualize these parameters, sound field information can be detailed and comprehensible. We show the efficiency of the visualization and its applicability to other situations.

8:00

4aEA2. Research and development of an open-source acoustic camera. Jesse Kaye (Mech. Eng., Univ. of Hartford, 81 Carola Dr., Watertown, CT 06795, jkaye@hartford.edu) and Nathan J. Grover (Mech. Eng., Univ. of Hartford, West Hartford, CT)

The development of an open-source acoustic camera is described and assembled using commercially available hardware and freely accessible software, providing the potential for a more financially feasible kit assembly project. All developed code, components, and hardware are intended to be documented on a GitHub page along with full build instructions. The benefits and drawbacks of several digital beamforming methods, PCB design for audio purposes, and coding performant software on low-performing machines are discussed. Preliminary results for a sample prototype will be presented. The performance, accuracy, and limitations of the design are compared to commercially available solutions.

8:20

4aEA3. Development of a prototype 3-dimensional (3-D) microphone array system for passive acoustic bird and bat detection. Hrafn S. Sigurdarson (Northeastern Univ., 120 Forsyth St., 129 Egan Res. Ctr., Boston, MA 02115, sigurarson.h@northeastern.edu), Arpita Ghosh, Max Radermacher (Northeastern Univ., Boston, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A prototype microphone array system is being developed for passive acoustic detection of birds and bats, with a focus on coastal environments. Drawing on experience with underwater acoustic arrays, this system adapts proven designs for in-air applications, enhancing the potential for ecological monitoring in diverse habitats. The array facilitates tracking of bird and bat movements and measuring vocal activity by employing beamforming techniques to isolate and localize sound sources with enhanced signal-to-noise ratios in noisy environments. The prototype array is designed to test both 1-D and 2-D configurations for beamforming and localization. Testing will address key factors such as power distribution, amplification, ADC synchronization, and dynamic range in both laboratory and outdoor environments. Beamforming techniques will enhance the signal-to-noise ratio, significantly improving the signal clarity and accuracy of sound localization under challenging conditions. The array's modular design provides flexibility, allowing for iterative improvements and potential scalability for future expansions. This initial system establishes a foundation for demonstrating the feasibility of in-air beamforming and sound localization. Insights gained from these investigations will guide the development of a larger, more robust array, advancing passive acoustic monitoring technologies for studying bird and bat behavior in diverse and demanding environments.

8:40

4aEA4. Investigating the generalization strength of sound event detection and localization models across reverberant polyphonic acoustic scenes. Adeoluwawale Adewusi (Dept. of Elec. Eng. and Comput. Sci., Technische Universität, Berlin, March straße 23, MAR 5.051, Berlin 10587, Germany, adewusi@ni.tu-berlin.de) and Klaus Obermayer (Dept. of Elec. Eng. and Comput. Sci., Technische Universität, Berlin, Berlin, Germany)

Room acoustics is an integral part of real auralization audio and environmental soundscapes. However, it is not desirable in the detection and localization of environmental sound events because it causes interaural decorrelation that potentially degrades the model's ability to detect a target source, if other co-occurring competing sound sources from other locations are present. Besides that, training specialized models for every combination of specific room acoustic conditions and number of sources is ideal but not practicable. Thus, the study provides an insight on how sound event

localization and detection (SELD) models can be optimally trained to achieve robustness across diverse reverberant polyphonic scenes. Hence, a deep network classifier was developed based on Convolution-Recurrent Neural Networks and Pre-trained audio teacher–student transformer model. This was followed by training several classifiers on simulated multichannel reverberant acoustic scene data using measured spatial room impulse responses from different reverberant room conditions. These models are thereafter cross-tested on data from unknown room conditions. Results showed a sensitivity and the generalization ability of SELD models across varying degrees of reverberant acoustic scenes. Keywords: room acoustics, generalization strength, sound event localization and detection

9:00–9:20 Break

9:20

4aEA5. Investigation of meta-stacks for thermoacoustic applications. Samarjith Biswas (Oklahoma State Univ., 201 General Academic Bldg., Stillwater, OK 74078, samarjith.biswas@okstate.edu), Rivelino Santana Juarez, and James M. Manimala (Oklahoma State Univ., Stillwater, OK)

Thermoacoustic devices could offer a sustainable approach to energy harvesting, noise mitigation, and environmental diagnostics but remain constrained by low power-to-volume ratios and narrow operational bandwidths. These limitations hinder their adoption in applications requiring compactness and broadband performance. To address these limitations, metamaterial-inspired stack (meta-stack) designs for thermoacoustic applications are investigated in this study. Utilizing a standing wave thermoacoustic scheme, various meta-stack configurations are simulated based on Rott's approximation and validated using experiments. Three designs are explored: the helical pore stack (HPS), the helical resonator stacks (HRS), and the pie-slice resonator stacks (PRS). The additively manufactured stacks were optimized for low frequencies ($<$500 Hz) and broadband response. HPS utilizes helical pore geometries to enhance broadband response. HRS achieves dual-frequency operation due to the presence of two convoluted acoustic paths, improving performance at low frequencies, while PRS features modular sections configurable for a wide frequency range. Experimental results correlate well with numerical predictions, demonstrating the potential to enhance thermoacoustic performance. Further, nonlinear and transient analyses are performed to better understand loss mechanisms. Integrating meta-stacks in thermoacoustic devices could provide a promising means to address their traditional limitations and deliver advancements for multifunctional applications.

9:40

4aEA6. Time-domain turbulence ingestion forcing. Michael T. Rose (The Penn State Univ., State College, PA 16802, mtr198@psu.edu), Margalit Goldschmidt (The Penn State Univ., State College, PA), and Michael L. Jonson (The Penn State Univ., University Park, PA)

Turbulence ingestion is a significant contributor to turbomachinery noise. While frequency-domain calculations are commonly used to compute the unsteady turbulence ingestion forces, some applications do not have inherently stationary inflows, motivating a time-domain approach. Frequency-domain methods can invoke a quasi-steady assumption to attempt a time-dependent solution. This work employs and extends Ishimaru's time-domain method to calculate turbulence ingestion forces from an analytical model of upwash correlation. The method is validated for stationary axisymmetric inflows, where the resulting prediction is equivalent to the frequency-domain model developed by Lysak. This work extends the literature to include guidelines on the limit of the quasi-steady assumption by calculating an explicitly time-domain solution and comparing to the quasi-steady frequency-domain prediction.

10:00

4aEA7. Evaluating the degradation of thermal interface materials in liquid immersion cooling systems using ultrasonic methods. Jacey Birkenmeyer (Purdue Univ., 500 Central Dr., West Lafayette, IN 47906, jbirkenm@purdue.edu), Bijay Chhetri, Shubhra Bansal, and Luz D. Sotelo (Purdue Univ., West Lafayette, IN)

Immersion cooling systems for data centers provide improved energy efficiency. However, the mechanisms of degradation of thermal interface materials (TIMs) over time and their interactions with coolant fluids have not been thoroughly studied, and there are no current nondestructive/noninvasive degradation monitoring methods. While high-frequency ultrasonics and scanning acoustic microscopy are ubiquitous in metrology and defect detection for electronic packaging, the ability to characterize with these modalities has been underutilized. This presentation will discuss the use of ultrasonic nondestructive evaluation to detect changes in the material properties of coolant fluids and TIMs. For this purpose, combinations of various commercially available coolants and TIMs were degraded and evaluated under simulated immersion cooling conditions (in accordance with JESD22-A103C test standard). Due to the limited quantities and various form factors of the TIMs and coolants, creative experimental solutions were implemented in the ultrasonic data collection process. The relative changes in sound wave speed and attenuation in coolant fluids and TIMs were used to quantify the changes in material properties, which reveal important information about the compatibility between coolants and TIMs. This demonstrates that ultrasonic nondestructive evaluation can be leveraged to periodically or continuously monitor degradation.

10:20

4aEA8. Analysis of the effect of retro-reflective tape during laser Doppler vibrometry measurements. Hubert S. Hall (Mech. Eng., Texas Christian Univ., 2840 W Bowie St., Fort Worth, TX 76109, H.HALL@tcu.edu) and William Cunningham (Mech. Eng., Texas Christian Univ., Fort Worth, TX)

Laser Doppler vibrometry (LDV) is a non-contact optical measurement technique used to determine the velocity of a vibrating surface. LDV is highly valued for its non-contact, non-invasive approach which captures target vibration without physically altering test article dynamics. Historically hardware limitations required the application of retro-reflective tape to measurement surfaces to support reflection levels for accurate measurements. However, improvements in photodetector technology have supported tape-free LDV measurements for more reflective surfaces in newer LDV measurement systems. Currently, in many situations, tape is still required for non-reflective surfaces. The study is an examination of the effectiveness of retro-reflective tape across different materials and surface roughness to improve measurement quality. Comparative measurements were conducted using the Texas Christian University (TCU) Polytec VibroGo LDV system. Frequency response functions (FRFs) were collected for test samples with both untreated surfaces and surfaces coated with retro-reflective tape. The reflected signal strength, measurement precision, and noise levels are compared across these configurations. For instances where both cases were successfully tested, the effect of the tape introducing unwanted surface dynamics or the effect of additional weight on the test article dynamics is explored.

10:40

4aEA9. Low-frequency noise reduction through open ventilation windows using resonator arrays. Sanjay Babu T (Mech. Eng., Indian Inst. of Technol. Madras, 736, Pampa Boys Hostel, Chennai, Tamil Nadu 600036, India, me23s016@smail.iitm.ac.in) and Chandramouli Padmanabhan (Mech. Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu, India)

Natural room ventilation provides fresh air circulation and thermal comfort but comes at the cost of unwanted noise in urban areas; the primary source is traffic, with noise levels being nearly 85 dB during the daytime. The dominant frequency range is usually from 200–800 Hz. The primary objective of this research is to develop a resonator array that can be retrofitted on a window to reduce sound levels transmitted through the window while keeping the open ventilation area to 85%–90% of the original. The initial target is to have three to four different resonator frequencies in the 400–600 Hz range. Suitable resonators are designed and numerically analyzed using MATLAB and COMSOL. The resonators are then fabricated for testing in an impedance tube to verify the results. Once the verification is completed, a resonator array tuned to the chosen multiple frequencies is made; different orientations (grazing and normal incidence) of the resonator array are tested in a small twin-room reverberation chamber using a 465 $\times$ 465 mm opening. The tests show an 8–12 dB drop in the intended frequency range when larger volume resonators are used.

Session 4aED

Education in Acoustics: General Topics in Education in Acoustics

Noah J. Parker, Chair
Acoustics, Penn State, 201 Applied Science Building, University Park, PA 16802

Contributed Papers

8:40

4aED1. The science of underwater sound as a tool for teaching STEM. Liesl A. Hotaling (Univ. of Rhode Island, South Ferry Rd., Narragansett, RI 02882, lieslhotaling@yahoo.com), Kathy Vigness-Raposa (INSPIRE Environ., Newport, RI), and Cristian Graupe (INSPIRE Environ., Jamestown, RI)

Underwater sound can be an engaging vehicle for students to explore and connect with the science of sound and other fundamental physics principles. Using recordings of underwater sounds for inquiry-based activities can inspire learners to develop a deeper understanding of complex acoustics topics. Furthermore, the impacts of underwater sound on marine life can provide an opportunity for students to critically evaluate human impacts on nature through the scientific method. The Discovery of Sound in the Sea (DOSITS) Project contributes to the understanding of underwater sound by synthesizing and making available the results of ocean acoustics research for a diverse audience of stakeholders. DOSITS recently launched an online professional development (PD) program focused on underwater acoustics for education professionals. The PD includes four online modules featuring curated resources from the DOSITS site. The modules include a scientific review of key underwater acoustics content related to the science of sound, people's use of sound underwater, and marine animal sound production and reception. Participants are required to complete all four modules and participate in three 60-min sessions to discuss the science content and how the content and resources could be integrated into their own educational programs to qualify for PD certification.

9:00

4aED2. Abstract withdrawn.

9:20

4aED3. Continued development of interactive online tools for undergraduate acoustics education. Noah J. Parker (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, np.acoustics@gmail.com) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

This presentation will explore the development of online interactive animations as teaching tools for undergraduate acoustics education. The development of a website with a growing collection of interactive animations is intended to provide educators with freely accessible resources that engage students and enhance learning outcomes. Based on observations from three undergraduate classes, several new tools and animations have been developed, including those for room reflections, vowel sounds created by varying filters, spherically propagating particle motion, live-editable frequency spectra, and source–filter combinations. This talk will demonstrate these tools and discuss how they can complement traditional teaching methods to foster increased student engagement and comprehension.

9:40

4aED4. Using animations to aid in acoustic teaching lectures. Matthew Luu (Penn State, 446 Bluecourse Dr., Apt. 907, State College, PA 16803, mbl5743@psu.edu)

In teaching acoustics, derivations can be difficult to convey to students. Oftentimes, a whiteboard is used to discuss the ideas of the math behind a certain derivation or equation. Using MANIM, an open-source Python library, animations can be made of derivations integrated with visual aids to help students understand parts of acoustic math. The library and creator have gained popularity over the years through the YouTube channel known as 3blue1brown which presents popular math topics such as convolution, neural networks, and linear algebra. Animations, specifically for introductory acoustics courses, have been developed using MANIM and will be presented in this work. These animations include long form videos of a mass-spring oscillator and finite difference derivations as well as short form videos of Gram–Schmidt orthogonalization, and bilinear transformations. This Python library can be downloaded from Anaconda Python distribution and is an excellent way to develop supplemental material to help students understand concepts within the acoustics field.

10:00–10:20 Break

10:20

4aED5. An audible speed of sound in air demonstration for outdoor settings. Nathan Wolek (Creative Arts, Stetson Univ., 421 N Woodland Blvd, Unit 8252, DeLand, FL 32723, nwolek@stetson.edu)

Many demonstrations of the speed of sound rely on specialized equipment and waveform visualizations in a laboratory setting. Is it possible to achieve similar results with minimal technical setup in an outdoor setting? Can it be done without graphs to make it more accessible for visually impaired students? Since late 2020, Young Sound Seekers has engaged blind and partially sighted youth in monthly outdoor educational excursions to national and state parks in Central Florida. Our lessons focus on natural soundscapes and field recording, while introducing acoustics and other scientific concepts whenever possible. The speed of sound was a topic that regularly came up in lessons, but we lacked a way to clearly demonstrate the phenomenon that fit the needs of our group. The goal was to design a demonstration using familiar equipment that the youth could help setup, and to make the speed of sound audible in real-time without visualizations. Our final design uses just one field recorder, two omnidirectional microphones, 150 feet of cabling, and stereo headphones. This presentation will summarize the design process, review feedback from the youth, and share generalized setup notes so that others can reproduce the demonstration.

10:40

4aED6. Creative ways to study for an acoustics qualifying exam. Philip G. Kaufinger (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78766-9767, pkaufinger@utexas.edu), Chirag A. Gokani, and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Studying for a Ph.D. qualifying exam can be an imposing task for graduate students. Many students prepare for their qualifying exam by reading acoustics textbooks, reviewing class notes, and solving practice problems. The speakers synthesized these conventional modes of study into creative activities when preparing for their own qualifying exams: PGK recorded video derivations [https://www.youtube.com/@AcousticsDerivations.] and CAG assembled a website [https://chiragokani.github.io/class/quals/]. In this talk, the speakers share how these creative activities helped them review their fundamentals, deepen their understanding, synthesize new ideas, and acquire a more holistic view of acoustics. Since having passed their exams, the speakers continue to refer to the resources they created. First- and second-year students in the Graduate Program in Acoustics at the University of Texas at Austin have begun using these resources for their own study as well. The speakers highlight the value of creativity in the learning process and hope to inspire other students to find creative ways of studying for their own exams. [PGK and CAG are supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

THURSDAY MORNING, 22 MAY 2025 BISSONET/CARONDELET, 11:10 A.M. TO 12:00 NOON

Session 4aID

Interdisciplinary: Plenary Lecture: Inclusive Speech Technology: Developing Automatic Speech Recognition for Everyone

Ann Bradlow, Chair
Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL

Chair's Introduction—11:10

Invited Paper

11:15

4aID1. Inclusive speech technology: Developing automatic speech recognition for everyone. Odette Scharenborg (Delft Univ. of Technol., Van Mourik Broekmanweg 6, Delft 2628 XE, Netherlands, o.e.scharenborg@tudelft.nl)

Automatic speech recognition (ASR) is increasingly used, e.g., in emergency response centers, domestic voice assistants, and search engines. Because of the paramount relevance spoken language plays in our lives, it is critical that ASR systems are able to deal with the variability in the way people speak (e.g., due to speaker differences, demographics, different speaking styles, and differently abled users). ASR systems promise to deliver an objective interpretation of human speech. Practice and recent evidence however suggest that the state-of-the-art ASRs struggle with the large variation in speech due to e.g., gender, age, speech impairment, race, and accents. The overarching goal of our research is to uncover bias in ASR systems to work toward proactive bias mitigation in ASR. In this talk, I will present systematic experiments aimed at quantifying, identifying the origin of, and mitigating the bias of state-of-the-art ASRs on speech from different, typically low-resource, groups of speakers, with a focus on bias against gender, age, regional accents, and non-native accents.

Session 4aMU

Musical Acoustics and Computational Acoustics: String Instruments I

Vasileios Chatziioannou, Cochair
Department of Music Acoustics, University of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria

Mark Rau, Cochair
Music and Theater Arts & Electrical Engineering and Computer Science, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Bldg W18-Room, Cambridge, MA 02139

Montserrat Pàmies-Vilà, Cochair
Department of Music Acoustics - Wiener Klangstil (IWK), University of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, mdw - Inst. 22, Vienna 1030, Austria

Invited Papers

8:20

4aMU1. Abstract withdrawn.

8:40

4aMU2. Convergence analysis and relaxation techniques for modal scalar auxiliary variable methods applied to nonlinear transverse string vibration. Riccardo Russo (Dept. of Industrial Eng., Univ. of Bologna, Via Terracini 28, Bologna 40131, Italy, riccardo.russo19@unibo.it), Michele Ducceschi, and Craig J. Webb (Dept. of Industrial Eng., Univ. of Bologna, Bologna, Italy)

Many models of string vibration are available in the literature, playing a fundamental role in musical acoustics and sound synthesis. In particular, models incorporating geometric nonlinearities due to large strains enable the reproduction of important perceptual features. For real-time synthesis, ensuring the stability and efficiency of the numerical algorithms is crucial. To this end, explicit schemes based on the scalar auxiliary variable (SAV) method have been proposed, offering the advantage of avoiding iterative solvers by introducing an auxiliary variable into the system. However, a recent study identified spurious artifacts arising from anomalous auxiliary variable behavior in the simulation of a specific nonlinear transverse string vibration model. This work revisits the same model. Discretization in space is performed using finite difference and modal methods, the latter benefitting from reduced numerical dispersion when the linear part is integrated in time using exact numerical techniques. The nonlinear part is integrated using SAV. A convergence analysis is conducted against a reference solution to check the accuracy properties of both approaches. Additionally, a novel technique adapted from collision modeling is proposed to constrain the auxiliary variable, significantly improving simulation performance at lower sample rates.

9:00

4aMU3. Shape optimization of a guitar soundboard: Numerical approach and experimental results. Pierfrancesco Cillo, Tharindu Nandalal (Inst. of Eng. and Computational Mech., Univ. of Stuttgart, Stuttgart, Germany), Pascal Ziegler (Inst. of Eng. and Computational Mech., Univ. of Stuttgart, Pfaffenwaldring 9, Stuttgart 70569, Germany, pascal.ziegler@itm.uni-stuttgart.de), and Peter Eberhard (Inst. of Eng. and Computational Mech., Univ. of Stuttgart, Stuttgart, Germany)

The unique tonal qualities of musical instruments are profoundly influenced by the natural variability in their material, a factor perceptible to musicians even in seemingly identical instruments. Luthiers address this challenge through geometric adjustments, guided by their experience. This work introduces a computational framework to account for material variability in classical guitar soundboards, providing a systematic approach to supplement traditional luthiery. We employ a geometrically parameterized finite element model to predict geometric modifications to achieve target modal properties. As a key enabler, Parametric Model Order Reduction (PMOR) is used to significantly reduce the computational cost of repeated model evaluations required during the optimization. Applying PMOR to parameterized models poses significant challenges, such as maintaining a consistent mesh topology and deriving affine representations for all applied finite elements. This is addressed through custom finite element formulations with local parametrization and global-to-local mapping. In this talk, the general computational framework will be briefly summarized. Numerical results of the shape optimization will be presented and compared in terms of agreement as well as efficiency. Furthermore, experimental measurements will be presented to (a) verify the validity of the method and (b) demonstrate the possibility to also use this framework for material identification.

9:20–9:40 Break

9:40

4aMU4. Individual violin sound identification using audio features and machine learning. Hugo Pauget Ballesteros (LAM, Institut d'Alembert, 5 Pl. Jussieu, Paris 75005, France, hugo.pauget@dalembert.upmc.fr), Philippe Lalitte (IReMus, Paris, France), and Claudia Fritz (LAM, Institut d'Alembert, Paris, France)

Few articles have addressed the issue of identifying individual instruments of the same type from their recordings. In this paper, we investigate violin sound identification using two datasets comprising different recordings from multiple violinists on multiple violins. We compare several long-term audio features and evaluate their performance in violin classification using classical machine-learning algorithms. Long-term MFCCs were shown to effectively distinguish individual violins beyond the variabilities induced by the players, enabling violin recognition from recordings. The influence of key parameters (including the number of violinists, recording time and musical excerpts) on the performance of the recognition was assessed. This provides guidelines to optimize future data collection in the context of capturing the sound signature of a violin.

10:00

4aMU5. How violin bow's moment of inertia affects bowing parameters in bouncing strokes. Víctor Salvador Castrillo (Institut Jean le Rond d'Alembert (UMR 7190), Sorbonne Université/CNRS, 4 Pl. Jussieu, d'Alembert, boîte 162, Paris 75005, France, victor.salvador_castrillo@sorbonne-universite.fr), Frédéric Ablitzer (Laboratoire d'Acoustique de l'Université du Mans (UMR CNRS 6613), Université du Mans, Le Mans, France), and Claudia Fritz (Institut Jean le Rond d'Alembert (UMR 7190), Sorbonne Université/CNRS, Paris, France)

The moment of inertia of a violin bow is a mechanical property to which violinists appear particularly sensitive, especially during bouncing bow strokes such as *sautillé* or *spiccato*. In this study, we explored how the bow's moment of inertia influences the performance of these bowing techniques. Sixteen violinists, including professionals, students, and skilled amateurs, participated in the experiment using a bow with an adjustable moment of inertia. They were invited to perform bouncing strokes on the same violin under varying experimental conditions, including string, dynamics, and bow inertia. The rhythm was kept constant across participants using a metronome. Bowing parameters were measured using optical motion capture and analyzed statistically with linear mixed models to evaluate the effects of the experimental conditions. The results indicate that increasing the moment of inertia significantly reduces the amplitude of rotational oscillations around two axes perpendicular to the bow's length. Additionally, the bouncing point shifts significantly toward the tip as the moment of inertia increases. These findings emphasize the role of the moment of inertia in the bow's dynamics during bouncing strokes and illustrate how bow design influences violin performance.

10:20

4aMU6. The Archtop Project: A digital archive exploring historical Archtop guitars. Thomas Nania (D'Addario & Co., 595 Smith St., Farmingdale, NY 11735, houseofluthiery@gmail.com)

Launched in 2022, the Archtop Project is a free, online resource for guitar makers, players, researchers, and aficionados alike. Each historical instrument featured in the archive was analyzed using CT scanning, modal analysis, and near-field acoustic radiation measurements combined with high-quality photography, instrument specifications, and audio recordings to experience these instruments as never before, from the inside out. This work was inspired by that of violin maker, Sam Zygmuntowicz, who created the Strad 3D project (Strad3D) with support from the Oberlin Acoustics Workshop, now called the Violin Acoustics Foundation. The Archtop Project is supported by The Archtop Foundation and D'Addario and Company. In addition to an overview of the Archtop Project, updates on guitar acoustics work will be presented including The Dreadnought Project, an analysis of over 150 pre-war CF Martin dreadnought guitars.

10:40

4aMU7. String materials for historical zithers. Henna Tahvanainen (Aalto Univ., Otakaari 5, Espoo, Uusimaa 02150, Finland, henna.tahvanainen@aalto.fi), Jolan Thomasset (Ecole National Superieure de l'Electronique et de ses Applications, Cergy, France), and Rauno Nieminen (Univ. of the Arts Helsinki, Ikaalinen, Finland)

In both physical and numerical restoration of historical string instruments, knowledge of the string materials and their acoustical properties is needed. In this research, we construct a test instrument with five different string materials for historical zithers: steel, copper, horse hair, gut, and nylon. By keeping equal tension and length for each string, we can study how different choices of string material affect the sound of the instrument in terms of partials, their decay times, and inharmonicity among other things. The results can be used for sound synthesis as well as determining what kind of string to use for copies of historical zithers.

Session 4aNS

Noise: General Topics in Noise: Community Noise Perception and Psychoacoustics

Andrew Christian, Cochair
Structural Acoustics Branch, NASA Langley Research Center, 2 N. Dryden St., M/S 463, Hampton, VA 23681

Randall Ali, Cochair
KU Leuven, Electrical Engineering, Leuven, Belgium

Contributed Papers

9:00

4aNS1. The right way or the fast way. Chris Hulik (Tetra Tech, Rochester, NY), Tricia Pellerin (Tetra Tech, 10 Post Office Square, Ste. 1100, Boston, MA 02109, tricia.pellerin@tetratech.com), and Kevin Fowler (Tetra Tech, Chicago, IL)

When permitting, project challenges present themselves in various forms. The most typical challenges are related to noise mitigation, public pushback, or engineering constraints. Sometimes the greatest challenge related to a project is the clients themselves. The client–consultant relationship should ideally be one of mutual respect, with both parties working on a back-and-forth to reach the end goal within a set budget and timeline. However, a client that is domineering or perhaps is just unfamiliar with the importance of considering acoustic impacts as part of permitting, can easily cost themselves millions of dollars and add years to their permitting timeline by doing things the "fast" way, instead of doing things the "correct" way. This paper will discuss several case studies of this unfortunate phenomenon and present solutions that have been successfully put into practice.

9:20

4aNS2. The assessment of multiple sound source masking effects on traffic noise perception via facial expression recognition. Xuejun Hu (Key Lab. of Cold Region Urban and Rural Human Settlement Environment Sci. and Technol., Harbin Inst. of Technol., 92 West Dazhi St., Harbin 150000, China, 523762533@qq.com) and Qi Meng (Key Lab. of Cold Region Urban and Rural Human Settlement Environment Sci. and Technol., Harbin Inst. of Technol., Harbin, China)

Sound masking is a common measure to improve the perception of traffic noise in urban public space, and the current indicators for the effect of sound masking mostly rely on questionnaire surveys with fixed indicators, which suffer from the problems of large sample requirements and inability to express dynamically. In this study, we conducted sound perception experiments using facial expression recognition (FER) technology and investigated the enhancement effect of the masking effect of multiple sound sources on the perception of traffic noise in urban public space. The results show that the sound of music, birdsong, running water, and speech, all have the effect of improving the negative impact of traffic noise, but the improvement effect of the music and birdsong is more significant. The efficacy of music sound and birdsong sound compounded with traffic noise will gradually increase with time change; while running water sound and speech sound compounded with traffic noise will decrease with time change.

9:40–10:00 Break

10:00

4aNS3. A longitudinal study on environmental sound exposure at home, auditory gating abilities, and cognitive development in 5- to 7-year-old children. Dick Botteldooren (Information Technol., Ghent Univ., Technologiepark 126, Gent 9052, Belgium, dick.botteldooren@ugent.be) and Nele De Poortere (Information Technol., Ghent Univ., Ghent, Belgium)

The influence of personal traits on response to environmental noise has been extensively studied in adults. However, in young children, auditory gating and the detection of sound in noise are still under development. As genetic factors underlying these traits may be expressed to a certain extent by the presence of environmental sound, assessing the relationship between sound at home and outcomes such as cognitive development becomes very complex. To shed light on this complex interaction, a longitudinal study was conducted in the Ghent area, Belgium involving 40 children in preschool and 1 year later after the first year of formal schooling. Cognitive abilities were assessed using the IDS-2 test and results were *z*-scored with reference to a normative age-matched sample. Auditory gating was assessed with a Mismatch Negativity test using verbal sounds. A sound in noise detection test was used to measure the child's ability to suppress background noise. Environmental sound at home was measured close to the child's bedroom window. It is found that the direct effects of environmental sound on cognitive development are limited to shared attention and inhibition tasks but that the effect on sound detection in noise is significant which in turn relates to better overall performance on the cognitive tasks. One of key differences with previous work that could explain these findings is the use of measured overall sound.

10:20

4aNS4. Abstract withdrawn.

10:40

4aNS5. Individual response trends in a psychoacoustic test comparing UAVs and road vehicles. Andrew Christian (Appl. Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov), Laura Thomas (Dept. of Mech. and Mater. Eng., Florida Int. Univ., Miami, FL), and Aaron B. Vaughn (Appl. Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

This presentation discusses a reanalysis of data from a psychoacoustic test in 2017 that compared annoyance responses due to the noise of uncrewed aerial vehicles (UAVs) and road vehicles. This time, instead of averaging the responses across subjects, the regression model is applied to a single subject's data at a time. The analyses show that the individual subjects' responses were not simply distributed randomly around the mean response—individuals had strategies for listening that were markedly different than the average would suppose and were based on responding to

different attributes of the sounds. Similarly, the model that is useful for fitting the averaged data is not found to be particularly useful to describe nearly any individual's response. This conclusion is compared with the results of other recent tests that have shown similar conceptual trends. These results are discussed in the context of the assumptions made by multilevel statistical analyses: What does it mean when one reads the typical comment about psychoacoustic results that "large individual variability was observed"?

THURSDAY MORNING, 22 MAY 2025 BALCONY J, 8:00 A.M. TO 11:00 A.M.

Session 4aPAa

Physical Acoustics and Computational Acoustics: Infrasound I

Philip S. Blom, Cochair
Earth & Environmental Sciences, Los Alamos National Laboratory, P.O. Box 1663, M/S F665, Los Alamos, NM 87545

David N. Green, Cochair
AWE Blacknest, AWE Blacknest, Brimpton, RG7 4RS, United Kingdom

Stephen Arrowsmith, Cochair
Southern Methodist University, 819 Lake Terrace Drive, Dallas, TX 75218

Invited Papers

8:00

4aPAa1. Statistical models for infrasonic propagation: Application to the detection capability of the IMS network. Alexis Le Pichon (CEA, DAM, DIF, Arpajon F-91297, France, alexis.le-pichon@cea.fr), Julien Vergoz, Constantino Listowski (CEA, DAM, DIF, Arpajon, France), and Patrick Hupe (Federal Inst. for Geosciences and Natural Resources (BGR), Hannover, Germany)

The detection capability of the International Monitoring System (IMS) deployed to monitor compliance with the Comprehensive Nuclear-Test Ban Treaty (CTBT) is highly variable in space and time. Previous studies estimated the source energy from remote observations using empirical yield-scaling relations. However, these relations simplified the complexities of infrasound propagation as the wind correction applied does not account for an accurate description of the middle atmosphere along the propagation path. In order to reduce the variance in the calculated transmission loss, massive frequency, and range-dependent full-wave propagation simulations are carried out, exploring a wide range of realistic atmospheric scenarios. A cost-effective approach is proposed to estimate the transmission losses at distances up to 4000 km along with uncertainties derived from multiple gravity wave realizations. Transmission loss statistics are combined with an explosive source model and noise statistics to quantify the 90% probability detection threshold of the IMS network. In the context of the future verification of the CTBT, this approach helps advance the development of network performance simulations in higher resolution at a global scale with limited computational resources.

8:20

4aPAa2. The 2024-Sep-18 Toropets explosions: Analysis of a series of seismo-acoustic events over ~15 hours. Alexandra Nippress (AWE Blacknest, AWE Blacknest, Brimpton, Reading RG7 4RS, United Kingdom, alex@blacknest.gov.uk) and David N. Green (AWE Blacknest, Brimpton, United Kingdom)

At ~00:56 UTC on 2024-Sep-18, the first seismo-acoustic event in a series over a period of ~15 h occurred at an ammunition depot in Toropets, Russia. Over 25 events have both seismic [recorded at FINES, Finland (638 km)] and infrasound [recorded at IS43, Russia (339 km)] arrivals associated, with the seismic station OBN (342 km) recording both seismic and air-to-ground coupled arrivals. Additionally, there are >40 events observed as infrasound-only arrivals. Where seismic arrivals exist, we can determine the origin time of the events and the celerities of the associated infrasound signals at IS43. The observed signals generally show two arrivals with typical stratospheric celerities (~307 and ~280 m/s for the first and second arrivals, respectively), followed by a later thermospheric arrival (~246 m/s). The two "stratospheric" arrivals show variation in amplitude and frequency content both relative to each other and through time. Seismic magnitudes also vary through time, indicating variable explosive yield and ground coupling. Atmospheric specifications predict a weak stratospheric duct that cannot explain the two stratospheric arrivals at IS43. Our analysis focuses on how to characterize complex event sequences using far-field infrasound. [UK Ministry of Defence © Crown Owned Copyright 2024/AWE.]

8:40

4aPAa3. An updated machine learning detector for infrasound array data. Jordan W. Bishop (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, jwbishop@lanl.gov), Philip S. Blom, and Jeremy Webster (Los Alamos National Lab., Los Alamos, NM)

Beamforming is a common processing method for infrasound data that typically involves estimating an optimal set of plane wave parameters and a coherence value for a signal of interest recorded on an infrasound array. A high coherence value across the spatially distributed array of microphones can be used to separate acoustic signals from pressure fluctuations due to wind. Recently, a machine learning detector (Bishop *et al*., 2022) was proposed to categorize infrasound signals based on the beamforming results, with noise, transient signals, persistent signals, and moving sources as categories. This categorization allows for a more complete characterization of signals of interest compared to the state-of-the-art adaptive F detector, which provides a binary detection result. Manual labeling of the training data is time-intensive, and the number of examples from realistic, observed, signal categories is largely unbalanced. Using a physical wind noise model (Raspett *et al*., 2008), a physical model for Doppler-shifted moving sources, and parametric models for persistent and transient signals, we generate a large synthetic training dataset. We retrain our machine learning model on the synthetic data, and we compare our updated tuned and untuned models with the original model predictions.

9:00

4aPAa4. From atmospheric infrasound to the crust and mantle: Demonstrating subsurface inversion using balloon data. Marouchka Froment (Solutions Dept., NORSAR, Gunnar Randers Vei 15, Kjeller N-2007, Norway, marouchka.froment@norsar.no), Quentin Brisaud (Solutions Dept., NORSAR, Kjeller, Norway), Sven Peter Näsholm (Dept. of Informatics, Univ. of Oslo, Oslo, Norway), Antoine Turquet, and Tina Kaschwich (Solutions Dept., NORSAR, Kjeller, Norway)

The atmosphere of the Earth allows for a coupling of seismic waves into acoustic waves, below the range of human hearing (infrasound). This process is expected to be sixty times stronger on our neighboring planet, Venus. Recent studies have shown that high-altitude balloon platforms offer an attractive solution for recording seismic infrasound in the harsh environment of Venus or in remote locations on Earth. This technology could thus be key to assessing seismic activity and subsurface properties in such areas. In this presentation, we present current advancements in balloon seismology, including recent detections of earthquake infrasound by stratospheric balloons on Earth. Balloon data have unique characteristics that must be accounted for in their analysis, such as specific infrasound noise characteristics, noise due to balloon buoyancy oscillations, as well as infrasound multipathing due to topography and other effects. Using a 2021 earthquake detected by multiple balloons on Earth, we demonstrate that balloon infrasound can successfully be used in a seismic inversion framework to jointly retrieve earthquake source location and seismic velocities of the subsurface. Results are in good agreement with those obtained with traditional seismic data. Based on these findings, we further explore seismo-acoustic coupling and infrasound propagation on Venus and the implications for prospective balloon seismology missions.

9:20–9:40 Break

9:40

4aPAa5. Statistical protocols for analyzing wind noise levels for site selection. Roger M. Waxler (Univ. of Mississippi, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu) and Claus Hetzer (Univ. of Mississippi, Tempe, AZ)

Noise generated at an infrasound sensor by local turbulent pressure fluctuations in the atmosphere, generally referred to as wind noise, is arguably the greatest impediment to infrasound signal detection. In recent years there has been extensive research on the underlying mechanisms for wind noise generation. While considerable progress has been, and continues to be, made there is still no comprehensive method for estimating, from some first principles, wind noise levels at a given site. At present, the best method for determining wind noise levels at potential deployment sites is to place test sensors at the sites and collect data for an extended period of time. Here we present a protocol we have developed for the analysis of such data. The method entails identifying the frequency band relevant to the proposed application and then estimating the appropriate band limited short time rms pressure levels at the test sensors. This generates a time history of rms pressure levels. We then develop a statistical model for the expected band limited noise levels which can be compared to an appropriate threshold model for the determination of site acceptability.

10:00

4aPAa6. Laser-based primary calibration for microbarometers. Chad M. Smith (The Penn State Univ., Appl. Res. Lab., State College, PA 16804, chad.smith@psu.edu), Thomas B. Gabrielson (The Penn State Univ., State College, PA), and B. J. Merchant (Sandia National Labs., Albuquerque, NM)

In previous work, the National Center for Physical Acoustics at the University of Mississippi designed and built a large chamber for secondary (referenced) calibration of microbarometers. Sandia National Laboratories (SNL) then refined and implemented this chamber at their Facility for Acceptance, Calibration, and Testing (FACT) site. These large chambers incorporate moving coil loudspeakers capable of operating in receive and transmit modes, and in more recent work, Penn State University (PSU) and SNL used these loudspeakers to implement a reciprocity-based primary (non-referenced) calibration technique. Unfortunately, while this technique has been shown to have suitable uncertainty above 0.05 Hz, uncertainty increases rapidly below this frequency. Due to this, the authors are investigating a new chamber-based calibration technique that may provide better results within the 0.01–0.05 Hz band. This method uses a laser-based sensor to monitor loudspeaker displacement, in combination with a model of the acoustic chamber, to estimate the acoustic pressure at a sensor under test. The use of these two calibration methods may enable primary calibration over the full band of interest (0.01–10.0 Hz) and lend credence to each method. This talk will discuss low-band calibration challenges, the laser-based method, and calibration results.

Contributed Papers

10:20

4aPAa7. Studying the correlation between wind noise levels and topography for wind noise mapping and site selection. Ceu Jesus (Univ. of Mississippi, University, Oxford, MS 38677, mnetode@go.olemiss.edu), Roger M. Waxler (Univ. of Mississippi, University, MS), Claus Hetzer (Univ. of Mississippi, Tempe, AZ), Lance Yarborough (Univ. of Mississippi, University, MS), Carrick Talmadge, Hank Buchanan (Univ. of Mississippi, Oxford, MS), and Naveen Thirunilath (Univ. of Mississippi, University, MS)

Turbulent pressure fluctuations around infrasound sensors, known as wind noise, are the primary factor masking infrasound detections of interest, such as signals generated by natural hazards. Identifying deployment sites with sufficiently low wind noise levels is crucial for achieving good signal-to-noise ratios. Noise levels are expected to correlate with local topography, which influences wind flow patterns and turbulence. This study investigates the relationship between wind noise levels and topographic features to support wind noise mapping and optimal sensor site selection. Wind noise data were collected from multiple test sites and analyzed in two frequency bands: 0.1–1 and 1–10 Hz. rms pressure levels were calculated for each band, and Kernel Density Estimation (KDE) was applied to estimate probability density functions (PDFs). The associated cumulative distribution functions (CDFs) provide the probability of noise levels falling below specific thresholds, offering a statistical characterization of wind noise at each site. Topographic data were derived from remote sensing techniques, including LiDAR and satellite imagery, to analyze ground elevation, vegetation density, and canopy height. A framework for understanding wind noise behavior across varying topographic conditions, supporting the development of effective methodologies for wind noise mapping and site selection, is developed.

10:40

4aPAa8. Wind noise at infrasonic frequencies and near-surface turbulent distortion. Gordon M. Ochi (U.S. Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil), Carl R. Hart, Luis A. Camacho (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), and Benjamin F. Racelis (U.S. Army ERDC, Champaign, IL)

In atmospheric turbulence, multiple theories suggest that the wind noise reduction of a screened microphone should approach that of an unscreened microphone at a low wavenumber relative to the size of the windscreen. Typically, this wavenumber regime falls within the infrasonic frequency range. Contrary to existing theory, experimental data indicates that wind noise reduction occurs at low wavenumbers. It is hypothesized that near-surface turbulent distortion is the mechanism by which wind noise reduction is achieved at low wavenumbers. An overview of a theoretical model based on rapid distortion theory is discussed, as well as a recent outdoor experiment. Data were obtained for characterizing the near-surface turbulent distortion and internal pressures of windscreens in atmospheric turbulence. Comparisons between the rapid distortion theory model and experimental data are used to assess the validity of the proposed hypothesis.

THURSDAY MORNING, 22 MAY 2025 BALCONY I, 8:00 A.M. TO 11:00 A.M.

Session 4aPAb

Physical Acoustics and Structural Acoustics and Vibration: Mesoscopics in Acoustics and Elasticity II

Marco Scalerandi, Cochair
Department of Applied Science and Technology, Politecnico Di Torino, Politecnico Di Torino—Disat—Corso Duca Degli Abruzzi 24, Torino 10134, Italy

Sandrine T. Rakotonarivo, Cochair
Aix-Marseille University and Laboratory of Mechanics and Acoustics, LMA—UMR 7031 AMU—CNRS—Centrale Marseille, 4 impasse Nikola Tesla, Marseille 13453, France

Invited Papers

8:00

4aPAb1. Ballistic wave in correlated disorder media. Arthur Le Ber (Institut Langevin, ESPCI Paris—PSL, Paris, France), Yamil Abraham, Carlos Negreira, Nicolas Benech (Instituto de Fisica, Facultad de Ciencias, Montevideo, Uruguay), Alexandre Aubry (Institut Langevin, CNRS, Paris, France), Xiaoping Jia (Institut Langevin, ESPCI Paris—PSL, Paris, France), and Arnaud Tourin (Institut Langevin, ESPCI Paris—PSL, 1, rue Jussieu, Paris 75005, France, arnaud.tourin@espci.psl.eu)

We present an experimental and theoretical study of the influence of disorder correlations on the behavior of the ballistic coherent wave traveling through a strongly scattering medium. We considered two types of samples: on the one hand, a 3-D granular suspension of randomly packed submillimeter glass beads and, on the other hand, 2-D assemblies of randomly distributed millimeter copper rods

immersed in water. In the 3-D case, for broadband emission in the MHz range, the transmitted signal exhibits strong dispersion, leading to a signal composed of a low-frequency ballistic wave (not observed in previous analogous experiments) followed by a higher-frequency multiply scattered wave, known as the Coda Wave. With increasing thickness, the Coda tends to disappear due to absorption, so that only the ballistic wave remains. Experiments on 2-D samples provide information on its origin: we compared the ballistic transmission through a dilute sample (well described in the ISA approximation) and through a compact stack of rods. A low-frequency ballistic wave only appears in the latter case. A parallel numerical study indicates that this effect does not require contact between the scatterers but appears at high surface fractions, typically of the order of 50% for which disorder correlations must be considered.

8:20

4aPAb2. Anderson localization of acoustic waves in three dimensional resonant microbead suspensions. Fanambinana Delmotte (Phys. and Astronomy, Univ. of MB, 104 - 295 Stradbrook Ave., Winnipeg, MB R3L 0J5, Canada, Fanambinana.Delmotte@umanitoba.ca), Thomas Brunet (Univ. Bordeaux, CNRS—Bordeaux INP—ENSAM, I2M, Bordeaux, France), Jacques Leng (Univ. Bordeaux, CNRS—Syensqo, LOF, Pessac, France), and John H. Page (Phys. and Astronomy, Univ. of MB, Winnipeg, MB, Canada)

Anderson localization is one of the most fascinating wave phenomena that may occur in strongly scattering heterogeneous media. Since the 1980s, the experimental search for this halt of diffusive transport in 3-D disordered systems has continued to be the focus of intense research, whether for quantum particles or classical waves. In the latter case, the experimental demonstration remains challenging for light waves, contrary to ultrasonic elastic waves, for which it has already been proven without ambiguity. In this talk, I will describe a new set of two independent time- and position-resolved experiments, using techniques that were first developed to clearly establish the existence of localized regimes in mesoglasses made of sintered beads. Here, I will focus on scalar acoustic waves and report unambiguous evidence of the transitions between diffusion and localization in suspensions made of soft metallic resonant beads, inspired by recent advances in the field of soft acoustic metamaterials. These experiments allow us to determine the mobility edges of the localization regimes with high precision. Finally, as it is easy to vary the concentration in our model system, we will show that there is an optimal intermediate concentration beyond which localization disappears.

8:40

4aPAb3. Abstract withdrawn.

9:00

4aPAb4. The seabed as a mesoscopic porous medium. Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, chotiros@utexas.edu)

The seabed is composed of a variety of solid materials permeated by seawater, and therefore it is a poro-elastic material. In the case of granular seabed types, such as gravel, sand, and silt, there is little doubt regarding its porous nature. In the case of softer marine sediments, such as mud and clay, it has also been shown to be a poro-elastic material, in which the presence of saltwater is a critical ingredient. Depending on the sediment type, the pore sizes can range from fractions of a micron to a few millimeters, which puts it in the mesoscopic range. There are two main loss mechanisms, the creep loss within the skeletal frame and the viscous drag due to relative motion between frame and pore fluid. The main difference between poro- and visco-elastic media is the relative motion between the skeletal frame and the pore fluid. When the effects of the relative motion become negligible, a simpler visco-elastic approximation is justified. The equations of the visco-elastic approximation may be derived from the poro-elastic model. [N. Chotiros was funded by the U. S. Office of Naval Research, Code 32 Grant N00014-23-1-2522.]

9:20–9:40 Break

9:40

4aPAb5. Non-affine grain motion controls anomalous acoustic attenuation in granular packings. Abe Clark (Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, abe.clark@nps.edu), Derek Olson (Oceanogr., Naval Postgrad. School, Monterey, CA), and Colton A. Kawamura (Phys., Naval Postgrad. School, Franklin, NC)

We use discrete-element method (DEM) simulations to study the frequency-dependent dispersion and attenuation in model marine sediments. DEM explicitly models the motion of every grain using pairwise contact forces. Thus, our results explicitly isolate the disordered granular packing structure, which is not included in previous models of sediment acoustics (e.g., viscous poroelasticity and viscous grain shearing models). We study acoustic propagation through disordered packings, varying the contact parameters and frequency, as well as the dimensionless pressure (the ratio of confining pressure to the elastic modulus of the grains). For disordered packings at low pressure, which are more realistic, we observe scaling laws for the frequency dependence of both phase speed and attenuation that do not agree with continuum theories based on viscosity. Results for high-pressure packings (more similar to foams) agree with viscous-like continuum models. For three-dimensional packings, attenuation is nearly proportional to frequency, as is observed in many direct measurements. We show that this is due to non-affine motion at the grain-scale, caused by excess low-frequency vibrational modes in low-pressure packings. Our results demonstrate that packing structure must be considered in theories of sediment acoustics, and it may explain the observed linear frequency dependence for attenuation.

10:00

4aPAb6. Ultrasonic—Sorption characterization of water-sorbing carbon xerogel. Ashoka Karunarathne (Dept. of Chemical and Mater. Eng., New Jersey Inst. of Technol., 161 Warren St., University Heights/Tiernan Hall, Newark, NJ 07103, ashokatgam@gmail.com), Stephan Braxmeier (Ctr. for Appl. Energy Res. e.V. (CAE), Wuerzburg, Germany), Boris Gurevich (Ctr. for Exploration Geophys., School of Earth and Planetary Sci., Curtin Univ., Perth, Western Australia, Australia), Alexei Khalizov (Dept. of Chemistry and Environ. Sci., New Jersey Inst. of Technol., Newark, NJ), Gudrun Reichenauer (Ctr. for Appl. Energy Res. e.V. (CAE), Wuerzburg, Germany), and Gennady Gor (Dept. of Chemical and Mater. Eng., New Jersey Inst. of Technol., Newark, NJ)

Adsorption of water vapor in nanoporous carbons is rather complex due to an interplay between their pore structure and surface chemistry. Deciphering the mechanism of adsorption requires the knowledge of the spatial distribution and the filling fraction of the adsorbed water. Sending an ultrasonic wave through a nanoporous sample allows to retrieve a wealth of information, such as properties and spatial distribution of fluids confined in the pores. Here, we studied the adsorption of vapor water on monolithic carbon xerogel, a porous material with a bimodal pore size distribution consisting of micropores (1 nm) and mesopores (8 nm). A novel adsorption-ultrasonic experimental setup was employed to record the ultrasonic waveforms propagated through the water-sorbing xerogel sample while measuring its water sorption isotherm. Analysis of the elastic moduli evolution suggested that confined water shows nearly bulk-like properties in mesopores, while in micropores its modulus noticeably differs from that of bulk water. Furthermore, the observed increase in ultrasonic attenuation during micropore filling indicated the spatial heterogeneity of the water-filled pore space within the overall sample volume. This study demonstrates the utilization of nondestructive ultrasonic testing to probe both the fluid adsorption mechanism in a nanoporous medium and the properties of the adsorbed phase.

10:20

4aPAb7. What is the aeroacoustic mesoscopic scale? Roger Oba (Acoust. Div., US Naval Res. Lab., 4555 Overlook Ave. S.W., Washington, DC 20375, roger.m.oba.civ@us.navy.mil)

Recently, Yost [JASA 154, 2333–2336] and Berger [JASA 155, 3604–3605] posited that classical Brownian noise (thermal fluctuation noise in the acoustic medium) gives a limiting threshold for hearing. One could ask whether bats' hearing thresholds provide evidence of this limit, as their eardrums are much smaller and attuned to ultrasonics. To explore this threshold, acoustical and classical statistical mechanical arguments show that acoustic propagation requires both sufficiently large sample volumes and long enough integration times. These sample limits are needed to provide meaningful averages such that the thermodynamic fluctuations in pressure and mean velocity are significantly smaller than the acoustic field variations. This sets a frequency-dependent time-volume size resulting in a classically defined aeroacoustic mesoscopic scale. This scale is larger by some orders of magnitude than the usual thermodynamic mesoscopic scale, which represents a regime of classical and quantum interaction. It is further argued that aeroacoustic propagation can largely neglect the collision terms unless there is some coupling with various relaxation processes. [This research is supported by the 6.1 NRL base program sponsored by the Office of Naval Research.

10:40

4aPAb8. Ultrasonic measurements to differentiate hybrid processes in additively manufactured 316L stainless steel. Jazmin Ley (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE 68588, jley3@huskers.unl.edu), Cristian Pantea (Mater. Phys. and Applications, Los Alamos National Lab., Los Alamos, NM), Mark A. Anderson (Mater. Sci. and Technol., Los Alamos National Lab., Los Alamos, NM), and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Hybrid additive manufacturing (AM) generates materials with spatial variations by integrating additional manufacturing processes, such as milling, laser peening, or ultrasonic peening, at specific locations during the fabrication process. These material variations encompass alterations in average grain size, dislocation densities, and the introduction or alleviation of residual stresses. Importantly, these property changes extend beyond the applied layer, exhibiting a cumulative impact on preceding layers. The specifics of each hybrid process have been shown to improve mechanical properties, thereby enhancing the performance of components. However, there are distinctive challenges associated with the nondestructive validation of such samples. Traditional ultrasonic techniques have proven effective in mapping material variations with satisfactory spatial resolution in hybrid components. In this study, ultrasonic responses, i.e., wave speed, attenuation, and diffuse backscatter measurements, are used to differentiate between three different hybrid processes (milling, laser peening, and ultrasonic peening) in additively manufactured 316L stainless steel. This work is anticipated to impact the qualification of metal AM parts.

Session 4aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session II

Gregory M. Ellis, Chair
Audiology and Speech Pathology, Walter Reed National Medical Military Center, 4494 Palmer Rd N, Bethesda, MD 20814

All posters will be on display from 8:00 a.m. to 11:00 a.m. Authors of odd numbered papers will be at their posters from 8:00 a.m. to 9:30 .a.m. and authors of even numbered papers will be at their posters from 9:30 a.m. to 11:00 a.m.

Contributed Papers

4aPP1. Pickleball noise—A qualitative description of the psychological and physiological effects on nearby residents. Kathleen M. Romito (The Lasiewicz Foundation, The Lasiewiecz Foundation, 4634 Leir Dr., La Canada Flintridge, CA 91011, kathleen@lfprograms.org)

36 million people play pickleball in the US at over 50 000 courts. The typical conversion of one tennis court to 4 pickleball courts can generate over 3000 impulsive pops per hour or 30–40 000 pops/day. The chronic, unwanted, and repetitive noise can quickly wear on neighbors, leading to complaints, conflicts, and lawsuits. Acoustic engineers have focused on measuring the acoustical sound qualities of the pickleball "popping." However, there has been no research about the possible non-auditory health effects of chronic exposure to unwanted pickleball noise. Data were collected from public sources with a focus on pickleball noise: social media posts, news reports, and legal filings. Content analysis was used to identify possible health-related concerns mentioned by residents living near courts. Most commonly mentioned was trauma/PTSD, closely followed by auditory hallucinations (i.e., hearing "phantom pickleball noises"). Other concerns included both physiologic and psychologic issues. Further research is needed to validate these initial findings and further evaluate the extent of the issue. As communities grapple with this new type of sound, both acoustical measurements and the human health impact should be considered.

4aPP2. Factors affecting performance on a temporal interval discrimination task during adolescence and young adulthood. Bruna S. Mussoi (Audiol. and Speech Pathol., Univ. of Tennessee Health Sci. Ctr., Knoxville, TN), Merri J. Rosen (Anatomy and Neurobiology, Northeast Ohio Medical Univ., Rootstown, OH), and Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., CPA A144, Kent, OH 44242, jhuyck@kent.edu)

Auditory temporal processing typically matures well into adolescence. We investigated the cognitive and neurophysiological factors that affect performance on a temporal interval discrimination task during adolescence and young adulthood (age 10–24 years). Psychometric functions were measured using the method of constant stimuli. Stimuli consisted of two 1-kHz, 15-ms tones separated by a standard interval of 100 ms or a signal interval of 100 ms + Dt. Listeners chose which of the three sounds had the longer interval (3AFC). Results were highly variable, and some listeners of all ages were excluded due to an inability to complete the task or poorly fit psychometric functions. Stepwise regressions were conducted to determine the factors affecting performance. Unexpectedly, thresholds (66.67% correct) did not change with age. However, they were predicted by gender (better thresholds in men), auditory working memory, sustained attention, attention switching, and auditory encoding (amplitude of the N1 and latency of the P1 from the acoustic change complex). Slopes were predicted by log-transformed age (steeper slopes with increasing age), gender (steeper slopes in men), auditory working memory, attention switching, and vocabulary. Neither model predicted more than 40% of the variance, suggesting that other factors contributed to the high variability in performance. [Funded by NIDCD.]

4aPP3. Abstract withdrawn.

4aPP4. Role of statistical learning and prediction in auditory scene analysis. Vibha Viswanathan (Neurosci. Inst., Carnegie Mellon Univ., 4825 Frew St., Baker Hall 342C, Pittsburgh, PA 15213, vibhavis@andrew.cmu.edu), Srinidhi Narayanan (Biomedical Eng., Carnegie Mellon Univ., Pittsburgh, PA), Jenny R. Saffran (Psych., Univ. of Wisconsin–Madison, Madison, WI), and Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA)

Humans leverage statistical regularities in acoustic scenes to perceptually segregate target sound sources from other competing sounds. Acoustic regularities present in natural sounds like speech, including temporal coherence and harmonicity, support bottom-up grouping. Other higher-level regularities like linguistic structure must be learned to form a mental "schema" (stored knowledge) of statistical patterns in the environment. Schema-based segregation has been proposed to rely on predictive modeling of the acoustic scene; yet, prior studies have not established a direct link between prediction and segregation. To address this gap, we combined electroencephalography (EEG) and behavioral experiments. Following established statistical learning paradigms, we exposed listeners to sequences of speech syllables with specific syllable-transition probabilities. Preliminary data suggest that post-exposure, detection of a target syllable in competition improves when learned knowledge of between-syllable transition probabilities in the attended stream predicts the target. This prediction benefit is smaller for speech perception in quiet compared to in competition, suggesting that statistical prediction aids source segregation. Preliminary EEG data suggest that the parietal P300 event-related potential accompanying target-syllable detection occurs earlier when the target is predictable compared to when it is not. These experiments provide initial insights into prediction benefits in auditory scene analysis.

4aPP5. Switching source online segmented recursive least squares for real-time stimuli tracking in auditory attention decoding. Masayoshi Sakakura (Univ. of Illinois at Urbana Champaign, 1003 West Clark St., 1, Urbana, IL 61801, ms4@illinois.edu), Yongjie Zhuang, and Andrew C. Singer (Elec. and Comput. Eng., Stony Brook Univ., Stony Brook, NY)

The stimulus reconstruction-based auditory attention decoding (AAD) model reconstructs the attended auditory envelope from an electroencephalogram (EEG) recording with a finite impulse response (FIR) filter of an arbitrary order. Nonetheless, by nature, the filter coefficients, as well as

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the source of interest, are free to independently vary over time. To model such a phenomenon, the existing AAD works utilize piece-wise constant linear predictor of fixed observation length to track the switching source of interest as well as the dynamic linear predictive coefficients. However, such a method requires an appropriate selection of the fixed partitioning size, *a priori*. A suboptimal selection can exacerbate the overall performance, especially given sequences where the underlying structure does not conform to a constant partition size. Our proposed algorithm, the switching source online segmented recursive least squares (SS-OSRLS), based on the recursive Bellman equation identifies a set of partition points and the corresponding piece-wise constant linear predictors for each detected sub-segments, as well as the detected speaker of interest, all in real-time. We aim to demonstrate the real-time performance of our algorithm using the real EEG/audio dataset (i) without and (ii) with the pretrained decoder weight.

4aPP6. Cochlear implant perception of prosody is especially vulnerable to noise. Harley J. Wheeler (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, wheel488@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Perceiving prosody is essential for understanding a talker's intended meaning. For listeners with typical hearing (TH), the perception of voice pitch can withstand a high amount of background noise. However, listeners with cochlear implants (CI) lack access to harmonic pitch perception, putting them at risk for poor perception of prosody when noise is present, even if they appear to perceive pitch adequately in quiet. The current study tested the hypothesis that prosodic focus would be more heavily affected by noise for CI users compared to listeners with typical hearing. Stimuli were sentences with a contrastive focus on a specific word as if to correct prior information. The sentences were embedded in various levels of speech-shaped noise and presented with accompanying text to test only prosody rather than word identification ability. Participants used a visual analog scale to report the perceived degree of focus on words in each sentence. Results from TH listeners were essentially unaffected by noise regardless of the noise level (even down to −5 dB SNR). Conversely, CI users showed a high probability of misinterpreting which word was emphasized in sentences with any background noise, even with a favorable SNR of +10 dB, compared to their performance in quiet.

4aPP7. A grounded theory approach: exploration of students' perceptions in the informal study environments with a focus on attention deficit and hyperactivity disorder (ADHD). Semiha Yilmazer (Dept. of Interior Architecture and Environ. Design, Bilkent Univ., Faculty of Art, Design and Architecture, Ankara 06800, Turkey, semiha@bilkent.edu.tr) and Ceren Şahmaran (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

This study explores students' perceptions of the acoustic and visual environments in selected informal study spaces on the main campus of Bilkent University, focusing on distinctions between students diagnosed with attention deficit and hyperactivity disorder (ADHD) and those without. The primary objective is to categorize these perceptions systematically and develop a conceptual framework that clarifies how students with varying needs interact with their surroundings, employing a grounded theory approach. The sample includes 20 participants, evenly split between those with ADHD and their peers without diagnosis. Data were collected from four informal learning environments at Bilkent University—Dormitory 77, the Fine Arts Building, the library, and the Faculty of Science Building—chosen for their diverse acoustic properties and significance in learning and social interaction. Utilizing *in-situ* sound level measurements (LAeq), structured questionnaires, and semi-structured interviews, the study examines key themes, such as sound sources, noise responses, coping mechanisms, and the influence of acoustic and visual elements on student experiences. The findings reveal that the environmental and spatial characteristics, individual traits, and the specific study context shape the audio-visual perceptions of the students. Differences in responses to stimuli highlight the importance of inclusive designs that address the diverse needs of students, particularly those with ADHD.

4aPP8. Can human listeners label and discriminate natural soundscapes based on diel acoustical patterns? Frederic Apoux (Departement d'Etudes Cognitives, Ecole Normale Superieure, 29 rue d'Ulm, Paris 75005, France, fred.apoux@gmail.com), Nicole Miller-Viacava (Departement d'Etudes Cognitives, Ecole Normale Superieure, Paris, France), Bernie Krause (Wild Sanctuary, Sonoma, CA), Jerome Sueur (Institut de Systématique, Évolution, Biodiversité, Muséum national d'Histoire naturelle, Paris, France), and Christian Lorenzi (Departement d'Etudes Cognitives, Ecole Normale Superieure, Paris, France)

A previous study reported that human listeners can discriminate natural soundscapes with a high sensitivity and that they appear to base their decision on cues related to biological sounds and environment acoustics when discriminating habitats [Apoux *et al*., J. Acoust. Soc. Am. 153, 2706 (2023)]. The present study replicated and extended these findings while focusing on the sensitivity to diel acoustical patterns. Twenty-five listeners were asked to discriminate but also identify natural soundscapes recorded during a one-year period in a temperate forest at four distinct and identifiable moments of the day (Dawn, Midday, Dusk, or Night). In one experiment, stimuli were restricted in the audio frequency domain or noise vocoded to, respectively, assess the contribution of spectral and temporal cues to moment of the day discrimination. Overall, identification performance was poor, but discrimination was well above chance, suggesting that human listeners can derive some global properties of soundscapes associated with the time of the day, but struggle to assign these properties to the correct moment. The filtering data indicated that temporal cues and spectral details are not critical for moment of the day discrimination. Like habitat discrimination, listeners may rely primarily on gross spectral cues.

4aPP9. A method to incorporate the effect of clothing accessories in the head-related transfer functions for enhanced binaural experience. Luis A. Azpicueta-Ruiz (Signal Theory and Communications, Universidad Carlos III de Madrid, Avda Universidad 30, Leganés 28911, Spain, lazpicue@ing.uc3m.es) and Daniel De la Prida (Tech. Univ. of Madrid, Madrid, Spain)

Head-related transfer functions (HRTFs) are widely used in applications based on virtual acoustics, including virtual and mixed reality, advanced music experiences, TV shows, and video games. Today, the available sets of HRTFs can even be personalized to provide an enhanced binaural experience. However, when someone wears clothing accessories such as hats, cups, or hoods, the diffractions and reflections that the sound suffers as it reaches the eardrums change drastically. Whenever in a virtual scenario with an avatar wearing such accessories, the employed HRTFs should take their effects into account. This paper presents a method to modify any set of HRTFs to consider the influence of a specific accessory. To this end, we have created a database containing complete 3-D sets of HRTFs measured on a dummy head with different accessories in an anechoic chamber. For each accessory, we propose a novel transfer function that compares the HRTFs with the accessory to those measured in a bare head state. Our algorithm uses this transfer function to easily modify any set of standard HRTFs. This study also allows to evaluate how some accessories change the perceived sound spectrum. For example, the hood of a raincoat produces an obvious comb filter effect.

4aPP10. Testing the relationship between focused thresholds and spatial tuning curves as estimates of electrode–neural interface integrity in cochlear implants. Heather Kreft (Psych., Univ. of Minnesota, N218 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, plumx002@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

This study explored the electrode–neural interface (ENI) quality in cochlear-implant (CI) users by analyzing the relationship between absolute thresholds, spatial selectivity, and stimulation mode. Two hypotheses were tested: (1) high-quality ENI is associated with low absolute thresholds and narrow psychophysical spatial tuning curves (STCs), and (2) focused stimulation modes, such as steered quadrupolar (sQP), produce narrower STCs than the standard monopolar (MP) stimulation mode. Twenty-one CI users (30 ears) were included. Absolute thresholds were measured across each electrode array using focused stimulation, and STCs in both MP and sQP

were measured for the two electrodes on each array with the highest and lowest focused thresholds. Consistent with hypothesis 2, STCs were broader with MP than sQP stimulation. However, neither STC bandwidths/slopes nor overall dynamic ranges were significantly different between the high- and low-threshold electrodes. Speech perception scores using PRESTO-N sentences were significantly correlated with MP threshold variability across the array, but not with sQP variability or STC measures. In summary, while focused stimulation produced narrower STCs than MP stimulation, lower focused thresholds did not predict narrower STCs, inconsistent with a strong relationship between ENI, focused thresholds, and STCs. [Work supported by NIH grant R01DC012262.]

4aPP11. Cue-profile in noise. Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Benjamin Amartey (Northwestern Univ., 2240 Campus Dr., Evanston, IL, benjamin.amartey@northwestern.edu), Richard A. Wright (Linguist, Univ. of Washington, Seattle, WA), and Pamela E. Souza (Northwestern Univ., Evanston, IL)

Speech understanding depends on the identification of phonemes, each of which is defined by a variety of cues. The most precise cues involve spectrotemporal modulations (STMs) such as formant transitions, but internal distortions (such as hearing loss) and external distortions (such as background noise) can reduce the ability to track STM information. In that case, temporal (envelope) cues can be used instead. Previous work showed that a speech cue profile test could be used to characterize individuals' use of STM and envelope cues, both with and without external distortions (time compression and amplitude compression). Here, external distortion in the form of background noise was used to further examine the ability of the cue profile test to characterize individual cue use. A computational model of the auditory nerve (Zilany *et al.*, 2014) was used to predict the degree of STM and envelope distortion for two types of noise: 12-talker babble and speech-shaped noise based on the spectrum of the babble. Listeners with and without elevated hearing thresholds completed the cue profile test in quiet and in both noise types. Results will be discussed with reference to previous results as well as by comparing the individual differences to the model predictions.

4aPP12. Investigating how narrator acoustics influence continuous speech-evoked multiband auditory brainstem responses. Eric A. Mitchell (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, mitc0698@umn.edu) and Melissa Polonenko (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Frequency-specific auditory brainstem responses (ABRs) are used to assess hearing function and can now be measured to continuous "peaky" speech audiobook stimuli. However, these speech-evoked multiband ABRs are small. The amplitudes can be boosted if using narrators with lower fundamental frequencies (f0s) and peaky speech composed of chirp-phase profiles. But even if set to the same optimal parameters, different narrators and stories evoke differently sized multiband ABRs, thereby impacting the length of testing required to ensure all participants have visible responses. This may limit what audiobooks can feasibly be used. Thus, this study aimed to determine what factors of the remaining narrator/story differences contribute to the variability of ABR amplitudes. Data comprised both measured ABRs across three previous studies in adults with normal hearing and modeled ABRs of the same stimuli. Stimuli comprised stories from different narrators and two stories from the same narrator, all with a chirp-phase profile and mean f0 of 90 Hz. Response amplitude differences were compared and correlated with stimulus f0 distribution statistics, such as variation, kurtosis, and skewness. Understanding which stimulus factors most influence ABR responses will expand the diversity of audiobooks that can be used for feasible and engaging multiband ABR measurement.

4aPP13. Hearing distortion to "hear the world". Jeff J. Stolley (Interdisciplinary Graduate Program in Neurosci., Univ. of Iowa, 240 Canal St., Apt. 234, Lawrence, MA 01840, Jeff.J.Stolley@gmail.com) and James Traer (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Separating sounds into distinct causal components (i.e., auditory scene analysis, as exemplified by the "cocktail party problem") is fundamental to hearing. We investigated how human comprehension of a single voice in a mixture is affected by various distortions (e.g., reverberation, filters, power compression, etc.), which are common in real-world scenes, in telecommunications technology, and induced by hearing aids. We demonstrate that distorting the mixture of voices reduces comprehension (as expected), but that distorting the target voice only (by comparable amounts) can aid comprehension. This shows distortion can sometimes aid listeners, presumably by making the voices more separable. In another experiment, we assess the human ability to recognize distortion itself, by asking listeners to match equivalent degrees of distortion across recordings of different voices. We show humans can robustly recognize both speaker identity and degree of distortion when both vary unpredictably. As distorted sounds are structured by both the source and the particular structure of the distortion, to successfully recognize the source and/or the transmission channel the auditory system must separately infer the two causal factors. Thus we propose that hearing a single distorted sound may itself be an auditory scene analysis problem, and can be studied as such.

4aPP14. Linking fast adaptation, slow adaptation, and mechanical creep in mammalian hair cell mechano-electric transducer channels. Varun Goyal (Mech. Eng., Univ. of Michigan, Ann Arbor, 3632, G.G.B. Labs., 2350 Hayward St., Ann Arbor, MI 48109, varungo@umich.edu) and Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI)

The ear relies on the transduction of sound input to mechano-electric transducer (MET) current by hair bundles (HBs) for normal hearing. These HBs, composed of hair-like stereocilia, have ion channels at the tip of all shorter stereocilia that enable transduction. Experiments have shown that HBs stimulated by a step-like fluid-jet force display a displacement creep post initial ascent, termed mechanical rise, and the MET current adapts with a single slow time constant. Under a step-like probe-actuated displacement stimulus, the creep disappears, and the current decays with two (fast and slow) time constants. Since adaptation plays a possible role in precluding cell damage and restoring HB sensitivity upon exposure to loud sounds, it is important to understand the underlying mechanisms for these seemingly disparate responses. We developed a single nonlinear model of an isolated mammalian HB capable of representing the mechanical and electrical response under both the slow fluid jet and much faster stiff probe stimulus. Linearizing the model enabled us to identify the three underlying system time constants (mechanical rise, fast, and slow adaptation), along with a mechanistic explanation of how these three different behaviors arise.

4aPP15. Effect of sound level on pitch and timbre discrimination using low- and mid-pitched complex tones. Wesley Bulla (Audio Eng. Technol., Belmont Univ., 1900 Belmont BLVD, Nashville, TN 37212, wesley.bulla@belmont.edu) and Song Hui Chon (Audio Eng. Technol., Belmont Univ., Nashville, TN)

Discrimination of pitch and spectral detail is an essential skill for audio engineers, musicians, and music professionals. Auditory tuning curves are known to expand with increasing signal strength (Evans, 1993), which in turn spreads temporary threshold shifts across cochlear bands (Regers, 2016). These observations suggest that auditory discrimination may be directly affected by the intensity of the signal. Psychophysical evidence was found in our earlier work that robust signal levels had a consistently negative influence on the accuracy of both pitch and timbre discrimination when compared to soft signals (Bulla and Chon, 2021). Discrimination performance was significantly better at 65 dBA SPL (typical conversational level) compared to performances at 55, 75, and 85 dBA SPL in a follow-up study (Bulla and Chon, 2024), suggesting a possible "discrimination sweet spot." However, while promising, these experiments utilized sinusoidal tones for the pitch task and complex tones in the timbre task. The present study expands prior work by utilizing both tasks complex tones with low- and mid-pitches to examine the effect of pure versus complex tones at different pitches. Data collection is ongoing and results will be reported with new observations and outlines for potential future research.

4aPP16. Level dependence of distortion-product otoacoustic emissions and intracochlear distortion products in mice. James Dewey (Dept. of Otolaryngol.—Head & Neck Surgery, Univ. of Southern California, Zilkha Neurogenetic Inst., Rm. 407, 1501 San Pablo St., Los Angeles, CA 90033, james.dewey@med.usc.edu)

Distortion-product (DP) otoacoustic emissions (DPOAEs) provide a noninvasive window onto cochlear mechanics and outer hair cell (OHC) function. While the growth of DPOAEs with stimulus level has been used to infer cochlear sensitivity and characterize aspects of the underlying nonlinearity, the relationship between the level dependence of DPOAEs and intracochlear DPs remains poorly understood. Here, this relationship was assessed by comparing DPOAEs with DPs measured from the apex of the live mouse cochlea using optical coherence tomography. With the f_2 stimulus frequency fixed at the characteristic frequency of the measurement location (~9 kHz), the dependence of $2f_1 - f_2$ DPOAEs and intracochlear DPs on the f_1 stimulus frequency and the levels of both stimulus tones was thoroughly examined. For f_2/f_1 ratios eliciting the largest DPOAE amplitudes, the growth rates of DPs and intracochlear DPs were often similar at low stimulus levels. However, at small f_2/f_1 ratios and/or high stimulus levels, DPOAEs could exhibit sharp notches and/or growth rates that were markedly different from those observed for the intracochlear DPs. The results are consistent with DPOAEs being strongly shaped by wave interference between spatially distributed sources. These interference effects and their stimulus dependence must be considered when interpreting DPOAE growth patterns.

4aPP17. Auditory detection of water sounds in a natural habitat: Psychophysical and modeling data. Matthieu Fraticelli (Ecole normale supérieure, Université Paris Sci. & Lettres, 45 rue d'Ulm, Paris 75005, France, contact@matthieufraticelli.com) and Christian Lorenzi (Ecole normale supérieure, Université Paris Sci. & Lettres, Paris, France)

This study explored the ability of humans to detect water sounds within natural scenes. Here, we assessed the detection performance of a stream sound in a Mediterranean forest. Acoustic samples were recorded at six distances (0–200 m) from a river, at three moments of the day and two seasons. 2-s long samples were presented diotically to 3333 participants tested online using a 2-interval, 2-alternative forced-choice detection paradigm. On a typical trial, listeners were presented in random order with a target interval playing a sample recorded between 0 and 100 m from the river and a comparison interval playing a sample recorded 200 m away from the river and then asked to report the interval containing water sounds. Each participant completed a single trial only. Water detection performance depended on distance and season only. Water detection was then simulated using an auditory texture model using the same task and stimuli. The model replicated the main trends found in human data when basing its decisions upon skewness and kurtosis of amplitude envelopes and the correlation of envelope fluctuations across modulation bands. Together, these results suggest that humans detect water in natural environments on the basis of sparseness and coordination of temporal-envelope cues. [Work supported by ANR AUDIECO/ALEAU.]

4aPP18. Simulating the contribution of medial olivocochlear efferents to amplitude-modulation unmasking in humans. Daniel R. Guest (Dept. of Biomedical Eng., Univ. of Rochester, 601 Elmwood Ave., MC 5-6483, Rochester, NY 14620, daniel_guest@urmc.rochester.edu) and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Amplitude-modulation (AM) detection thresholds for a tonal carrier in noise are often improved by the presentation of a precursor [e.g., Wojtczak *et al.*, JARO 20, 395–413 (2019)]. This phenomenon, AM unmasking, may reflect activation of the medial olivocochlear reflex (MOCR) by the precursor, which would reduce cochlear gain and enhance AM coding during the tone. However, evidence regarding the MOCR's role in AM unmasking is mixed. We simulated AM unmasking using a new physiological model of the peripheral and subcortical auditory system that includes an MOCR. At all carrier sound levels (40–80 dB SPL), precursor noises enhanced neural representations of AM and improved AM detection thresholds in at least some tonotopic channels, consistent with a role for the MOCR in AM unmasking. At low levels, channels tuned near the carrier frequency were generally the most informative and exhibited the greatest AM unmasking. At high levels, the informative channels were tuned further (>0.5 octaves) from the carrier frequency, suggesting the use of "off-frequency listening." Consistent with behavioral results, noise precursors produced more simulated AM unmasking than pure- or complex-tone precursors. These results suggest an important role for the MOCR in the perception of complex sounds. [Work supported by NIDCD F32 DC022143.]

4aPP19. Perceptual and neural correlates of sensory and cognitive priming in chord sequence. Anjelica J. Ferguson (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, 4720 Ctr. Ave., Apt. 5F, Pittsburgh, PA 15213, anjelicaferguson@gmail.com), Jackson E. Graves (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI), Barbara Tillmann (LEAD CNRS UMR5022, Université de Bourgogne, Lyon, France), and Anahita H. Mehta (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI)

Harmony perception in music is subject to cognitive priming based on tonality as well as sensory priming from spectral overlap and pitch cues. This study aims to determine the relative contributions of cognitive and sensory priming to harmonic expectation, measured using behavioral responses as well as the early right anterior negativity (ERAN) from the electroencephalogram (EEG). We used seven-chord sequences divided into three conditions: expected, unexpected, and atonal (no strong expectations). To control for spectral priming, the context chords and the final chord were filtered into mutually exclusive spectral regions. To control for pitch priming, the context chords shared no pitches with the final chord in one condition. Listeners rated stimuli on a scale from 1 (unexpected) to 7 (expected) and then listened to the same stimuli in a passive listening EEG paradigm. Behavioral data showed a consistent effect of harmonic expectation across filtering conditions, confirming a cognitive explanation for this effect. EEG data showed an ERAN in "unexpected" conditions when pitch priming is present, but with a less pronounced ERAN in conditions that control for pitch priming, suggesting that pitch adaptation and sensory priming play a role in early brain responses to harmony. [Work supported by NIH grant R00DC017472 (AHM)]

4aPP20. Impact of mild to moderate sensorineural hearing loss on single and multiple pitch perception. Jackson E. Graves (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, 29 rue d'Ulm, Paris 75005, jgra@med.umich.edu), Sophia G. Riegle (National Tech. Inst. for the Deaf, Rochester Inst. of Technol., Rochester, NY), Tess M. Starr (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI), Kelly L. Whiteford (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, Minneapolis, MN), and Anahita H. Mehta (Kresge Hearing Res. Inst., Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI)

Everyday auditory scenes often contain multiple sound sources, each with a different fundamental frequency (F0) and corresponding pitch. The ability to focus on one talker in competing sounds or listen to harmony in music partly relies on isolating one pitch from a mixture. Mild to moderate hearing loss (HL) degrades spectrotemporal cues and adversely affects single-pitch perception. Although extensive research has been conducted on speech perception in listeners with HL and the efficacy of hearing aids, little is known so far about how HL affects the perception of multiple-pitch combinations and whether hearing aids alleviate these deficits. In this behavioral study, we measured aided and unaided thresholds in listeners with mild to moderate HL for four tasks that involved single-pitch stimuli and multiple-pitch combinations. Preliminary data suggest that HL results in mild deficits in single-pitch perception that become severe for multiple pitches, with hearing aids providing little to no improvement. These findings highlight the need to better characterize neural mechanisms of multiple pitch perception to inform effective restoration of these cues in listeners with HL. [Work supported by NIH grant R00DC017472 (AHM)]

4aPP21. Effects of age on modulation masking patterns for different shapes of masker waveforms. Neha Rajappa (Psych., Univ. of Minnesota, 75 E River Pkwy, Minneapolis, MN 55455, rajap013@umn.edu), Andrew J. Byrne, and Magdalena Wojtczak (Psych., Univ. of Minnesota, Minneapolis, MN)

Modulation-rate selectivity, measured using a modulation-masking paradigm, has been shown to be differentially affected by advancing age (reduced modulation selectivity) and age-related hearing loss (increased modulation selectivity). In this study, we systematically explored the relationship between modulation masking patterns and age-related loss of cochlear nonlinearity. The study compared modulation masking for younger (18–30 years) and older (55–75 years) adults with clinically normal hearing for frequencies up to 4 kHz. Modulation masking patterns were measured for different shapes of modulation masker waveforms: sinusoidal, ramped and damped, and two modulation waveforms with the magnitude spectrum of a rectangular modulation but different phase spectra. The non-sinusoidal masker shapes were chosen to test the generalizability of the modulation power spectrum model. The signal was sinusoidal amplitude modulation (SAM) with modulation rates at and around the masker frequency. The modulating waveforms were applied to a noise carrier lowpass-filtered at 4 kHz. Loss of cochlear nonlinearity was estimated using the growth of distortion-product otoacoustic emissions. We found that the effects of age and strength of cochlear nonlinearity on modulation masking depend on the modulation rate and the shape of the masker waveform. [Work supported by NIH R01 DC 021362 (Wojtczak)]

4aPP22. Development of the neural signatures of stream segregation from childhood to adulthood. Elena Benocci (CRCN, Université Libre de Bruxelles, Brussels, Belgium), Claude Alain (Baycrest, Toronto, ON, Canada), and Axelle Calcus (CRCN, Université Libre de Bruxelles, 50, av. F. Roosevelt, Brussels 1050, Belgium, acalcus@ulb.ac.be)

When faced with noisy environments, listeners perform auditory scene analysis, which allows them to selectively focus on the relevant auditory target while ignoring interferences. Stream segregation involves organizing similar sound waves into a coherent stream, while distinguishing dissimilar acoustic components and attributing them to distinct sources. Two event-related potential components have been identified as "neural signatures" of stream segregation: the Object-Related Negativity (ORN) and the P400. Our study aims to examine (i) the maturation of neural correlates of stream segregation and (ii) the development of the relationship between these neural correlates and speech perception in noise. ORN/P400 were recorded while 8–23 year-olds ($n = 75$) performed an active stream segregation task based on temporal coherence. Participants also performed speech identification in noise tasks (behaviorally). Behavioral results indicate an improvement in both stream segregation and speech perception in noise from childhood to adulthood. The amplitude of the ORN (but not P400) decreases, and the latency of both ORN and P400 decreases throughout development. Critically, P400 amplitude significantly predicts stream segregation performance. Overall, our results suggest that the neural mechanisms underlying stream segregation follow a prolonged maturation trajectory, and support the progressive maturation of auditory scene analysis and speech perception in noise.

4aPP23. The influence of puberty on the maturation of the central auditory system. Ellen Saliën, Ugo Benrubi (CRCN, Université Libre de Bruxelles, Brussels, Belgium), and Axelle Calcus (CRCN, Université Libre de Bruxelles, 50, av. F. Roosevelt, Brussels 1050, Belgium, acalcus@ulb.ac.be)

Late Auditory Event-Related Potentials (LAERs) are widely used to investigate central auditory responses to sounds. Components of the LAERs reflect successive stages of sound processing throughout the auditory pathway and are influenced by visual information. From infancy to old age, LAERs show morphological changes. Their maturation is thought to follow a stepwise trajectory, marked by distinct changes at the start and end of adolescence. Interestingly, the beginning of adolescence coincides with puberty onset, which triggers a cascade of hormonal changes that may lead to enhanced neural plasticity. Puberty could thus contribute to the maturation of sensory brain regions, including the central auditory cortex in the temporal brain region. The aim of this study was to investigate the effect of puberty on the development of speech processing. Participants ($n = 110$, ranging from pre- to late-pubertal stages) were presented with speech stimuli in an oddball paradigm across three conditions: audio only, audiovisual congruent, and audiovisual incongruent. Preliminary findings suggest that puberty drives certain changes in LAER morphology throughout adolescence. Puberty may play a role in the functional maturation of the central auditory pathway, potentially contributing to the fine-tuning of adolescents' ability to process speech in ecological settings.

4aPP24. Neural representation of auditory sound categories across cognitive tasks in electroencephalography.. Moïra-Phoebé Huet (Johns Hopkins Univ., 3400 North Charles St., Baltimore, MD 21218, mphuet@jhu.edu) and Mounya Elhilali (Johns Hopkins Univ., Baltimore, MD)

Auditory processing is influenced by task goals, which shape how we perceive and prioritize sound stimuli. Speech, a key category for human communication, often coexists with sounds such as music, environmental sounds, and noise. Tasks that require either comprehending speech in a noisy environment or scanning for multiple sound categories engage distinct neural mechanisms. To investigate these task-driven modulations, participants were exposed to the same auditory scene, composed of speech, music, other sound categories, and noise, under two distinct goals: focusing solely on speech or identifying all sound categories. EEG recordings and an attentional decoding method were used to reconstruct the original sound categories from brain activity. The findings revealed distinct behavioral and neural patterns across sound categories, demonstrating that each category is associated with different reconstructed spectrotemporal modulation profiles. Notably, speech showed the strongest neural tracking, particularly in the comprehension task compared to the detection task. These results show that task goals not only shape attentional allocation but also influence the neural representation of sound. This demonstrates the flexibility of the auditory system in adapting to cognitive demands, shedding light on how selective attention dynamically shapes auditory processing.

4aPP25. Modulation-filter shapes in the budgerigar inferred from behavioral reverse correlation. Kenneth S. Henry (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave., Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu)

Much information in speech is carried by amplitude modulation (AM). AM perception is adversely impacted by competing AM fluctuations present in the noise background, called modulation masking. Modulation masking in humans increases for masker AM frequencies closer to the target, suggesting a bandpass modulation filterbank analysis. The modulation filterbank model is widely adopted in hearing science, but animal models are relatively undeveloped. Using narrowband modulation maskers, we recently found that budgerigars, a parakeet species, show bandpass behavioral modulation masking patterns consistent with the modulation filterbank model. Here, we used wideband modulation maskers and a behavioral reverse correlation approach to characterize modulation shapes. Masker seeds were stored for stimulus reconstruction on a trial-by-trial basis. Animals were trained to perform a single-interval, two-alternative, forced-choice, AM detection task with operant conditioning. The reverse correlation approach estimated modulation filter shapes related to the difference in average stimulus statistics (e.g., envelope spectrum) between response categories (e.g., false alarms versus correct rejections). Most animals showed clear differences in stimulus statistics between response categories which were in some cases explainable by bandpass modulation filtering. Behavioral reverse correlation results are compared to modulation masking patterns with narrowband modulation maskers. [Work supported by R01-DC021953.]

4aPP26. Geometric gain approximation dictates the accuracy of hair bundle models. Varun Goyal (Mech. Eng., Univ. of Michigan, Ann Arbor, MI) and Karl Grosh (Mech., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48105, grosh@umich.edu)

Sound perception arises from oscillations of stereocilia, hair-like structures that convert mechanical motion into electrical signals via ion channels. These signals, depending on the stereocilia's location, drive neural

activation and nonlinear sound amplification, crucial for normal hearing. Mathematical models of isolated hair bundles (HBs), composed of stereocilia, provide a mechanistic understanding of these processes. A key parameter in these models is geometric gain, which estimates how mechanical deflections impact gating spring extension, modulating ion channel activity. Traditionally, this gain is approximated as the ratio of horizontal spacing between adjacent stereocilia pivots to their average height. However, this simplified ratio can lead to inaccuracies in predictions of HB sensitivity and stiffness. We analyzed gain at five cochlear locations in adult mice, comparing two definitions: the approximated "stick geometric gain," based on two infinitesimally thin stereocilia, and the "true geometric gain" derived using our nonlinear two-row isolated HB model with complete morphology. We found that the stick model overpredicts gain by a factor of 1.5–2, resulting in overestimated changes in gating spring tension, sensitivity, stiffness, and narrower activation curves. We discuss the implications of gain-overprediction on the interpretation of experimental results and theoretical models. [Work supported by NIH-R01-NIDCD04084].

4aPP27. Deficits in "aided" hearing: The role of distorted tonotopy in sensorineural hearing loss. Hari Bharadwaj (Commun. Sci. and Disord., Univ. of Pittsburgh, 3600 Atwood St., Forbes Tower 5063, Pittsburgh, PA 15213, hari.bharadwaj@pitt.edu), Satyabrata Parida (Oregon Health & Sci. Univ., Portland, OR), Homeira I. Kafi (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), Joshua M. Alexander, and Michael G. Heinz (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Restoration of audibility through frequency-specific amplification is central to the clinical management of sensorineural hearing loss (SNHL). Yet, patients often struggle to understand audible speech, especially in noisy environments. These suprathreshold deficits are conventionally attributed to reduced frequency selectivity and to non-peripheral factors. However, our cross-species studies show that damage to cochlear hair cells not only broadens auditory filter "tips" as previously recognized, but can also distort the fundamental tonotopy of the cochlea such that the temporal responses of the base are commandeered by "off-frequency" (i.e., low-frequency) sound fluctuations through overzealous filter "tails." This effect is especially pronounced with naturalistic stimuli and background noise possessing pink-like spectra, which contain intense low-frequency components alongside softer but informative higher-frequency content. In a chinchilla model of SNHL with noise-induced permanent threshold shifts, single-unit auditory-nerve measurements revealed that hypersensitive tuning-curve tails were the dominant driver of degraded speech envelope coding, even with sound amplifier gains akin to modern hearing aids. In parallel human studies, individuals with mild or moderate SNHL showed hypersensitive tuning-curve tails linked to impaired speech envelope tracking as measured through electroencephalography. Crucially, individual differences in the estimated degree of distorted tonotopy predicted aided speech-in-noise outcomes.

4aPP28. Association of hearing assessment methods with cognitive impairment and dementia: A systematic review. Guangxiang Zhong (Rehabilitation College, Kunming Medical Univ., No. 1168, Chunrong West Rd., Yuhua St., Chenggong District, Kunming 650500, Yunnan, China, nac18263750@163.com), Xin Zhao, Yao Liu (Rehabilitation College, Kunming Medical Univ., Kunming, China), and Lei Yang (No. 2 People's Hospital of Kunming, Kunming, China)

Background: Auditory impairment may precede the decline in cognitive function and the onset of Alzheimer's disease. The question arises whether auditory function testing can predict these conditions. Methods: A systematic literature search was conducted using PubMed, MEDLINE, Web of Science, and Cochrane Library (last search done in December 2023). The methodological quality was rated using the Consensus-based Standards for the selection of health Measurement Instruments (COSMIN) checklist. Results: Fifty-four studies met our inclusion criteria. In these studies, there were approximately 340 390 participants. For the assessment of central auditory processing impairment, SSI-ICM (RR, 12.48; 95% CI, 2.69–57.82; OR, 1.52; 95% CI, 1.14–2.03), Dichotic Digits Test (HR, 2.66; 95% CI, 1.31–5.42; Sensitivity = 25.9%, Specificity = 92.6%), and Dichotic Sentence Identification test (HR, 4.18; 95% CI, 2.37–7.38; Sensitivity = 83.8%, Specificity = 58.6%) results were found to be significantly associated with dementia and memory impairment, and the sensitivity was high. Peripheral hearing loss is linked to dementia risk (HR, 1.25; 95% CI, 1.01–1.55), with PTA positively associated with cognitive impairment risk (OR, 1.82; 95% CI, 1.27–2.61), particularly for high-frequency loss. Conclusions: The results of this systematic review indicate that there is a good correlation between different hearing test results and the prediction of cognitive function at various levels, with varying degrees of validity.

4aPP29. Lateral position discrimination using wavefield synthesis in an open environment to test the effect of observer-controlled motion on the ventriloquist effect. Anjelica J. Ferguson (Psych., Carnegie Mellon Univ., 4720 Ctr. Ave., Apt. 5F, Pittsburgh, PA 15213, anjelicaferguson@gmail.com), Sungjoon Park (Psych., Carnegie Mellon Univ., Pittsburgh, PA), Daniel Rosenberg Munoz (Design, Carnegie Mellon Univ., Pittsburgh, PA), and Laurie M. Heller (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

We studied the ventriloquist effect, in which sound locations are perceived near visual objects, under naturalistic conditions wherein observers moved real objects in an open environment. A wavefield synthesis array produced sound sources that were either collocated with the object or offset to its left or right. Using motion tracking cameras, the sound sources could either move along with the objects while observers moved them (congruent motion) or remain stationary while the object was moved (incongruent motion). First, a 200-ms 79 dB SPL broadband sound was presented on either side of a stationary object. Twenty seated listeners discriminated the relative position of the sounds in a 2IFC task in a three-down one-up adaptive track. As expected from the ventriloquist effect, lateralization threshold was higher when the object was present than when it was absent. Next, we tested the hypothesis that auditory and visual motion congruence would promote the binding of sounds and objects, while incongruent motion would disrupt it. Lateralization thresholds were higher in the congruent motion condition in which the sound was collocated with the moving object. However, inconsistent with our predictions, thresholds were just as high in the incongruent motion condition. [Work supported by ASA SURIEA, CMU Dietrich Grant.]

4aPP30. Predicting cochlear bandwidth with biomechanical measurements. Aleksandrs Zosuls (Mass Eye and Ear, 243 Charles St., Boston, MA 02114, ALEKSANDRS_ZOSULS@MEEI.harvard.edu), Stefan Raufer (Harvard Speech and Hearing Bioscience Technol., Staefa, Switzerland), Andrew Tubelli (Mass Eye and Ear, Boston, MA), and Darlene R. Ketten (Neurosciences/Biology, Brown University/WHOI, Boston, MA)

For many species, it is impractical or prohibited to obtain audiograms by invasive methods. We hypothesize that anatomical and mechanical measurements made on postmortem tissue input into models can be used to predict the bandwidth of the cochlea and the tonotopic frequency map. Measurements were made on control species with known cochlear maps and tissue conditions to develop models and establish the degradation effects of tissue. Anatomical measurements of the cochlea were performed using histology and computed tomography. Measurements were made of the stiffness, and in some cases, the vibration patterns of the basilar membrane at multiple locations. Comparison of the model data with audiograms of chinchilla, gerbil, human, bottlenose dolphin, and harbor porpoise were used as controls. Measurements were performed in 13 species on a total of 152 individual cochleae. For a given species, measurements more basal along the cochlea were stiffer or tuned higher in frequency. In the controls, the model predictions agreed reasonably with audiogram upper frequency limits. Measurements in the apical regions were too compliant to measure reliably or the basilar membrane was found to have lost its tension. Echolocating species have a significantly higher stiffness for a given width of the basilar membrane compared to non-echolocators.

4aPP31. Investigating the effect of rate on frequency modulation and frequency change detection of harmonic complex tones with resolved or unresolved harmonics. Penelope J. Corbett (Psych., Univ. of Minnesota, 1447 Macey Way, Stillwater, MN 55082, corbe168@umn.edu), Kelly L. Whiteford (Kresge Hearing Res. Inst., Dept. of Otolaryngology-Head & Neck Surgery, Univ. of Michigan, Minneapolis, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

The dual-code theory posits two neural mechanisms for frequency-modulation (FM) detection with pure tones or complex tones with spectrally resolved harmonics: At slow rates, precise neural phase-locking to the tones' instantaneous frequencies conveys FM; at faster rates, detection is aided by amplitude-modulation (AM) cues, produced via cochlear filtering and FM-to-AM conversion. With unresolved harmonics, FM detection seems based on extraction of the instantaneous fundamental frequency (F0), via phase-locking to the temporal envelope, with no alternative cues available at faster FM rates. To test this theory, FM sensitivity was measured for complex tones with resolved or unresolved harmonics (F0 = 100 Hz) at four rates (2, 5, 10, or 20 Hz) and compared with performance on equivalent F0 change-detection tasks. The theory predicts more degradation with rate for FM and frequency-change detection with unresolved harmonics, and for frequency-change detection with resolved harmonics, than for FM detection with resolved harmonics, where FM-to-AM conversion may aid performance at faster rates. Preliminary results do not fully match these predictions, suggesting instead similar patterns as a function of rate for resolved and unresolved harmonics in each task, but some differences in patterns with rate between the FM-detection and frequency-change-detection tasks. [Work supported by NIH, Grants R21DC019409 and R01DC005216.]

THURSDAY MORNING, 22 MAY 2025 BALCONY M, 9:20 A.M. TO 11:00 A.M.

Session 4aSA

Structural Acoustics and Vibration: Aerospace and Structural Acoustics

Peter Kerrian, Cochair
ATA Engineering Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128

Bhisham Sharma, Cochair
Mechanical Engineering - Engineering Mechanics, Michigan Tech University, 1400 Townsend Drive, Houghton, MI 49931

Contributed Papers

9:20

4aSA1. Predictability of elastic properties and density of 316L stainless steel manufactured by Selective-Laser-Melting. Gabriela Petculescu (Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504, gp@louisiana.edu)

Ultrasonic investigations on 316L stainless steel manufactured by Selective-Laser-Melting (SLM) aimed to correlate elastic properties with fabrication parameters. In general, as a new technology matures, qualification is essential, especially when the technology is to be used remotely. In particular, accurate properties are required when designing prototypes. For the investigated sample set, laser power and scanning speed were varied systematically from 40 to 100 W, and 600 to 1200 mm/s, respectively. Resonant Ultrasound Spectroscopy and Pulse-Echo measurements were performed, complementarily. Numerical mode-analysis was also used to investigate the effect of void distribution, which was found inconsequential for most cases (except those of very low density). The extensive data allow for observations of specific characteristics in the elastic properties and microstructure as a function of deposited energy. Most importantly, the multi-variate data imply that standard volumetric energy density cannot be used to account for differences in the properties of the fabricated material. An effective volumetric energy density, which better accounts for the realistic non-adiabatic nature of the heating process of the melt pool, is adopted. In the new representation, the current and prior results from literature with different fabrication parameters can be compared, revealing a universal behavior of 316L produced through the SLM process. A Goldilocks region of predictability is identified.

9:40

4aSA2. Effects of variability in 3-D printed acoustic liner geometry on acoustic impedance. Laura Thomas (Dept. of Mech. and Mater. Eng., Florida Int. Univ., 10555 W Flagler St., Miami, FL 33174, lthom184@fiu.edu), Matthew B. Galles, and Andrew Christian (Appl. Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

Acoustic liners can be installed within the nacelle of turbofan engines to reduce the emitted noise. These liners are generally composite panels with a honeycomb core sandwiched between a hard panel and a perforated panel which act as resonators that can be tuned to attenuate particular frequencies by manipulating the geometry of these elements. 3-D printing is an attractive way to fabricate liner prototypes to be tested in a laboratory environment to aid in concept and model development. This project sought to understand the effects on acoustic impedance resulting from the variability in as-printed prototype liner geometry due to the precision of fused deposition modeling (FDM) 3-D printing. Eleven liner samples were prepared using both variable and fixed layer height FDM processes. The depths of the cavities were measured and were found to be typically precise within a printed sample but sometimes biased away from the intended depth. The samples were tested in a normal incidence tube and impedances were educed. The resonant frequencies of the samples were found to be

systematically below those targeted by around 5%. Also, large variability was observed in acoustic resistances at frequencies much lower than the resonant targets.

10:00

4aSA3. Development of biobased polyurethane acoustic porous material using particulate leaching process. Francoise Cote (Mech. Eng., Polytechnique Montreal, 2500 Chem. de Polytechnique, Montreal, QC H3T 1J4, Canada, francoise.cote@polymtl.ca), Mohamed Amin Ben Lassoued, Edith Roland Fotsing, and Annie Ross (Mech. Eng., Polytechnique Montreal, Montreal, QC, Canada)

Biocomposites are emerging as promising alternatives to petrochemical-derived materials, yet they remain a significant challenge in specific applications such as aerospace, where mechanical and acoustic performance is critical [1–3]. This research examines the incorporation of biosourced components into polyurethane acoustic foams to optimize mechanical and acoustic properties while reducing the isocyanate-to-biobased polyol ratio. A particulate leaching process is employed to control pore size, a key factor influencing both structural stiffness and acoustic absorption [4–7]. Bulk samples with varying isocyanate-to-polyol ratios are produced and characterized using a dynamic mechanical analyzer (DMA) [8]. Then, polyurethane foams are produced using the same ratios, with salt crystals (500–700 μm) to create interconnected pores. These foams are then tested for acoustic absorption using an impedance tube and for compressive modulus through compression tests. Preliminary results indicate that higher isocyanate content enhances stiffness in both bulk samples and foams. For foams with a 50–50 polyol-isocyanate ratio, the storage modulus reaches 1.3 GPa, and the acoustic absorption coefficient ranges from 0.7 to 0.9 in the 3000–5000 Hz range. These findings demonstrate that incorporating up to 50% biosourced polyol can achieve mechanical and acoustic performance comparable to conventional materials, while contributing to environmental sustainability [5].

10:20

4aSA4. Using wavenumber spectra measured on a cyclic plate to deduce the convective velocity of a Corcos turbulent boundary loading. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov) and Ihab F. El-Kady (Sandia National Labs., Albuquerque, NM)

Turbulent boundary layers exert a partially coherent pressure loading upon the surfaces over which they pass. Considering such a surface and the wavenumber spectrum associated with turbulent loading together as a system, we seek to determine the convective velocity of turbulence features passing over the surface based on the vibration measured on the side of the surface opposite the flow. We consider in our model a simulated circumferentially connected Corcos-coherent loading applied to a 2-D semi-periodic plate as a proxy for flow over a cylinder. The wave number space transfer function of the plate is used in connection with the spectrum measured on the interior side as the basis of an inverse problem to determine the convective velocity. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

10:40

4aSA5. Temperature dependence of internal damping in metal Euler beams. Joshua T. Mills (Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, jmills27@byu.edu), Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Micah Shepherd (Brigham Young Univ., Provo, UT)

Zener's model for the internal damping of metals is proportional to the temperature of the specimen. However, most loss factor tables for metals do not include temperature. Precise damping estimates of hot or cold metals would benefit metal constructions operating in extreme environments. To this end, a thermometer is added to an apparatus measuring the internal damping of beams. Aluminum and stainless-steel Euler beams are heated in an oven or cooled with liquid nitrogen and then suspended at nodal lines, removing boundary condition losses. The beams are also suspended in a vacuum chamber with a gauge pressure lower than 1 Torr to remove acoustic radiation losses. A J-type thermocouple is used to monitor the specimen temperature, and beam velocity is recorded at a single location using a laser vibrometer. Loss factors were calculated from this data using the half-power method. Resulting loss factor curves are compared against Zener's model for thermoelasticity. Loss factors have a positive correlation with temperature, which agrees with Zener's model. However, the aluminum specimen follows the model better than the stainless-steel specimen. Possible limitations explaining this discrepancy will be briefly discussed.

Session 4aSC

Speech Communication: Speech Perception Poster Session II

Stefon M. Flego, Chair
English, Virginia Tech, 181 Turner St. NW, Blacksburg, VA 24061

All posters will be on display from 9:00 a.m. to 11:00 a.m. Authors of odd numbered papers will be at their posters from 9:00 a.m. to 10:00 a.m. and authors of even numbered papers will be at their posters from 10:00 a.m. to 11:00 a.m.

Contributed Papers

4aSC1. False hearing in L2-accented sentences. Zach Helfand (Psychol. and Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, zhelfand@gmail.com) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Unfamiliar accents complicate the task of speech perception, leading to reductions in intelligibility and increases in the effort listeners must put forth. One adaptive mechanism people may use to accommodate this challenge is to rely more on top-down processing during comprehension. To test this hypothesis, this study employed a false hearing paradigm (Sommers *et al.*, 2015). Participants listened to sentences, half produced by an American English L1-accented speaker and half by an L2-accented speaker. Each sentence ended with a target word that was either highly predicted by the semantic context (e.g., "The captain steered the SHIP") or was a phonological neighbor of the predicted word (e.g., "The captain steered the SHEEP"). We hypothesized that listeners would rely more on semantic context and less on bottom-up acoustic information for L2- versus L1-accented speech and thus show higher false hearing rates. Preliminary data show this to be the case: in incongruent sentences, young adults showed significantly higher false hearing rates in L2-accented sentences than in L1-accented sentences.

4aSC2. Cross-language mapping using neutral acoustic referents. Kenneth J. de Jong (Dept. of Linguist, Indiana Univ., Ballantine Hall, Bloomington, IN 47405, kdejong@indiana.edu) and Yuka Tashiro (Dept. of Linguist, Indiana Univ., Bloomington, IN)

Research in second language acquisition often relies on constructing a map between two phonological systems. Mapping studies typically use productions from a second language, requiring learners to provide labels in their first language. This study reports a method for determining cross-language maps using a synthetic array of fricative sounds. The fricatives consist of 588 stimuli generated with orthogonal variation of four synthesis parameters which determine the overall spectral coverage and spectral shape of the synthetic fricative noises. Native speakers of English and Japanese engaged with the stimuli in a self-directed search protocol, locating English /ʃ/ in very similar locations, but differing for /f//θ//h/ and /s/. Regression models predicting the probability of stimulus labels find a strong similarity between Japanese labels of English fricatives with Japanese C + V categories: /h/ = /ha/, /s/ = /sɯ/, and /ʃ/ = /çi/, with added effects of the native English category location. Two-category maps also are evident, e.g., /f/ and /h/ both map onto /ha/, and /ʃ/ maps onto /çi/ and /çi/. Overall, the synthetic reference technique revealed merged categories across the languages for learners with a small adjustment for learning the native English category.

4aSC3. Assimilative phonetic parsing as perceptual grouping: The influence of attention. Yuka Tashiro (Linguist, Cognit. Sci., Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 504, Bloomington, IN 47405, yutash@iu.edu)

Phonetic context affects the perception of speech sound in two different ways, either making it more like the context or less like the context, in a dissimilative manner or an assimilative manner. Rysling *et al.* [AP&P 81(4), 1127–1146 (2019)] suggested that assimilation happens when listeners encounter spectral ambiguity, and then group upcoming spectral frequencies with the preceding sound. Following Murgia *et al.* [Vision Res. 126, 69–79 (2016)], which investigated attention in color perception, the current paper studies whether multi-stabile intentional grouping can cause assimilation while stabile unintentional grouping causes dissimilation in speech perception. Stimuli were ɛCV and CV where C is an ambiguous fricative drawn from a [s-ʃ] continuum and V is either [i] or [u]. In one condition, participants were told to focus on the ɛC portion in the ɛCV stimuli to identify the consonant, and in another, the participants were told to focus on the CV portion to identify the consonant and the vowel. In a third condition, the participants heard only CV stimuli and were told to identify the consonant and the vowel. The results indicate that intentional grouping with multi-stability triggers an assimilative parsing, which is consistent with Murgia *et al.*'s claim.

4aSC4. Assessing the relationship between short-term nasalance measures and phonetic transcription. Laura Koenig (Adelphi Univ., 300 George St., New Haven, NY 06511, koenig@haskins.yale.edu), Areti Okalidou (Univ. of Macedonia, Salonika, Greece), and Ariana Sambade (Adelphi Univ., Garden City, NY)

A common question in speech science is how well instrumental measures correspond to listener perception. Here, we evaluate the relationship between nasalance and listeners' transcription of consonants. Nasalance is the ratio between the energy of two microphones separated by a divider at the upper lip. Most studies employing nasalance have obtained measures over long units of speech such as sentences or paragraphs. However, the system used here, the Kay Pentax Nasometer, outputs a nasalance measure every 8 ms, permitting a more fine-grained assessment. We drew on a dataset of Greek-learning children with normal hearing (NH) and cochlear implants (CI) producing words with initial nasals; voiceless stops; and voiced stops, which may be prenasalized in some speakers of Greek. Initial observations revealed cases where the transcription did not correspond as expected to the nasalance signal; for example, a sound transcribed as nasal might show zero nasalance in the closure. Our presentation explores these cases, taking into account the amplitude envelope in two microphone signals along with the nasometer signal. We observe that zero values of nasalance do not necessarily reflect an absence of nasal energy, but nasal energy that is not sufficiently strong or greater than the oral energy.

4aSC5. Integration of syntactic and prosodic boundary cues in second language processing. Jinxin G. Ji (The Hong Kong Polytechnic Univ., Wuhu Residence 19 e, 111 Wuhu St., MN102, Kowloon City, Kowloon, Hong Kong, 23036913r@connect.polyu.hk), Wencui Liu (Tongji Univ., Shanghai, China), Jiaxin Li (The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), Xiaohu Yang (Tongji Univ., Shanghai, China), and Gang Peng (The Hong Kong Polytechnic Univ., Hongkong, Hong Kong)

Recent research has explored factors influencing second language (L2) prosody–syntax integration, such as proficiency, language transfer, and processing strategies. However, it remains unclear whether L2 processing resembles native mechanisms. This study aimed to examine L2 prosody–syntax integration by comparing English sentence processing between native English speakers and Chinese L2 learners, and first language (L1) Chinese and L2 English processing within L2 learners. Eighty-nine participants (30 native speakers, 30 higher-proficiency, and 29 lower-proficiency learners) completed Rapid Prosody Transcription experiments. The strengths of acoustic cues were manipulated across nine steps at syntactically licensed and unlicensed positions. Results showed syntactic constraints and salient acoustic cues independently influenced prosodic boundary judgments in L1 and L2. Compared with native speakers, Chinese L2 learners exhibited a stronger reliance on syntactic information in English. Moreover, L2 learners depended more on L2 syntax than L1 syntactic knowledge, indicating limited transfer from their native language. These findings highlight an interaction between syntax and prosody in L2 processing, consistent with parallel-processing models. The reliance on syntactic knowledge in sentences with unambiguous structures refines the scope of the Shallow Structure Hypothesis, which originally proposes that L2 learners are less sensitive to syntactic constraints and rely more on non-structural information.

4aSC6. Exploring speech learning ability at an advanced age: A comparison of the efficacy of Mandarin tone training with young adults and seniors. Sidsel H. Rasmussen (English, Aarhus Univ., Aarhus, Denmark) and Ocke-Schwen Bohn (English, Aarhus Univ., Ndr Ringgade, Aarhus DK-8000 C, Denmark, ocke.bohn@cc.au.dk)

The present study tests the claim that speech learning ability extends over the entire life span. This claim is well supported for learners up to the age of ca. 35 years, but studies of older learners are very scarce. We compared the efficacy of High Variability Perceptual Training with Mandarin tones for 18 participants each in two age groups, 19–34 and 61–77 years, with the non-tone language Danish as their L1. Results were compared with age-matched control groups ($n = 9$ each). Perceptual accuracy increased significantly for the two age groups of trainees, who participated in 10 training sessions over 3 weeks, between pre-training and post-training. However, the mean increase in identification accuracy was significantly higher for juniors (from 56.2% to 93.6%) than for seniors (from 54% to 74.0%). Whereas all juniors benefited significantly from training, several seniors did not. Participants' identification accuracy before and after training will be compared to individual measures of musical sophistication and pitch discrimination thresholds. Overall, the study suggests that an advanced chronological age does not necessarily affect speech learning ability negatively. [Research supported by a grant from the Independent Research Fund Denmark, grant DFF2 0132-00008B.]

4aSC7. Changing cues? How L2 Korean learners navigate VOT and F0 in Korean stop categorization. Jacob Boudreau, Danica J. Kim, Hyoju Kim, Seoyeon Kim (Linguist, Psychol. and Brain Sci., Univ. of Iowa, Iowa city, IA), and Ethan Kutlu (Linguist, Psychol. and Brain Sci., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com)

English distinguishes stop consonants through a binary voicing contrast, with voiced stops having shorter Voice Onset Time (VOT) than their voiceless counterparts. In contrast, Korean distinguishes stops into three categories: lenis, aspirated, and fortis. In both languages, VOT and the fundamental frequency (F0) of the following vowel are critical cues to distinguish stop contrasts. Korean listeners primarily rely on F0 for Korean stops (particularly when distinguishing lenis from aspirated) whereas English listeners primarily rely on VOT for English stops (Schertz *et al*., 2015). This presents a challenge for English learners of Korean, who must adapt to using different phonetic cues for categorizing ambiguous stops. This study investigates which phonetic cues are most informative to distinguish Korean stops by measuring adult L1 Korean speakers' and L2 Korean learners' speech categorization patterns using a Visual Analog Scaling Task, an updated approach to understanding speech categorization (Apfelbaum *et al*., 2022). Participants' exposure to Korean and English is measured through a Social Network Questionnaire (SNQ). Similar to prior findings (Kutlu *et al*., 2024), L2 listeners who have more exposure to Korean, as measured by their SNQ, are expected to show more gradient categorization in Korean.

4aSC8. How do Koreans judge Korean-accented English? Including the voices of ethnic minorities. Danica J. Kim (Linguist, Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA) and Ethan Kutlu (Linguist, Psychol. and Brain Sci., Univ. of Iowa, 459 Phillips Hall, Iowa City, IA 52242, ethankutlu@gmail.com)

Accents are deeply tied to social identity and serve as a basis for discrimination due to unrealistic societal expectations. They convey socio-indexical information about speakers and reflect social hierarchies, leading listeners to make rapid judgments about unfamiliar accents. Consequently, Englishes that are spoken outside of the inner circles are often labeled as "nonstandard" or "foreign," pressuring individuals with immigrant backgrounds to suppress their identities to conform to dominant norms. Despite the increased awareness and efforts to promote inclusivity in multilingual environments, ethnic minorities remain underexplored in research. To address this gap, this study aims to investigate perceptions of Korean-accented English among Koreans, Korean Americans, and Americans residing either in South Korea or in the United States. Participants listen to spontaneous speech recorded by speakers of English from different language backgrounds including Korean, Turkish, Hindi, and American English. Next, they complete a comprehensibility task, rate the speakers' accentedness on a continuous scale, and respond to a language attitude questionnaire for each recording. This will be coupled with a social network survey and a language background questionnaire to examine the contextual factors influencing their responses. We predict that Korean Americans will hold the most negative perceptions of Korean-accented English due to the internalization of societal stigma.

4aSC9. Perception of Hindi stop voicing and aspiration by native English-speaking learners of Hindi. Sreeparna Sarkar (Univ. of Pennsylvania, 260 South Main St., Newark, DE 19711, sree@udel.edu)

Hindi exhibits a four-way stop contrast, yielding voiceless unaspirated (T), voiced unaspirated (D), voiceless aspirated (TH), and voiced aspirated (DH) stops. Voicing and aspiration as additional properties compared to the (T) stop, affect perceptual accuracy in native Bengali speakers (Sarkar 2023), i.e., (D, DH) > (T, TH) and (TH, DH) > (T, D). The DH stops with both properties have the highest perceptual accuracy and T the lowest. This study tests whether voicing and aspiration as additional properties have the same effect on natively English-speaking learners of Hindi (a related language to Bengali with the same four-way stop contrast), using a listening experiment. This study investigates the perception of these complex stops from a learner's standpoint. Results from data collected from 22 participants show a higher perceptual accuracy percentage of the DH (87%) stops with both voicing and aspiration compared to D (77%), TH (76%), and T with the least, (72%) perceptual accuracy. Based on these results, teaching methods focusing on the learner's ability to perceive and identify the T, D, and TH stops better must be developed because they appear to be more challenging to perceive by the learners, which may lead to further confusion and spelling errors.

4aSC10. Inter-talker second-language (L2) phonetic similarity reflects L2 intelligibility and L1 background. Seung-Eun Kim (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, seungeun.kim@northwestern.edu), Matthew Goldrick, and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

Second-language (L2) speech deviates from L1 speech along multiple acoustic-phonetic dimensions. Free classification tasks have shown that L1 listeners judge similarities (or differences) between L2 speech based on, for instance, a talker's gender, degree of foreign accent, or perceived L1

background. However, the acoustic-phonetic dimensions underlying these judgments remain poorly understood. We examined the perceptual organization of L2 speech using a self-supervised machine learning model trained on a large set of L1 speech. Sentence recordings of 63 L2 English talkers from 5 L1 backgrounds (11–14 talkers/L1, 118–120 sentences/talker) were transformed into multi-dimensional representations by this pre-trained model. Average inter-talker similarity within this multi-dimensional space was significantly related to L2 intelligibility and L1 background. Specifically, pairs of L2 English talkers with higher L2 intelligibility or shared L1 background were represented as more phonetically similar in the space than pairs of L2 talkers with lower intelligibility or different L1 backgrounds. Our investigation thus proposes a novel way of studying the cognitive representations of speech. The application of machine-learning techniques to the representation and classification of speech samples in a pre-trained, self-supervised, high-dimensional representation space opens the possibility of breakthroughs in our understanding of the multitude of acoustic-phonetic dimensions that underlie speech variation.

4aSC11. A perceptual explanation for the adaptation of aspirated stops in Indic languages. Jahnavi Narkar (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, jnarkar@ucla.edu) and Max Meszaros (Linguist, UCLA, Los Angeles, CA)

Indic languages and Indian varieties of English have an adaptation pattern whereby English aspirated stops are adapted as unaspirated, even though Indic languages have contrastive aspirates. We tested the possibility of this unexpected adaptation pattern having a perceptual explanation. Aspirated stops in English and various South Asian languages differ in their acoustic properties (Wiltshire, 2005; Wiltshire and Harnsberger, 2006). Moreover, the acoustic properties of voiceless stops in Indian varieties of English are comparable to those of Indic languages (Narkar and Staroverov, 2022). Specifically, Indic aspirated stops have longer voice onset times (VOT), and the consonant-induced f0 (Cf0) following these stops is lower in Indic languages than in English. We tested English and Hindi listeners' perceptions of 50 tokens on the [ka]-[k^{h}a] continuum in a categorization task. The stimuli were resynthesized by combining 10 VOT values and 5 f0 values from the natural speech of a Marathi (Indic) speaker. Results show that English and Hindi listeners do perceive aspiration differently—English listeners' VOT categorization boundary is lower than Hindi listeners' and English listeners' boundary becomes lower as the f0 increases, whereas f0 has no effect on Hindi listeners' categorization boundary.

4aSC12. Dialect exposure and auditory priming for allophones. Marie Bissell (Dept. of Linguist & TESOL, Univ. of Texas at Arlington, 2948 Garden Bluff Trail, Fort Worth, TX 76118, marie.bissell@gmail.com)

In this study, I examined how listeners' exposure to allophonic variation for two linguistic variables, /ai æ/, affected their lexical decision behaviors in an immediate auditory priming task. I analyzed data from listeners with more and less exposure to allophonic variation for each linguistic variable. I expected that listeners with more exposure to allophonic variation for a linguistic variable would be primed less by prime–target pairs with mismatching allophones than with prime–target pairs with matching allophones as compared to listeners with less exposure to allophonic variation. That is, I predicted that exposure to allophonic variation would influence listeners' lexical representations. A statistical analysis of response times revealed no priming had occurred in this task, potentially related to the nature of the stimuli. I discuss how these results could shape future studies of allophonic priming.

4aSC13. The accent atlas: A geolocation-based assessment of non-native accent familiarity and linguistic diversity. Yuting Gu (Lang. Sci., Univ. of California, Irvine, Social Sci. Plaza B, Irvine, CA 92697, yuting.gu@uci.edu), Seth Cutler (National Inst. of Health, Rochester, NY), Xin Xie (Lang. Sci., Univ. of California, Irvine, Irvine, CA), and Chigusa Kurumada (Brain and Cognit. Sci., Univ. of Rochester, Rochester, NY)

A key to successful comprehension of natural language is the ability to adapt to its variability. Recent studies have suggested that linguistic diversity in daily speech input predicts monolingual native speakers' ability to adapt to an unfamiliar non-native accent. However, self-reports of linguistic diversity can be prone to recall errors and bias. We propose a census-based approach to quantifying the likelihood of non-native accent exposure at both the state and zip code levels. We conducted a large-scale (N = 647) conceptual replication of Clarke and Garrett (2004) to examine monolingual English listeners' rapid adaptation to Chinese-accented speech. Results indicate that the estimated level of accent exposure at the zip code level, but not at the state level, predicts the magnitude of accent adaptation. This suggests that daily accent exposure may vary across local environments and that the current geolocation-based measure of linguistic diversity will be a useful tool to complement self-reports. [Research reported in this work was supported by NIH-NICHD grant R01HD111936.]

4aSC14. Testing the role of L1 influence in phonetic accommodation by Indian English bilinguals. Jupitara Ray (Dept. of Linguist, Boston Univ., 621 Commonwealth Ave. Boston, MA 02215, jupitara@bu.edu) and Charles B. Chang (Dept. of Linguist and Translation, City Univ. of Hong Kong, Hong Kong, Hong Kong)

This study examines whether and how early sequential bilingual speakers of Indian English (IE) accommodate General American English (GAE) in the production of the alveolar lateral /l/. To explore first language (L1) influence on second language (L2) accommodation of IE, we compared Hindi–English bilinguals (HEBs), who consistently produce clear [l], with Telugu–English bilinguals (TEBs), who produce clear [l] word-initially but retroflex [èÙ] word-finally. We hypothesized that HEBs' laterals in IE, particularly in the word-final position, would differ more from GAE's dark laterals than TEBs' laterals, leading to more convergence toward GAE for HEBs. Participants (N = 10) completed a baseline production task and a shadowing task, repeating words spoken by a GAE interlocutor. A 30-ms steady-state interval was annotated for each lateral, with F1 and F2 measured within this interval. At baseline, HEBs' laterals were, as predicted, more distant from GAE laterals than TEBs' laterals. For F1, HEBs showed slight convergence toward GAE for word-initial and word-final /l/, whereas TEBs showed even less. For F2, both groups converged, but TEBs showed greater convergence than HEBs for word-final /l/. Taken together, these findings provide evidence of L1 effects but do not fully support the hypothesis that greater phonetic distance leads to more accommodation.

4aSC15. Perception of novel sounds in the presence of background noise. Michelle Kapolowicz (Commun. Sci. & Disord., Univ. of South Florida, 4202 E Fowler Ave., Tampa, FL 33620-9951, kapolowicz@usf.edu), Vahid Montazeri (Qualcomm Technologies, San Diego, CA), and Peter F. Assmann (Psych., Univ. of Texas at Dallas, Richardson, TX)

In everyday experiences, listeners often need to resolve auditory ambiguity of incoming sounds without prior knowledge of the target sound source. When listeners are exposed to an auditory scene with unfamiliar sounds, target sounds are often identified only when they are repeated across non-repeating competing sounds, supporting the idea that repetition provides a basis for segregating a single source from competing novel sounds (Montazeri *et al.*, 2021; McDermott *et al.*, 2011). These results were based on listeners exposed to a repeating target sound with varying distractor sounds during the exposure (i.e., perceptual learning) stage but tested with the target presented in quiet. Our present study extends this by testing whether listeners' perception of the target sound remains robust when also presented with a novel distractor during the test stage. We found that exposure to a repeating target with either a single distractor or in quiet provided limited benefit for detecting the target with a novel distractor. However, exposure to the target with varying distractors enabled robust recognition against new distractors, suggesting that listeners adapt their expectations to distinguish the target sound only when exposed to varying distractor sounds.

4aSC16. Individual differences in perceptual assimilation patterns across languages. Melissa Baese-Berk (Linguist, Univ. of Chicago, 1115 E 58th St., Rosenwald Hall, Rm. 203, Chicago, IL 60657, mmbb@uchicago.edu), Charlie Nagle, and Shelby Bruun (Univ. of Texas—Austin, Austin, TX)

When listening to an unfamiliar language, listeners perceive speech sounds through the lens of their first language. That is, they tend to

assimilate sounds into their first language categories, especially sounds that are very similar to their first language categories. These patterns of assimilation have been formalized using the Perceptual Assimilation Model (PAM; Best, 1995) and have been tested within individual language pairs. However, while PAM makes predictions about how speech sounds should be assimilated across many languages, it does not make strong predictions about potential individual variability in assimilation patterns. In the present study, we ask how native English listeners assimilate stop consonants from Thai, Korean, and Spanish into their L1 categories. Each listener was asked to assimilate sounds from continua in each of the three aforementioned languages into existing English categories (e.g., /p/ and /b/) and rate the goodness of fit for each sound as an exemplar of that category. These data allow us to investigate three questions. First, how does assimilation compare across different target languages within the same listener population. That is, are patterns similar for Thai and Korean stops? Second, is there individual variability among listeners in assimilation patterns within or across languages? Third, do linguistics or other cognitive factors of the listeners impact assimilation patterns within or across languages?

4aSC17. Training type and composition moderate efficacy of perceptual learning for speech. Alexandra M. Kapadia (Linguist, Univ. of Chicago, 1115 E 58th St., Rm. 203, Chicago, IL 60637, alexandrakapadia@uchicago.edu), Carissa A. Diantoro (Linguist, Univ. of Oregon, Eugene, OR), Santiago Jaramillo (Univ. of Oregon, Eugene, OR), and Melissa Baese-Berk (Linguist, Univ. of Chicago, Chicago, IL)

Perceptual learning for novel speech sounds can occur via active training, in which a discrimination task is coupled with feedback. Although effective, active training is significantly more effortful than passive exposure to the target sounds. While passive exposure alone is insufficient for learning, a combination of active and passive training has been shown to be just as beneficial for perceptual learning gains (Wright *et al*., 2015). However, the underlying mechanisms at play, and therefore the ratio and order of active and passive training required to achieve learning, remain unclear. In the current study, participants completed a training paradigm to discriminate a dental–retroflex stop contrast. Participants were assigned to one of 8 conditions ($n = 20$ per condition) varying by the amount of training type, whether mixed training was blocked or interleaved, and training type order, and whether passive training was a nonlinguistic task or task-free. Preliminary results demonstrate learning in both mixed training conditions (blocked and interleaved) but only when active training came before passive, and opposite effects of passive training type (nonlinguistic task versus task-free) depending on mixed training organization. While data analysis is ongoing, results suggest a complex interplay between attention to target stimuli and the organization of other elements of training.

4aSC18. Social network indices impact spoken word recognition across the adult lifespan. Sarah E. Colby (Linguist, Univ. of Ottawa, Hamelin Hall, 70 Laurier Ave. E., Ottawa, ON K1N 6N7, Canada, scolby@uottawa.ca), Ethan Kutlu (Linguist, Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), Samarium Knight, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Social engagement is critical for cognitive well-being in older adulthood, but little is known about the relationship between the mechanisms of language processing and social networks as listeners age. In young, normal-hearing listeners diverse social networks are known to support more flexible speech processing (Kutlu *et al*., 2024). The competition dynamics of word recognition change with age, even in people with normal hearing (Colby and McMurray, 2023). Thus, we ask whether large, diverse social networks can bolster language processing in older adulthood. The current study tested a large group of adults ($N = 76$, 30–80 years old) on a Visual World Paradigm task to assess the real-time competition dynamics underlying spoken word recognition and a Social Network Questionnaire as a measure of the quality and quantity of their social engagement. Data collection is ongoing, but preliminary analyses suggest listeners with more dense social networks activate words faster, and those who regularly converse with more individuals better manage competition between words. This work confirms the importance of maintaining social bonds in older adulthood.

4aSC19. Phonetic learning can transfer between bilingual speakers' languages, despite cross-language phonetic mismatch. Yevgeniy Melguy (Linguist, Univ. of Chicago, 1115 E. 58th St., Rosenwald Hall, Rm. 203, Chicago, IL 60637, ymelguy@uchicago.edu), Arthur Samuel (Psych., Stony Brook Univ., Donostia - San Sebastián, Gipuzkoa, Spain), and Clara Martin (Basque Ctr. on Cognition, Brain and Lang., Donostia - San Sebastián, Gipuzkoa, Spain)

Listeners exposed to an unfamiliar accent can rapidly adjust their sound category representations in response, a phenomenon known as *phonetic recalibration*. When exposed to words where a sound category has been altered to be phonetically ambiguous (e.g., super→ [ʃ/s]uper), listeners can expand the category to accommodate this new pronunciation. We investigated the mechanisms underlying such learning in Basque-English bilinguals ($N = 38$). While both Basque and English contrast alveo-palatal and alveolar fricatives (/ʃ/ versus /s/), English and Basque /s/ are articulatorily and acoustically distinct. To test how phonetic similarity constrains the transfer of learning within and across languages, we exposed listeners to Basque words produced with an unfamiliar accent where either /s/ (session 1) or /ʃ/ (session 2) were replaced with ambiguous [ʃ/s]. We then assessed learning in both Basque and English via a standard categorization task that involved phonetic continua between critical fricatives. Results show listeners recalibrated their Basque /ʃ/-/s/ boundary but failed to generalize learning to other Basque fricative contrasts. However, there was a generalization of learning to the English /ʃ/-/s/ contrast, despite cross-language phonetic mismatch. This suggests that similar cross-language sound categories may be linked, despite production differences between languages. Crucially, however, such learning effects were only consistently present within the most ambiguous portion of each phonetic continuum.

4aSC20. Perceptions of talker attitude from trusting and doubting prosody: Affective dimensions and acoustic correlates. Abbey L. Thomas (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., Minneapolis, MN 55455, abbeyt@umn.edu)

This study examined how 86 listeners perceived the talker attitude from 714 tokens produced with neutral, doubting/disbelieving, or trusting/believing prosody. Within-subject acoustic analysis of these tokens revealed that, compared to neutral prosody, doubting prosody featured a slower speech rate and trusting prosody featured a higher utterance mean F0. Listeners responded to tokens by selecting at least two attitudes/emotions from a grid of 20 labels to describe the talker's affect. Relationships between recordings and selected attitudes/emotions were analyzed with nonparametric multivariate methods. Although the labels "disbelieving" and "trusting" were selected infrequently, distinct clusters of labels were used in response to tokens produced with each attitude. Doubting tokens were often described with negative-valence labels (e.g., "annoyed"), while trusting tokens were assigned positive-valence labels ("kind"). In contrast, neutral tokens were paired with low-arousal terms like "calm," "sad," or "neutral." Positive-valence labels were attached to tokens featuring a higher F0 and faster speech rate, while negative-valence labels described tokens with a slower speech rate and a rising F0 slope. These results suggest that rather than maintaining a one-to-one mapping of categorically distinct pitch contours or speaking rates to affects, listeners may use prosodic cues to sort tokens along basic affective dimensions, like valence and arousal.

4aSC21. Stressing the suprasegmentals: L2 phonetic training of Spanish stress. Corey McCulloch (Linguist, Univ. of Kansas, Blake Hall, 1541 Lilac Ln., Lawrence, KS 66045, corey.mcculloch@ku.edu) and Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS)

Second language (L2) learners from first languages (L1) which do not use stress contrastively often face difficulty in perceiving stress in the L2. While English and Spanish both use stress contrastively, L2 Spanish stress is often still difficult to acquire. Spanish stress is realized purely suprasegmentally, whereas suprasegmental cues play a diminished role in English compared to spectral cues, leading to difficulties in L2 acquisition. Phonetic training (PT) is effective for improving the perception of L2 phones but has often focused on segmental or tonal contrasts specifically. This study investigates whether the perception of Spanish stress may be improved through PT. The Automatic Selective Perception model (Strange, 2011) discusses the importance of attention in the learning of L2 sounds, though few studies have examined its role during PT for suprasegmentals. This study investigates this as well, clarifying the role of attentional focus in stress acquisition. 60 L2 Spanish learners will receive 4 days of PT, with pre- and post-training performance measured through a sequence recall task, two discrimination tasks, and an identification task. Our results will help clarify whether focusing attention on suprasegmentals is effective in improving stress perception for L1–L2 pairs which feature differing phonetic instantiations of contrastive stress.

4aSC22. Surprisal in non-native speech sound learning: A replication and extension of Nixon (2020) with Japanese vowels, Mandarin fricatives/affricates, and Southern-Min tones. Adam A. Bramlett (Second Lang. Acquisition, Carnegie Mellon Univ., 1314 Fox Hunt Dr., Cheswick, PA 15024, abramlet@andrew.cmu.edu)

Non-native speech acquisition involves more than statistical tracking of speech cues. This study explores how surprisal (i.e., prediction-error) drives speech sound acquisition, replicating and extending Nixon's (2020) error-driven learning framework. We examine surprisal in the learning of Southern-Min tone contrasts and extend the experiment to Japanese vowel length and Mandarin fricatives and affricates, representing both suprasegmental and segmental contrasts. A total of 146 native English-speaking participants (51 learning Japanese, 41 Mandarin, 54 Southern-Min) participated in one of two stimuli presentation orders: discriminative (speech-sound then image) or non-discriminative (image then speech-sound). This study makes two key contributions: First, it replicates Nixon's (2020) framework by expanding the generalizability of error-driven learning across different speech contrasts. Second, it includes a methodological extension by incorporating eye-tracking to operationalize surprisal, providing real-time insights into cognitive processes during training. Results show that surprisal, particularly in low-frequency items, facilitates learning in the discriminative condition. These findings challenge the view that speech-sound learning is purely statistical and suggest that eliciting surprisal enhances the learning of challenging speech contrasts, such as Japanese vowel length, Mandarin fricatives, and Southern-Min tones, with broader implications for the acquisition of non-native speech sounds (Burnham, 2013; Kim, 2023).

THURSDAY MORNING, 22 MAY 2025 — STUDIOS 7/8, 7:55 A.M. TO 11:00 A.M.

Session 4aSPa

Signal Processing in Acoustics and Underwater Acoustics: Universal and Doubly Adaptive Methods for Signal Processing I

John R. Buck, Cochair
Electrical and Computer Engineering, UMass Dartmouth, 285 Old Westport Road, Dartmouth, MA 02747

Kathleen E. Wage, Cochair
George Mason University, 4400 University Drive, Fairfax, VA 22151

Andrew C. Singer, Cochair
Stony Brook University, Stony Brook, NY

Chair's Introduction—7:55

Invited Papers

8:00

4aSPa1. Multiple and hierarchical universality. Meir Feder (EE, Tel Aviv Univ., Dept. of EE-Systems, Ramat Aviv, Tel-Aviv, N/A 69978, Israel, meir@tauex.tau.ac.il)

Universal methods have been suggested in acoustics for many signal processing applications, where the acoustical channel parameters, the operating parameters and so on are unknown. This approach is based on results from information theory. The recent advances in applying huge models for such problems pose a challenge to understand and control how "over-parameterized" model can indeed explain the data and generalize usefully. The multiple universality proposed solution and explanation describes the huge parametric model as a union of subsets of models and various complexities. Sometimes this construction is done in a hierarchical way. The learning or estimation task is not only to find the parameters but also the right subset of explanations. As in many universal solutions, I'll show how a mixture or ensemble approach over the various possible complexities not only explains the success of the recent huge models but actually leads to better performance.

8:20

4aSPa2. An experimental comparison of universal beamformers. Jeff Tucker (George Mason Univ., 1511 Black Eyed Susan Ln., Vienna, VA 22182, jtucke16@gmu.edu) and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

The challenge of rejecting interference and noise to reveal a signal of interest is common in array processing. Buck and Singer introduced the Universal Dominant Mode Rejection (UDMR) beamformer, which calculates filter weights by blending multiple dominant mode rejection (DMR) beamformers with different subspace dimensions [IEEE SAM, 2018]. Tucker and Wage's Performance Weighted Blended (PWB) beamformer adopts a similar strategy, combining classical estimators with different tapers [IEEE SAM, 2024]. While the UDMR beamformer requires a computationally expensive eigen-decomposition, the PWB beamformer can be implemented more efficiently using fast Fourier transforms. UDMR is doubly adaptive, adjusting its blend weights to combine adaptive DMR beamformers. In contrast, PWB is singly adaptive, blending classical fixed-taper beamformers. Since UDMR blends adaptive beamformers, it can provide better interference suppression. However, PWB's computational efficiency may be desirable for resource-limited applications. This talk presents an experimental comparison of the two algorithms using the PhilSea10 dataset [Worcester *et al*., JASA (2013)], benchmarking both against fixed rank DMR beamformers. [Work supported by ONR.]

8:40

4aSPa3. Low-complexity adaptive beamforming for acoustic pulse-echo imaging. Håvard Kjellmo Arnestad (Dept. of Informatics, Univ. of Oslo, Gaustadalléen 23B, Oslo 0373, Norway, haavaarn@ifi.uio.no), Sven Peter Näsholm, Ole M. Rindal, and Andreas Austeng (Dept. of Informatics, Univ. of Oslo, Oslo, Norway)

For years, the research community has viewed adaptive beamforming as a promising approach to enhance image quality in pulse-echo imaging applications such as medical ultrasound and sonar. Methods like the minimum-variance beamformer offer the potential for improved resolution and contrast, enabling sharper representation of key features in the image. However, a significant challenge lies in the high computational cost, stemming from the construction and manipulation of the spatial covariance matrix, which is typically calculated on a per-pixel basis. To address this, several low-complexity methods have been developed that retain the essential benefits of the minimum-variance beamformer while achieving computational efficiency suitable for real-time applications. A fundamental aspect is how to universally apply such beamformers, as the pulse-echo nature allows double adaptiveness over the transmit and/or receive beamforming steps. This presentation provides an overview of the theory and practical applications of these approaches, drawing on over 15 years of research at the University of Oslo and elsewhere. We start with the foundational principles of the minimum-variance beamformer, discuss the motivation for transitioning to low-complexity adaptive beamforming, assess the performance in real-world scenarios, and conclude with recent findings and future directions for this field.

9:00

4aSPa4. Universal switching adaptive beamforming. Manan Mittal (Elec. and Comput. Eng., Stony Brook Univ., 1308 W Main St., #119, Urbana, IL 61801, mittalmanan99@gmail.com), Yongjie Zhuang (College of Eng. and Appl. Sci., Stony Brook Univ., Stony Brook, NY), Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, Chicago, IL), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), and Andrew C. Singer (College of Eng. and Appl. Sci., Stony Brook Univ., Stony Brook, NY)

An adaptive beamformer may be thought of as trading white noise gain for interferer suppression. The beamformer can respond to changing environmental statistics through updates to the sample covariance matrix. In time-varying environments, adaptive beamformers are frequently used with pre-determined sliding windows or forgetting factors for such sample covariance estimation. Thus, an adaptive beamformer must *a priori* select the regions over which the data are assumed stationary. Such methods perform poorly when the environment suddenly changes, such as strong interferers entering or exiting the acoustic scene. Many real-world environments have intermittent interferers, and such a beamformer may waste degrees of freedom suppressing an interferer that is no longer active or neglecting to suppress one that is. We propose the use of universal methods over a class of time-partitioned beamformers. While there are an exponential number of possible partitions of a block of data into locally stationary regions, methods from universal data compression and prediction for piece-wise stationary sources provide a path for implicitly implementing, and mixing over them all, with only polynomial complexity. We employ a linear transition diagram from this literature to enable efficient performance-weighted mixing of beamformers of all possible such partitions.

9:20–9:40 Break

9:40

4aSPa5. A universal beamformer that allocates degrees of freedom between adaptive and conventional beamformers. Savas Erdim (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, serdim@umassd.edu) and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

In dynamic environments, fast-moving interferers traverse resolution cells quickly, challenging the minimum variance distortionless response (MVDR) beamformer to place accurate notches in the direction of interferers. The hybrid MVDR (HMVDR) beamformer generates serendipitous wide notches by factoring the beampattern into adaptive and fixed components. The adaptive MVDR factor implicitly places notches in the interferer direction for suppression and reduces some background noise, while the fixed conventional beamformer (CBF) factor improves white noise gain. The HMVDR product beampattern flexibly allocates degrees of freedom (DoFs) between the adaptive and fixed components. However, optimizing DoF allocation for the adaptive part remains a challenge, especially in environments with an unknown time-varying number of interferers. To address this challenge, we propose the universal HMVDR (UHMVDR) beamformer, which integrates a mixture of experts' universal algorithms into the HMVDR beamformer, blending different factorizations to dynamically allocate DoFs between the adaptive and fixed components. Simulations and microphone array experiments confirm that the UHMVDR beamformer achieves better array output power and white noise gain than competing beamformers. [Work supported by ONR 321US.]

10:00

4aSPa6. A score function for adapting large array partitions for source detection. Yongjie Zhuang (Stony Brook Univ. & Amazon (This work was done prior to joining Amazon), Light Eng., Stony Brook, NY 11790, yongjie.zg@gmail.com), Manan Mittal, Ningyuan Yang (Stony Brook Univ., Stony Brook, NY), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

Large aperture arrays improve detection performance with higher gain, especially in low signal-to-noise ratio (SNR) applications such as underwater acoustic (UWA) source detection. However, large arrays are susceptible to phase errors because of limited spatial coherence of signals or a mismatch between the assumed signal models and the true signal models. To mitigate this issue, the array can be partitioned into smaller segments of sensors known as subapertures. The subapertures are processed coherently, and then the power outputs of the subapertures are combined. This processing is the spatial analog of the classic Welch power spectrum estimator which averages periodograms across time windows of a recording. Identifying the subaperture size which optimizes detection performance remains an open problem. We proposed a score function that indicates the detection performance of different array partitions without access to the ground truth. Numerical experiments using the SWellEx-96 data corrupted by additional noise show that the subaperture maximizing this detection score function achieves a better receiver operating characteristic (ROC) in low SNR cases compared to any fixed array partition candidate. [Work supported by ONR Code 321US.]

10:20

4aSPa7. Space-partitioned adaptive beamforming using context trees. Manan Mittal (Elec. and Comput. Eng., Stony Brook Univ., 1308 W Main St., #119, Urbana, IL 61801, mittalmanan99@gmail.com), Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, Chicago, IL), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), and Andrew C. Singer (College of Eng. and Appl. Sci., Stony Brook Univ., Stony Brook, NY)

An adaptive beamformer suppresses interferers and provides spatial filtering gains by making use of the sample covariance matrix. Updates to the sample covariance matrix reflect changes in the environment to which the beamformer must adapt. In environments with intermittent interferers, it is beneficial to remember the “state” that represents a specific pattern of interferer activity. In such cases, an adaptive beamformer that simply averages all the snapshots may result in reduced performance (with respect to an omniscient, context-aware beamformer that is aware of the interferer state) as the beamformer wastes degrees of freedom suppressing interferers that are always not active. By using the directional cosine of the peak of the beamformer scanned response as an information-bearing sequence, we partition the space into angular sectors that represent beamformers averaging a different set of snapshots. To represent and efficiently mix the output of all beamformers represented by such partitions, we employ a context tree that has been previously used for data compression and piecewise linear prediction. We use the context tree to achieve the signal estimation error of the best piecewise adaptive beamformer that can choose the partition of the directional cosine space.

10:40

4aSPa8. On universal diagonally loaded beamformer under steering vector mismatch. Akshay S. Bondre (Wireless Connectivity Group, Apple, Tempe, AZ) and Christ D. Richmond (Elec. and Comput. Eng., Duke Univ., Rm. 327, Gross Hall, Box 90984, Durham, NC 27708, christ.richmond@duke.edu)

We explore universal adaptive beamforming (UABF) under the non-ideal conditions of steering vector mismatch and signal contamination of data. We focus on the optimal choice of diagonal loading level as the regularization parameter. Assuming a statistical characterization for the steering vector mismatch, we use the average output signal-to-interference plus noise ratio (SINR) loss that accounts for signal mismatch as the loss function metric used by the UABF to determine the best coefficients for linear combination in the formation of the UABF weight vector. The SINR loss metric requires an accurate estimate of the interference-only covariance matrix in practice. Our initial goal here, however, is to focus on the ability of the UABF to choose good coefficients for linear combinations that provide improved performance with respect to the choice of diagonal loading level. Thus, we consider a simpler version of the universal beamformer, where it is assumed that interference-only training samples are available to compute this loss function metric. Numerical results showing the usefulness of this simplified beamformer will be presented that provide insight and motivation to consider more practical versions of the universal beamformer where signal-free training data may not be as available.

Session 4aSPb

Signal Processing in Acoustics, Acoustical Oceanography and Computational Acoustics: Machine Learning in Underwater Acoustics II

Kendal Leftwich, Cochair
Physics, University of New Orleans, 1021 Science Building, New Orleans, LA 70148

Youngmin Choo, Cochair
Sejong University, 209 Neungdong-ro, Gwangin-gu, Seoul 05006, Korea

Shaun Pies, Cochair
Physics, University of New Orleans, 2000 Lakeshore Drive, New Orleans, LA 70148

Contributed Papers

8:20

4aSPb1. Automatic detection and range estimation using a passive acoustic monitoring sensor network in a stratified and range-dependent waveguide. Mark Goldwater (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., M.S. 55, Woods Hole, MA 02543, mgoldwater@whoi.edu), Daniel P. Zitterbart, and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

In shallow-water environments, the sound generated by low-frequency sources interacts significantly with the sea surface and seabed, inducing a strong dependence on source range in the received pressure field due to modal dispersion. As a result, these signals are ideal for range-based localization. However, in complex environments that are range dependent or that exhibit stratified water column and/or seabed properties, inverting the received signal for range in real time is nontrivial. Here, we develop a deep-learning approach to detect and range received signals that accounts for the distributed nature of the deployed sensor network and the stratified nature of the environment which varies with spatial position. The model is trained using simulated data generated based on the parameter distributions observed in the target environment. The detection and ranging method is validated using both simulated and experimental marine data. [Work supported by NDSEG and ONR.]

8:40

4aSPb2. Source localization with temperature variation in a laboratory tank. Natalie Bickmore (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, njbickmore@gmail.com), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Corey E. Dobbs (Brigham Young Univ., Provo, UT)

Environmental variability complicates deep learning applications to ocean acoustics. Transfer learning has the potential to improve the deep learning model's ability to perform in a new environment. This work illustrates the potential of transfer learning for the case of predicting source-receiver range in a laboratory tank with changing water temperature. A four-layer convolutional neural network is trained to predict the source-receiver range. The training dataset consists of time-averaged spectral levels from chirps (50–100 kHz) measured at room temperature. Due to time-dependent ambient noise, data measured on multiple days are combined to form training datasets with 1806 samples. Early stopping is employed, and the trained models are validated on test datasets also measured in room temperature water. The trained models are then applied to data measured in warmer water, and the error in the predicted ranges increases. Transfer learning is performed using a dataset with 50 samples measured at warmer temperatures, and new testing is performed. Transfer learning refines the trained model and leads to reduced errors in the predictions. This work provides an example of how transfer learning could be applied to accommodate environmental variation in ocean applications of deep learning. [Work supported by ONR.]

9:00

4aSPb3. Online multi-target acoustic tracking for mobile assets. Joseph Walker (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, jlw222@ucsd.edu), Zheng Zeng (Dept. of Elec. and Comput. Eng., Univ. of California San Diego, San Diego, CA), Bruce Thayre, Sean Wiggins, and Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Passive acoustic sensing is increasingly used for environmental and situational awareness in autonomous maritime systems. However, real-time processing on resource-constrained edge devices poses significant challenges. To address this, we propose a lightweight, automated multi-target tracking system optimized for low latency (~1.7 ms) and low power (~1.3 W) on a Raspberry Pi Zero 2W. Our method employs a hybrid cluster-filter approach, combining DBSCAN clustering for estimating the number of sources and Kalman filtering for continuous tracking of targets in terms of their direction rather than absolute position. We demonstrate the system's effectiveness using passive acoustic recordings of diving beaked whales in open ocean conditions. Neural networks with post-training optimization are integrated to enhance performance while minimizing computational demands. Designed for cross-platform flexibility, this system can operate on mobile assets like gliders, drifters, and profilers. Preliminary results highlight its potential for real-time autonomous underwater applications, offering a robust, energy-efficient alternative to existing multi-target tracking implementations.

9:20

4aSPb4. Machine learning localization of ice fracture events from 3-axis geophone data from using a physics-based feature space. Erin M. Fischell (JPAnalytics, LLC, 82 Technol. Park Dr., East Falmouth, MA 02536, efischell@acbotics.com), Nicholas R. Rypkema (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Oscar Viquez (JPAnalytics, LLC, East Falmouth, MA)

Three-axis geophones were deployed in an array for more than a month as a part of the 2021 Sea Ice Dynamics Experiment (SIDEx'21) with the objective of classifying and localizing flexural ice crack events. However, using conventional geophone localization methods requires detailed knowledge of frequency-dependent and time-dependent ice characteristics that were inadequately characterized by calibration data from the beginning of

the deployment. As a result, we instead used a localization method with a physics-based feature space and machine learning. First, an observation feature vector is calculated from field or simulation data time series consisting of what we term delay angle (a time delay of arrival metric that abstracts out group speed), amplitude ratio (an amplitude metric that abstracts out attenuation and source level), and arrival angle (based on longitudinal/transverse wave relationships). To localize field data, OASES OASP was first used to generate training, test and evaluation time series datasets for a variety of environments and source locations; Monte Carlo was used to introduce noise into the traces, and feature vectors were extracted. The resulting SVM model is then used to estimate the location of each real-world event time series, using the calibration dataset for real-world performance evaluation. [Work funded by ONR.]

9:40–10:00 Break

10:00

4aSPb5. Analyzing seasonally dependent performance of rain detection from underwater acoustic power spectral density. James Bourgeois (Elec. and Comput. Eng., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, jbourgeois@umassd.edu), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), and Amit Tandon (Mech. Eng., UMass Dartmouth, Dartmouth, MA)

Rainfall and other natural processes can be empirically monitored by analyzing characteristic spectral features in the ocean's ambient sound. Previous work to detect and estimate rainfall from passive underwater acoustics used linear transformations of these spectral features; Ma and Nystuen (2005) measured acoustic power at a few narrowband frequencies, later extended by Mallary *et al*. (2023) and Berg (2023) to principal component analysis (PCA), which represents broadband spectra with a small number of linear coefficients. This research proposes a new broadband detection scheme that constructs separate PCA subspaces by rain and season. Separating the linear transformations before training produces subspaces more finely tuned for each class. PSDs are computed using Welch's method and are separated into dry ($<$2.4 mm/h) and rainy ($>$2.4 mm/h) recordings for each season. A linear dimension reduction matrix is defined for the dry PSDs of each season using eigenvectors of their covariance matrix (the principal components), while preserving over 98% of variance. Rainfall can then be detected using a likelihood ratio test of dimension-reduced PSDs for each season. Performance varies substantially by season and wind conditions, with detection ranging from 30% to 80% seasonally at a 1% false alarm rate. Accounting for wind may improve rainfall detection and, more generally, monitoring of natural processes from underwater acoustic recordings. [Work supported by SMART Scholarship and ONR/MUST.]

10:20

4aSPb6. Acoustic multipath characterization leveraging machine learning techniques for bottom and surface bounce interactions in flat, parallel free surfaces. Grant Eastland (Test & Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

This research explores the characterization of acoustic multipath phenomena involving bottom and surface bounce interactions in environments characterized by flat, parallel free surfaces. Propagation paths are derived using the Method of Images, implementing a geometric construction framework referenced to the sea surface via the principles of analytic geometry. A comprehensive model for path length determination is developed, incorporating dependencies on the source, receiver, water column depth, and the number of interfacial interactions. Machine learning techniques, particularly supervised learning algorithms, are integrated to enhance the prediction and analysis of multipath modes. These algorithms facilitate the identification and selection of optimal source and receiver combinations, ensuring precise determination of path lengths for potential multipath modes. The incorporation of machine learning significantly improves the accuracy and efficiency of the multipath characterization process, providing a robust framework for acoustic signal propagation analysis in marine environments.

10:40

4aSPb7. Modulation and coding scheme mode selection based on machine learning in underwater acoustic channels. Hyun-Woo Jeong (Radio Commun. Eng., Korea Maritime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Pusan KS012, Korea, gusdn0930@g.kmou.ac.kr), Yegwon Hong, JaeHun Lee, and Ji-Won Jung (Radio Commun. Eng., Korea Maritime and Ocean Univ., Busan, Korea)

This paper proposes a machine learning-based optimal modulation and coding scheme (MCS) mode selection in underwater acoustic communication (UAC) channels. MCS is an efficient method for improving the system efficiency by changing transmission parameters according to channel conditions in UAC channels. As a UAC channel has time-varying characteristics such as intersymbol interference and Doppler frequency shift, it is impossible for a UAC system to overcome with a large variety of communication impairments well by only using fixed MCS. Thus, this paper proposes a machine learning-based approach for optimal MCS mode selection by utilizing various channel state information (CSI) such as received signal-to-noise ratio, Doppler effects, and multipath propagation. CSI datasets of various UAC channels are generated through simulation and are subsequently used to train machine learning models to predict the optimal MCS mode for varying channel conditions. In time-varying underwater channels, we proved the proposed machine learning-based MCS selection method is better than conventional fixed table-based methods in aspect to transmission efficiency and reliability. [This research is supported by a KRIT grant funded by the Korea government (DAPA) (KRIT-CT-23-035-01, Multi AUV operation Technology for Mine Detection ('23–'28))]

Session 4aUW

Underwater Acoustics, Acoustical Oceanography and Signal Processing in Acoustics: Directional Sensing: Applications and Methods I

Kaustubha Raghukumar, Cochair
Integral Consulting Inc., 200 Washington Street, Suite 201, Santa Cruz, CA 95060

Aaron M. Thode, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Drive, MC 0206, La Jolla, CA 92093

Kerri D. Seger, Cochair
Applied Ocean Sciences, 11006 Clara Barton Drive, Fairfax Station, VA 22039

Ludovic Tenorio, Cochair
CIMAS/NOAA, 75 Virginia Beach Drive, Key Biscayne, FL 33149

Contributed Papers

7:00

4aUW1. Maximum likelihood matched field source range and depth localization with a single acoustic vector sensor. Ivars P. Kirsteins (NUWCDIVNPT, 1176 Howell St., Newport, RI 02813, i.kirsteins@gmail.com)

Motivated by the problem of localizing a source in range and depth in an ocean waveguide using a single acoustic vector sensor with pressure, horizontal, and vertical particle velocity channels, we derive and examine the optimum maximum likelihood-based matched field processors when the signal waveform emitted by the source is unknown. In our derivations, the ocean waveguide parameters are assumed to be known so that pressure and particle velocity replicas can be calculated and that the ambient noise is Gaussian distributed. Two different signal cases are considered: (1) the signal emitted by the source is unknown but deterministic and (2) the source signal is a Gaussian stationary random process with an unknown power spectrum. These two assumptions lead to estimator architectures with totally different invariance properties to signal and noise levels that we will discuss. We also present some propagation-model-based simulation examples that are compared to the Cramer–Rao lower bounds.

7:20

4aUW2. Multi-resonant sub-wavelength acoustic vector sensor for underwater environments. Justin Ivancic (Phys., U.S. Naval Acad., Chauvenet Hall, 572C Holloway Rd., M.S. 9E, Annapolis, MD 21402-5002, ivancic@usna.edu) and Fabio Alves (Phys., Naval Postgrad. School, Monterey, CA)

This paper reports on a multi-resonant micro-electromechanical systems (MEMS) based acoustic vector sensor (AVS) capable of determining the direction of arrival (DOA) of various underwater sound sources. The AVS consists of two MEMS sensors aligned orthogonally and an omni-directional hydrophone. Signals from the sensors are directed to a microprocessor which conditions the data and calculates DOA. The AVS provides unambiguous 360° coverage, calculated in real time. This paper follows up on research into single-resonant sensor designs. This multi-resonant design represents important changes in sensor design characteristics and offers significant improvements in signal-to-noise ratio (SNR), bandwidth, and provides lower frequency detection capability than the single-resonant design. Data for this research were collected at a Navy facility designed for underwater acoustic experimentation. It minimizes underwater acoustic reflections. In the experiment, a stationary underwater acoustic source projected multiple acoustic signals while the AVS was rotated to adjust the DOA of the sound. The DOA accuracy of the AVS was measured over a 360° rotation for various sound sources. The average DOA error over a full rotation was measured to be less than 4.5°. These results indicate that this MEMS-based AVS is very attractive for many naval and other underwater acoustics applications.

7:40

4aUW3. 3-D printing of architected hydrophones. Victor Couedel (Univ. of California, Berkeley, Hearst Memorial Mining Bldg., Rm. 150, Berkeley, CA 94720, vcouedel@berkeley.edu), Haotian Lu, Jiayan Zhang, and Xiaoyu Zheng (Univ. of California, Berkeley, Berkeley, CA)

Piezoelectric hydrophones play a key role in underwater applications such as communication and seafloor mapping. Conventional ceramic manufacturing techniques constrain hydrophone designs to simple geometries, limiting their sensitivity, directivity, and frequency bandwidth. This work presents a novel class of high-performance 3-D-printed piezoelectric hydrophones with rationally designed micro-architectures. By employing a high-resolution light-based printing technique combined with optimized liquid-phase sintering, the piezoelectric coefficients and electromechanical coupling factor achieve 92% and 85%, respectively, of the values seen in pristine materials. A systematic framework was developed to tune piezoelectric properties by adjusting the spatial arrangement and thickness of unit cell struts, resulting in hydrophones with 15 dB higher sensitivity compared to commercial counterparts and customizable directivity patterns. This allows for Direction of Arrival (DoA) estimation without the need for hydrophone arrays and achieves a hydrostatic figure of merit up to five times higher than that of commercial hydrophones. Initial machine learning studies for sub-hydrostatic frequency DoA estimation also show promising results, specifically when it comes to the resilience of the device to noise or acoustic crosstalk—the latter even becomes an advantage for our device whereas it is rather a source of confusion for traditional transducers arrays.

8:00

4aUW4. Numerical modeling of sound source depth estimation by single receiver in shallow water. Sergey A. Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst. of RAS, Moscow, Russian Federation), and Alexey Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

The holographic method for estimating the depth of a low-frequency broadband source using a single receiver in shallow water is developed in this paper. By using broadband signals from the receiver, the source interferogram (sound intensity distributions) is generated in the frequency-time domain. Within the framework of holographic signal processing (S. Pereselkov and V. Kuz'kin, JASA 151 (2), 666–676), the interferogram is analyzed using a two-dimensional Fourier transform (2D-FT). The result of the 2D-FT is referred to as the Fourier hologram (source hologram). Using the inverse 2D-FT, the processed source interferogram, containing information about mode amplitudes, is reconstructed. The ratio of neighboring mode amplitudes is used to estimate the source depth. The results of a numerical experiment on source depth estimation using the holographic method in the low-frequency band (100–300 Hz) are analyzed. The stability of the holographic method to errors in measuring mode amplitudes and variations in waveguide parameters is considered. It is shown that the error in source depth estimation tends to stabilize as the noise level increases. A qualitative and quantitative explanation of the simulation results is provided in the paper. [This research was supported by a grant from the Russian Science Foundation (Grant No. 23-61-10024)]

Invited Paper

8:20

4aUW5. Directional acoustic sensing. Corey Bachand (BTech Acoust. LLC, Fall River, MA) and David A. Brown (ECE/CIE, Univ. Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsdb@gmail.com)

Directional acoustic sensing for underwater acoustics applications such as communications and ambient noise monitoring is receiving increased attention. This paper summarizes several approaches we have developed and tested including multimode cylindrical and spherical hydrophones, sparse arrays, compact self-baffled arrays, USBLs, directional acoustic motion sensors (aka vector sensors), and spiral wave beacons. Performance and tradeoffs including frequency response, beam patterns, size, and weight are discussed.

Contributed Paper

8:40

4aUW6. Physics-based source separation using acoustic vector sensors in three dimensions, with comparisons to adaptive beamforming and subspace methods. Alison B. Laferriere (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92109, alaferriere@ucsd.edu) and Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Previously, we introduced a method leveraging vector sensors—compact devices that measure both acoustic pressure and particle velocity—for extracting time signatures and bearings from two angularly separated sources overlapping in time and frequency. This physics-based approach demonstrated that a two-dimensional vector sensor could resolve azimuths, amplitudes, and relative phases of two horizontal plane waves using closed-form inversion formulas, even for the case of a single FFT snapshot. In this work, we extend the algorithm to three dimensions and evaluate its performance against conventional and adaptive beamforming approaches, including Minimum Variance Distortionless Response (MVDR) beamforming and the MUltiple SIgnal Classification (MUSIC) algorithm. Through quantitative comparisons, we highlight the strengths and limitations of the physics-based approach in resolving spatially separated sources, particularly in scenarios that are challenging for standard approaches, such as rapidly moving sources. Simulations and experimental data illustrate the refined algorithm's performance relative to conventional beamforming, adaptive beamforming, and subspace methods, as well as diagnostics for identifying calibration issues. [Work sponsored by ONR TFO.]

9:00–9:20 Break

Invited Paper

9:20

4aUW7. Wind-driven movement ecology of blue whales detected by acoustic vector sensing. John P. Ryan (Res., MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039-9644, ryjo@mbari.org), Paul R. Leary (Phys., Naval Postgrad. School, Monterey, CA), William Oestreich, Kelly Benoit-Bird (Res., MBARI, Moss Landing, CA), John Calambokidis, James Fahlbusch (Cascadia Res. Collective, Olympia, WA), David Cade (Stanford Univ., Pacific Grove, CA), Vanessa M. ZoBell (Machine Listening Lab, Scripps Inst. of Oceanogr., La Jolla, CA), Tetyana Margolina, John Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA), Jarrod Santora (Univ. of California, Santa Cruz, CA), Andrew DeVogelaere (Monterey Bay National Marine Sanctuary, Monterey, CA), Chad Waluk, Christopher Wahl, Francisco Chavez (Res., MBARI, Moss Landing, CA), Brandon Southall (Univ. of California, Santa Cruz, CA), Kevin B. Smith (Phys., Naval Postgrad. School, Monterey, CA), and Jeremy Goldbogen (Stanford Univ., Pacific Grove, CA)

Endangered blue whales in the eastern North Pacific seasonally occupy essential foraging habitat in Monterey Bay National Marine Sanctuary (MBNMS) off central California. Acoustic vector sensing within MBNMS, together with biologging and ecosystem observations, is revealing blue whale movement ecology across a range of temporal scales. Over days to months, blue whales were observed to repeatedly track wind-driven upwelling plumes within which their essential prey—krill—form dense aggregations. Over the years, blue

whales predominantly occupied shelf-slope habitats, however, exceptionally strong upwelling was associated with a shift to greater occupancy of offshore deep-water habitats. The strongest upwelling year exhibited higher abundances of krill and greater offshore reach of coastal upwelling plume habitat. At both time scales, these acoustically detected behavioral patterns within the regional blue whale population are consistent with behaviors of individual whales revealed by biologging. Supporting the recovery of endangered whale populations requires understanding where, when and how they live, and how risks from human activities intersect. Biologging and periodic visual monitoring are essential to understanding the spatiotemporal patterns of whale habitat occupancy and movement ecology, and these patterns can be resolved in greater detail and with greater persistence through continuous acoustic monitoring of highly soniferous species.

Contributed Paper

9:40

4aUW8. Using complex intensity to detect signals masked by shipping noise. Kevin Souhrada (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093, kcsouhrada@gmail.com) and Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Detecting weak signals amidst strong interfering sources remains a relevant ocean acoustics research problem, especially in regions with significant shipping activity and/or fast-moving sources. Acoustic vector sensors, which measure particle velocity along with pressure, provide an opportunity to detect weak signals under circumstances where hydrophone arrays are impractical. These sensors can instantaneously estimate a field's predominant directionality by constructing a complex intensity vector. Plotting this directionality as an "azigram," analogous to a spectrogram, provides new avenues for signal detection in an acoustic field dominated by a spatially compact source. Here, we use theory and data to examine how a vector sensor can enhance the detection of whale calls in the presence of shipping noise strong enough to mask the calls on a conventional hydrophone without employing computationally intensive conventional beamforming. We investigate how the presence of a secondary source influences the direction and magnitude of the active/reactive intensity and the polarization of the velocity field, as well as analyze the combinations of signal-to-interferer ratio (SIR) and angular separation over which weak source detection may be feasible. These approaches have applications for passive acoustic monitoring of biological activity in the presence of anthropogenic noise. [Work sponsored by ONR TFO.]

Invited Papers

10:00

4aUW9. Angle of arrival estimation of low-frequency sources using a compact hydrophone array towed by an autonomous surface vehicle. Davis Rider (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, drider3@gatech.edu), Matthew McKinley (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., Atlanta, GA), James S. Martin (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Laurent Grare, Luc Lenain (Scripps Inst. of Oceanogr., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Long-range detection and localization in deep water underwater sources with moving platforms typically rely on estimating the angle of arrivals at low frequency (i.e., associated with large wavelengths). Traditional methods typically use a long towed array of hydrophones to provide the required angular directivity at low frequency. However, such arrays are typically costly and complex to operate and require a powerful vehicle capable of towing such arrays. Here, an alternate method is demonstrated using a small autonomous surface vehicle platform capable of low-frequency source localization. The vehicles—Liquid Robotics Wave Gliders—are equipped with a compact four-element tetrahedron-shaped hydrophone array and a CTD suite capable of profiling to 150 m depth. These vehicles were deployed above the New England Seamounts in 2023 and 2024 and recorded low-frequency (200–300 Hz) transmission at ranges upwards of 200 km from bottom-moored sources deployed in the SOFAR channel. Pressure gradient-based signal processing methodology is applied to the compact tetrahedron hydrophone array recordings to emulate the low-frequency angular directionality performance of a vector sensor. The influence of environmental variability and seamount bathymetry on the estimated elevation and azimuthal angles with this compact array is investigated experimentally and numerically using ray-tracing simulation. [Work sponsored by ONR.]

10:20

4aUW10. Atmospheric river noise source discrimination using a cabled dual acoustic vector sensor array. Paul R. Leary (Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, pleary@nps.edu), Derek Olson (Oceanogr., Naval Postgrad. School, Monterey, CA), John P. Ryan (Res., MBARI, Moss Landing, CA), and Kevin B. Smith (Phys., Naval Postgrad. School, Monterey, CA)

Wind and rain at the ocean surface, along with distant shipping, represent the background baseline of the underwater ambient noise environment. While many studies have focused on the levels and frequencies associated with surface weather observations, the directionality and spatial distribution of surface-generated noise is poorly understood, in part due to the difficulties involved in taking directional underwater acoustic measurements. Here, we present results from a cabled, dual acoustic vector sensor system deployed on the continental slope outside Monterey Bay, CA at 900 m depth, during a series of atmospheric river events in late 2022. We present an analysis of the directionality of various surface noise sources and correlations in the direction of these signals with available meteorological measurements. Leveraging adaptive beamforming techniques, optimized for discriminating multiple sources, we examine the directional distribution of ambient noise during intense wind and rain and show a complex soundscape comprised of localized and disbursed weather inputs. We suggest that the commonly observed omnidirectional ambient noise level can be a composite of spatially discrete noise sources that may be discriminated directionally. Improving our understanding of noise direction has a variety of applications including improving sonar performance, optimizing threat surveillance, oceanographic inference, bioacoustics, and marine mammal monitoring.

10:40

4aUW11. Self-localization of a distributed array of small autonomous underwater platforms using a shipping source of opportunity. Eddy Boufarah, Davis Rider (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., Atlanta, GA), Matthew McKinley (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Laurent Grare, Luc Lenain (Scripps Inst. of Ocean., La Jolla, CA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Using a network of autonomous underwater platforms as a distributed coherent sensing array requires precise positioning of each sensor node. Here, a self-localization (i.e., totally passive) method using only acoustic sources of opportunity (such as surface vessels) to locate the mobile sensor nodes is presented as an alternative to conventional active short or long baseline systems. Existing underwater self-localization methods have mainly been developed for mobile platforms equipped with time-synchronized hydrophones and rely only on the time difference of arrivals between multiple pairwise combinations of the mobile hydrophones as inputs for a complex non-linear inversion procedure. Instead, we present a three-dimensional self-localization method using a linear least square formulation between mobile time-synchronized underwater platforms equipped with a compact directional hydrophone array based on their acoustic recordings of a distant surface vessel and their inertial navigation systems measurements. The influence of acoustic refraction and environmental variability on the performance of this self-localization method is investigated using deep water acoustic data collected near the Atlantis II seamounts in the Northwest Atlantic by two Wave Gliders instrumented with underwater towed acoustic modules located above and below the sonic layer depth for separation distance up to a few kilometers. [Work sponsored by ONR.]

THURSDAY AFTERNOON, 22 MAY 2025 GALERIE 3, 1:00 P.M. TO 4:40 P.M.

Session 4pAAa

Architectural Acoustics, Signal Processing in Acoustics and Computational Acoustics: Data-Driven Room Acoustics II

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180

Xenofon Karakonstantis, Cochair
Electrical Engineering, Technical University of Denmark, Ørsteds Plads, Building 352, Kgs. Lyngby 2800, Denmark

Contributed Papers

1:00

4pAAa1. Evaluating the influence of building information modeling's level of development on the accuracy of acoustic simulations of residential, office, and educational spaces. Harrison W. Freedman (Architecture, Univ. of Florida, 1480 Inner Rd., Gainesville, FL 32601, harrison.freedman@icloud.com), Hassan Azad, and Nawari Nawari (Architecture, Univ. of Florida, Gainesville, FL)

This study investigates the impact of Building Information Modeling (BIM) Level of Development (LOD) on the accuracy of acoustic simulations in office, residential, and classroom environments. Revit models are created at three LODs of 100, 350, and 500, and acoustic simulations are performed using the EASE and ODEON acoustic computer programs. The research evaluates how increasing model detail, such as refined geometric representations and material properties, influences the prediction of key acoustic metrics and auralization results. Standardized simulation parameters are utilized to ensure consistency across all LODs and environments. Preliminary results indicate that higher LODs enhance the accuracy of acoustic simulations, providing more reliable insights into sound behavior within complex spaces. Additionally, the findings address challenges related to data exchange between BIM platforms and acoustic tools, proposing strategies to improve workflow efficiency and reduce errors. By focusing on non-concert hall environments, this research expands the scope of acoustic design, emphasizing its critical role in optimizing everyday spaces for functionality and comfort. The study offers valuable contributions to integrating BIM and acoustic simulation tools, demonstrating their potential to streamline design processes and improve architectural outcomes.

1:20

4pAAa2. Exploring the intersection of sound reverberation and quantum interference behavior. Juliette Tudoce (Quantum Optics Theory, ICFO, 150 Carrer de la Industría, Barcelona 08025, Spain, juliettetudoce@me.com), Reiko Yamada, and Maciej Lewenstein (Quantum Optics Theory, ICFO, Barcelona, Spain)

While the relationship between quantum physics and sound may not be immediately apparent, it offers profound insights into wave behavior and interference patterns. This work investigates how acoustic reverberation—a classical sound phenomenon—compares to quantum interference because they share principles of wave interaction within bounded systems. Reverberation occurs when a sound source interacts with an acoustic cavity and this interaction creates complicated interference patterns that resemble the

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quantum behaviors observed in cavity quantum electrodynamics. Investigating these parallels shows connections between acoustic and quantum systems and links spectral enhancements in sound to phenomena like the Casimir effect in near-field physics. The main objectives of this research are to use simulated reverberation tools in artistic and electro-acoustic music creation, designed by harnessing quantum interference principles; explain real-world reverberative sound behavior in controlled environments by providing a quantum framework; and theoretically create the equivalence of acoustic reverberation in quantum interference. This project results in a cross-disciplinary approach that bridges physics and acoustics, allowing applications in sound design and quantum-inspired audio synthesis.

Invited Papers

1:40

4pAAa3. A novel algorithm for model evaluation in acoustic studies with small datasets: A comparison of linear and neural network models. Semiha Yilmazer (Dept. of Interior Architecture and Environ. Design, Bilkent Univ., Faculty of Art, Design and Architecture, Ankara 06800, Turkey, semiha@bilkent.edu.tr) and Zekiye Şahin (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

Traditional neural network models rely on dividing data into training and testing sets to evaluate performance. However, when working with small datasets—a common challenge in acoustic studies—the test model's sensitivity to the selection of training and testing data poses a significant limitation, often leading to inconsistent results. To address this issue, we propose a novel algorithm that provides an alternative structure for model evaluation. This study introduces the proposed algorithm and demonstrates its effectiveness by applying it to a small acoustic dataset. As a case study, we use the algorithm to compare the efficacy of a linear model with an artificial neural network model. Our results highlight the potential of this approach to improve reliability in neural network applications within the field of acoustics, especially when data availability is limited.

2:00

4pAAa4. Efficiency and uncertainties of room-acoustic decay parameter estimation. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu) and Zühre Sü Gül (Dept. of Architecture, TED Univ., Ankara, Turkey)

Sound energy decay analysis plays a fundamental role in a broad range of room-acoustic applications. This paper addresses the challenges of analyzing multiple-slope energy decays often encountered in experimentally measured data. Previous efforts by Xiang *et al.* [J. Acoust. Soc. Am., 129, 741–752 (2011)] have established a parametric model derived from Schroeder integration, that breaks the Schroeder decay functions down to single or multiple exponential decays. Several advanced methods based on this parametric model, such as nonlinear regressions, Bayesian methods, and artificial neural networks have been developed to cope with the challenges in decay parameter estimations. Using these methods, a wide range of data resolutions can meet the need for room-acoustic decay analysis. Yet for high efficiency, acousticians can use lower resolutions, still adequately representing energy decay processes. This paper discusses conditions of representing Schroeder integration by desirable, sufficiently less data points for higher efficiency of the decay parameter estimation. At the same time, increased efficiency brings uncertainties. Within the Bayesian framework, the numerical uncertainties are investigated against those of experimental measurements. Driven by experimental data in performing arts venues, this paper quantifies uncertainties for leveraging between adequate accuracies and the analysis efficiency and discusses the probabilistic versus deterministic estimations.

2:20

4pAAa5. A finite element approach to predicting free-field diffusion coefficients according to ISO 17497-2. Luiz A. Alvim (RPG Acoust. Systems LLC, Passaic, NJ), Rinaldi P. Petrolli (RPG Acoust. Systems LLC, Florianópolis, Santa Catarina, Brazil), and Peter D'Antonio (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com)

The experimental measurement of the diffusion coefficient according to ISO 17497-2 is time-consuming. It requires several sample periods of a diffusor, an anechoic or large reflection-free volume, and far-field conditions. Wave-based methods have accurately predicted the diffusion coefficient without the need to fabricate 3-D print scale models or full-scale samples for testing. This work describes a Finite Element approach with a Perfectly Matched Layer that predicts the free-field diffusion coefficient directly from a 3-D CAD file, addressing limitations found with the Boundary Element Method, such as the thin-shape breakdown, while also allowing frequency and region-dependent complex fluid properties to predict scattering from equivalent-fluid models. The results are validated with scale model boundary-plane goniometer measurements and numerical references, verifying that it is possible to predict diffusion coefficients following ISO 17497-2 and overcome the limitations of the BEM method.

2:40

4pAAa6. Advancements in iterative optimization for room acoustics. Rinaldi P. Petrolli (Res., REDI Acoust., Florianópolis, Santa Catarina, Brazil) and Peter D'Antonio (Res., REDI Acoust., 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com)

This presentation introduces a palette of innovative, iterative optimization software for critical listening rooms of any size and shape. The Non-cuboid Iterative Room Optimizer (NIRO) is a pioneering tool that optimizes room geometry, listener placement, and both free-standing and soffit-mounted loudspeaker positioning for non-rectangular spaces. The Room Optimization for Cuboid Spaces (ROCS) algorithm focuses on rectangular rooms, simultaneously optimizing room dimensions and the placement of listeners and speakers. The Treatment Optimization for Room Acoustics (TORA) software is the first to determine the optimal acoustical treatment for all of the allowable architectural locations from a library of tested treatment options. Additionally, the Acoustical Treatment Calculator (ATC) is available online as a free resource to design absorption treatments. This suite of tools uses a combination of analytical calculations,

wave-based and geometrical acoustics numerical methods, and optimization algorithms to improve different room acoustic metrics within the allowable architectural constraints. Examples will be presented.

3:00–3:20 Break

Contributed Paper

3:20

4pAAa7. Objective assessment of coloration in active acoustic systems: Real-room measurements with time-invariant filter equalization. Simon Neeten (Amadeus Acoust., Hamburgerstraße 10/10, Vienna 1050, Austria, simon@amadeus-acoustics.com), Clemens Frischmann (Rohde Acoust., Salzburg, Austria), Fabio Kaiser, and Volker Werner (Amadeus Acoust., Vienna, Austria)

Active acoustic systems are designed to enhance or modify a room's natural acoustics, using a matrix-based processing of loudspeakers and microphones. Feedback loops in these systems cause a maximum amplification level, known as gain before instability (GBI). A well-adjusted gain limit is essential to avoid unwanted acoustic feedback. A safety margin of 2 dB or more is recommended, with some sources advising a margin of 5–7 dB for time-invariant systems. Even below the GBI, uneven energy distribution may cause unwanted coloration. State-of-the-art methods improve the GBI in active acoustic systems; however, time-variant filters, often used in these approaches, can introduce additional artifacts. Therefore, time-variant filters are excluded from consideration in this paper. In order to control room parameters like clarity, proximity, or lateral fraction, it might become necessary to deviate from an even energy distribution to achieve the desired results. This stability aspect has not been previously addressed and requires a separate examination. In this paper, a spectral-flatness-based method to measure the coloration of an active acoustic system in an objective way is introduced. By applying this method across different system equalization stages, we provide a comparison of coloration effects, using measurements taken in real rooms with a functional active acoustic system.

Invited Papers

3:40

4pAAa8. Spatial extrapolation of room impulse responses using point-neuron learning. Amy Bastine (School of Eng., Australian National Univ., 34 Marcus Clarke St., Academie House Unit 110, Canberra, Australian Capital Territory 2601, Australia, amy.bastine@anu.edu.au), Thushara D. Abhayapala, and Prasanga Samarasinghe (School of Eng., Australian National Univ., Canberra, Australian Capital Territory, Australia)

Accurate rendering of room acoustics requires dense Room Impulse Response (RIR) data across large volumes, a process often constrained by resource-intensive data acquisition. To address this, various soundfield reconstruction techniques have been developed, with physics-guided machine-learning models emerging as the most efficient. This paper presents a method to spatially extrapolate RIRs by applying a recently developed point-neuron learning framework that embeds the fundamental solution of the wave equation directly into its architecture ensuring strict adherence to physical laws. By leveraging prior knowledge of the acoustic environment and generalizable features such as dominant reflection locations and relative frequency response, our approach effectively integrates early reflections and late reverberation into a cohesive model. Comparative evaluations against competing methods demonstrate that the proposed method consistently achieves superior reconstruction performance across diverse acoustic scenarios.

4:00

4pAAa9. A comparison framework for sound field reconstruction methods using Bayesian inference. Antonio Figueroa-Duran (Dept. of Elec. and Photonics Eng., Tech. Univ. of Denmark, Ørsteds Plads Byg 352, Kgn. Lyngby 2800, Denmark, anfig@dtu.dk), Xenofon Karakonstantis (TrackMan, Kgs. Lyngby, Denmark), and Efren Fernandez-Grande (Polytechnic Univ. of Madrid (UPM), Madrid, Spain)

In recent years, the use of data-driven methods in sound field analysis has advanced rapidly. These approaches often require a large number of observations, which are scarce in acoustics. Hybrid approaches offer a promising alternative by integrating computational techniques with sound field models and theoretical considerations, thereby restricting the solution space. This work introduces a Bayesian framework for sound field reconstruction that incorporates general acoustic constraints through prior distributions. The framework presents a modular design, whereby multiple sound field models (plane waves, spherical waves and diffuse kernel) can be used, as well as regularization in both time and frequency domains. Additionally, the Bayesian formulation enables cross-frequency information sharing by selecting the appropriate frequency covariance matrix. The framework's performance is evaluated in two different real-life scenarios, including interpolation and extrapolation from the measurement array. This comparison highlights that the optimal sound field model is contingent on the characteristics of the observed sound field. The outcome of this study is twofold: first, it presents a modular framework that merges classical sound field reconstruction methods with a Bayesian inference formulation; second, it evaluates the performance of different reconstruction methods in different experimental scenarios.

Contributed Paper

4:20

4pAA10. A method for visualization of volumetric acoustic wave propagation in Rhinoceros 3D. Matt Skarha (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, mskarha@thresholdacoustics.com), Laura C. Brill, Chris Springthorpe, and Dawn Schuette (Threshold Acoust., Chicago, IL)

Wave-based simulation methods are valuable tools in room acoustic modeling due to their natural ability to incorporate both geometrical behavior and diffraction by directly solving the wave equation in three spatial dimensions. These simulations can enhance both education and architectural acoustic design by enabling detailed visualization of acoustic wave propagation, particularly early reflection patterns in performance spaces. PFFDTD is an open-source implementation of the finite-difference time-domain (FDTD) method for 3-D room acoustics by Brian Hamilton incorporating frequency-dependent impedance boundaries (Hamilton, 2021). This paper gives a method for visualizing volumetric room impulse responses taken as an output from PFFDTD in the NURBS-based modeling software Rhinoceros 3D, using the Grasshopper plug-in as a control interface. We present case studies of early reflections in models of music performance spaces, showcasing the potential for Rhino-based wave visualization in architectural acoustic design.

THURSDAY AFTERNOON, 22 MAY 2025 GALERIE 2, 12:55 P.M. TO 5:20 P.M.

Session 4pAAb

Architectural Acoustics, Noise and Underwater Acoustics: Memorial Session Honoring David Lubman

Braxton Boren, Cochair
Performing Arts, American University, 4400 Massachusetts Avenue NW, Washington, DC 20016

Francesco Martellotta, Cochair
Department of Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy

Jason E. Summers, Cochair
Applied Research in Acoustics LLC, 1222 4th Street SW, Washington, DC 20024-2302

Chair's Introduction—12:55

Invited Papers

1:00

4pAAb1. David Lubman's service with the Paul S. Veneklasen Research Foundation. John LoVerde (Paul S. Veneklasen Res. Foundation, Cypress, CA, johnloverde@gmail.com) and David W. Dong (Paul S. Veneklasen Res. Foundation, Cypress, CA)

David served for many years on the Board of Directors of the Paul S. Veneklasen Research Foundation, a non-profit dedicated to advancing acoustical research. Evaluating grant proposals from all subfields of acoustics, David's breadth of knowledge was invaluable to the work of the Foundation. David was particularly passionate about projects that had the potential to directly help students and underserved communities.

1:20

4pAAb2. David Luman: A career of advocacy and mentorship. Robin Glosemeyer Petrone (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, robin@thresholdacoustics.com) and Laura C. Brill (Threshold Acoust., Chicago, IL)

David Lubman's advocacy work in creating a standard for classroom acoustics has impacted the learning environments for students and teachers for decades—an achievement to be celebrated on its own. But within the Acoustical Society, his enthusiasm for acoustics was contagious, creating profound impacts on young professionals. We will share the stories of ourselves, two individuals entering the field of acoustics nearly two decades apart, whose careers were shaped by our involvement in classroom acoustics advocacy and through David's encouragement.

1:40

4pAAb3. In-field measurements of sound-energy distribution in classrooms. Dario D'Orazio (Dept. of Industrial Eng., Univ. of Bologna, Viale Risorgimento, 2, Bologna 40126, Italy, dario.dorazio@unibo.it), Jack Harvie-Clark (Apex Acoust., Newcastle Upon Tyne, United Kingdom), Giulia Fratoni, and Virginia Tardini (Dept. of Industrial Eng., Univ. of Bologna, Bologna, Italy)

David Lubman's interest in classroom acoustics contributed to the development of the ANSI S12.60 standard. Classroom standards, either explicitly or implicitly, aim to ensure an adequate signal-to-noise ratio (SNR) for all students in normally occupied positions. This objective is achieved through strategies adhering to well-known room criteria, such as reverberation time (T). Assuming a uniform distribution of background noise, SNR then depends on the spatial distribution of sound energy. This latter aspect depends on T and the arrangement of absorbing materials. This is why standards, in some cases, provide informative recommendations on the placement of absorbing materials, as exemplified in Annex C of ANSI S12.60-1. The study experimentally verifies the slope of the spatial distribution of sound energy within classrooms of different types, comparing ANSI S12.60 with two European standards: the Italian UNI 11532-2 and the British BB 93. The distributions of G and G50 from in-field measurements were analyzed at mid-range frequencies (500–1000 Hz) and in the 250 Hz octave band. Measurements show greater spatial decays than those generally reported in the literature. The recommendations regarding the distribution of absorbent material suggested by the three standards are discussed.

2:00

4pAAb4. Acoustics of Worship Spaces: Forty years after Lubman and Wetherill volume. Francesco Martellotta (Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy, francesco.martellotta@poliba.it)

Acoustics of Worship Spaces was published in 1985 having David Lubman as senior editor. At that time, published studies on worship acoustics could be counted on hands and often were focussed on individual case studies. Conversely, the 1985 book proposed for the first time a systematic approach, where architectural and acoustical data were given for a number of spaces, allowing professionals and researchers to compare and evaluate performances, understanding the role of shape and materials, and always keeping an eye on practical issues. Forty years later, pondering the amount of research, publications and books that arose on the topic, we can clearly appreciate how inspiring that work was. Similarly, "The History of Western Civilization Told Through the Acoustics of its Worship Spaces" presented at ICA 2001, shedding a new light on the connection between architecture and sound, sparked more studies that extended to broader fields. A review of some of such studies is presented, selecting them according to their relevance, geographic span, targeted audience (professionals/researchers), and, most importantly, outlining their debit toward David Lubman's work.

2:20

4pAAb5. Room as communication channel: David Lubman's role in casting the relationship between program and room acoustics as an information-theoretical problem. Jason E. Summers (ARiA, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Speakers, liturgists, performers, and acousticians have long recognized the interplay between room acoustics, program, and performance. Specific programs and styles of performance are better suited to specific spaces, suggesting that liturgical and musical forms developed to suit performance and worship spaces. In their 2001 paper "The History of Western Civilization Told Through the Acoustics of its Worship Spaces," Lubman and Kiser went a step further to propose a causal mechanism; specifically, that changes in liturgical styles that evolved with changes in church architecture reflected the need to provide optimum encoding in an information-theoretic sense. While information theory had been used in prior work as a tool to predict channel capacity in rooms (e.g., methods for speech intelligibility developed by Peutz) and tied to elements of delivery (e.g., speech rate), the development of liturgical and musical forms had not been cast as a communication problem of maximizing information transfer. In demonstrating the empirical plausibility of this conjecture, Lubman laid the groundwork for future theoretical and empirical development in room acoustics. Here, prior development inspired by Lubman's work is chronicled and opportunities for future work that can leverage recent developments in information theory and audition to explain subjective findings are highlighted.

2:40

4pAAb6. Mexico City Metropolitan Cathedral acoustical computational model. Ricardo Teo Vazquez Turner (Facultad de Musica, Universidad Nacional Autonoma de Mexico, Xicotencatl 126, Mexico, CDMX 04100, Mexico, vaztur12@gmail.com), Braxton Boren (Performing Arts, American Univ., Washington, DC), Alejandro Ramos (Laboratorio de Oceanografía Física, Instituto de Ciencias del Mar y Limnologia, Mexico, CDMX, Mexico), Pablo Padilla (Departamento de Matemáticas y Mecánica, Instituto de Investigaciones en Matemáticas Aplicadas y en Sistemas, Mexico, CDMX, Mexico), Guadalupe Caro (EHE, Instituto Tecnologico y de Estudios Superiores de Monterrey, Mexico, Estado de Mexico, Mexico), and Jezzica Zamudio (Instituto Politecnico Nacional, Mexico, CDMX, Mexico)

In the present work, the 3-D model and the impulse-response data of the Mexico City Metropolitan Cathedral (Boren *et al.*, 2016) help us develop and calibrate an acoustical computational model of the worship space. The aim of the work is to model the acoustics of the Cathedral, as it would have sounded in the early days of its finished construction in the 19th century and in the present in order to investigate the repertoire of the Christmas ceremony and mass and the musical organs performance to analyze and auralizate their musical soundscape.

3:00–3:20 Break

3:20

4pAAb7. Sound and history: David Lubman's contributions to a systematic understanding of the evolution of acoustic spaces. Braxton Boren (Performing Arts, American Univ., 4400 Massachusetts Ave. NW, Washington, DC 20016, boren@american.edu)

David Lubman's decades of work within the ASA included contributions to the study of noise, diffusion, classroom and worship space acoustics. One of his most significant achievements was his work examining the role of sound in history. While previous work in "archaeoacoustics" was mainly limited to archaeology departments, David's work combined technical expertise with an insatiable curiosity and an unwillingness to be hemmed in by disciplinary boundaries. This talk will examine the echoes of David's work in many modern applications of archaeoacoustics research today. It will also consider ways in which David's example can help acoustics researchers think about history more comprehensively, as well as making the case for the importance of sound to traditional historians and humanists more generally.

3:40

4pAAb8. Archaeoacoustic research and public outreach: Experience of the Past Has Ears at Notre-Dame. Brian F. Katz (Sorbonne Université, CNRS, d'Alembert, CNRS, Paris 75012, France, brian.katz@sorbonne-universite.fr), David Poirier-Quinot (Sorbonne Université, CNRS, Paris, France), Jean-Marc Lyzwa (Conservatoire National Supérieur de Musique et de Danse de Paris, Paris, France), Julien De Muynke (Sorbonne Université, CNRS, Paris, France), Mylène Pardoen (Maison des Sci. de l'Homme 5.02 reviews Res. Inst. in Lyon, Lyon, France), Elliot K. Canfield-Dafilou, and Sarabeth Mullins (Sorbonne Université, CNRS, Paris, France)

The recent fire at Notre-Dame Cathedral in Paris spurred research aimed at understanding its evolving acoustics across centuries and examining its influence on music. These efforts form part of two related initiatives launched in response to the fire: the French interdisciplinary research project *The Past Has Ears at Notre-Dame* (PHEND) and the European cultural heritage project *The Past Has Ears* (PHE). Recognizing the cultural significance of this work, a substantial effort and budget were devoted to public outreach—an uncommon step for such projects. As a result, the research findings were transformed into immersive public experiences through several audio productions that utilized the project's methodologies and insights to engage audiences via storytelling: a radio-fiction, an audio-guide application, and a virtual concert. Each of these productions benefited from immersive spatial audio and historically informed geometric acoustic simulations and soundscape reconstructions. This paper provides an overview of the rationale behind these productions, their development process, and their public reception. We hope these experiences will guide and inspire other researchers in the emerging field of archaeoacoustics to broaden their reach beyond academic circles while upholding scientific rigor.

4:00

4pAAb9. Sounds like… culturally significant auditory illusions at archaeological sites. Steven J. Waller (Rock Art Acoust., 1952 Sonoma Ln., Lemon Grove, CA 91945, wallersj@yahoo.com)

Unusual sound effects have been noted at many archaeological sites. Tangible cultural evidence supports hypotheses that certain sound phenomena were misinterpreted. Cases of auditory illusions include: (1)Chirped echoes from the staircases of Mesoamerican pyramids, resembling calls of the quetzal bird, as proposed by David Lubman; (2)Echoes in canyons perceived as spirits calling out from the rocks, matching ancient petroglyph designs that coincide with mythical descriptions of echo spirits; (3)Thunderous reverberation in caves with prehistoric paintings of stampedes of hooved animals corresponding to mythical thunder gods; (4)Interference patterns from two droning bagpipes radiate nodes of silence, giving the impression of acoustic shadows from a ring of massive objects blocking the sound. This pattern matches the arrangement of Stonehenge, and is consistent with a legend that two pipers led maidens to dance in a circle and they all turned to stone; (5)Resonance in gothic cathedrals, leading to the belief that angels were singing along in the ceiling. These examples demonstrate the significance of sound to ancient cultures and underscore the need for further acoustical studies at archaeological sites. Implied from these results is the need for conservation efforts to preserve the acoustics at archaeological sites.

4:20

4pAAb10. Resonating rocks: The acoustics of Hindu cave temples. Shashank Aswathanarayana (Performing Arts, American Univ., 4400 Massachusetts Ave. NW, Washington, DC 20016, shashank@american.edu) and Braxton Boren (Performing Arts, American Univ., Washington, DC)

Sound has played a fundamental role in Hindu worship since the early Vedic period (ca. 2000 BCE). History indicates that Indian civilizations had a strong oral tradition and relied on knowledge being transmitted through oral means rather than written text. The start of the post-Vedic period (ca. 1st century BCE) was marked by the emergence of temple worship, with rock-cut cave temples among the first structures. While cave temples are critical to the evolution of Hindu religious practices, their acoustics remain an unexplored area. This work addresses the gap by analyzing the acoustics of four Hindu cave temples in the Badami region of present-day Karnataka, Southern India. These caves carved into red sandstone date back to the 6th and 7th centuries CE, with two dedicated to Lord Vishnu and two to Lord Shiva. We analyzed the impulse responses and decay curves measured at Cave temples 1, 2, and 3 in Badami as well as the Ravana Phadi cave temple in Aihole. Standard acoustic parameters like T30, C80, D50 as well as non-standard parameters like resonance quality and resonance width are computed to provide insight into their acoustic properties.

Contributed Papers

4:40

4pAAb11. Proxy churches as tools for historical virtual acoustic reconstruction: The 12th-century acoustics of Notre-Dame de Paris. Sarabeth Mullins (Sorbonne Université, CNRS, Couloir 55-65, Bureau 313, 4 Pl. Jussieu, Paris 75005, France, Sarabeth.Mullins@sorbonne-universite.fr) and Brian F. Katz (Sorbonne Université, CNRS, Paris, France)

Accurately reconstructing the acoustics of lost historical spaces requires interdisciplinary approaches and careful consideration of uncertainty. This study presents the development of an acoustical model of a Romanesque cathedral, termed Notre-Dame II, that was replaced in the 12th century by the current cathedral in Paris. This was done to examine the acoustic conditions under which early Parisian polyphonic music may have been performed. Notre-Dame II, described in surviving documentation and found in ruins underneath the modern cathedral, required reconstruction through indirect evidence due to the absence of detailed architectural plans or descriptions. A proxy church of similar era and construction was selected to approximate key geometrical and material parameters. Material properties were estimated based on comparable historical structures, and the calibration relied on known acoustic metrics in the proxy, including reverberation time and clarity. Uncertainty arose at multiple stages, including limited documentation of Notre-Dame II, assumptions about proxy church materials, and the inherent limitations of computational acoustics methods. This study highlights how iterative calibration and sensitivity analyses mitigate these challenges, providing insights into the use of extant structures to study the acoustic experience of lost buildings.

5:00

4pAAb12. Reconstructing ancient soundscapes: An acoustic analysis of three Early Dynastic tombs at North Saqqara. Nima Farzaneh (Music, Stanford Univ., 47 Olmsted Rd., Apt. 423, Stanford, CA 94305, nfarzan@stanford.edu)

Inspired by David Lubman's pioneering work in archaeoacoustics and his investigations into the interplay of architecture and sound in historical contexts—most notably at Mesoamerican sites like Chichén Itzá—this study explores the sonic properties of three Early Dynastic tombs constructed during the reign of Den at North Saqqara. Although Egyptian funerary architecture is often studied for its visual and material characteristics, examining its acoustic dimensions provides insight into how these spaces were experienced. Drawing on geometry, materials, and architectural details from archaeological findings, we developed three-dimensional models and used digital acoustic simulations to measure impulse responses at multiple source and listening positions. The results revealed significant sound attenuation (25–35 dB) between interior and exterior areas, suggesting that ritual activities within the burial chambers were likely inaudible outside. Extended reverberation in coupled spaces may have enhanced ritual or musical resonance while reducing speech intelligibility. Additionally, niche-style "palace façades" scattered sound, lowering overall loudness and mitigating flutter echo to improve clarity. By foregrounding these acoustic properties, we show how funerary architecture regulated privacy and the auditory atmosphere, influencing ceremonial practices and shaping mourners' experiences, thereby deepening our understanding of ancient Egyptian funerary rituals.

THURSDAY AFTERNOON, 22 MAY 2025 GALERIE 4, 1:20 P.M. TO 4:20 P.M.

Session 4pAB

Animal Bioacoustics: General Topics in Animal Bioacoustics

Brittany L. Jones, Chair
National Marine Mammal Foundation, 4982 Santa Cruz, San Diego, CA 29107

Contributed Papers

1:20

4pAB1. Rotation-independent discrimination of object shape by echolocating bottlenose dolphins (*Tursiops truncatus*). Siena Merk (Biologic and Bioacoustic Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, siena.merk@nmmpfoundation.org), Katie A. Christman (Biologic and Bioacoustic Res., National Marine Mammal Foundation, San Diego, CA), Jason Mulsow, and James Finneran (US Navy Marine Mammal Program, San Diego, CA)

Previous research has examined the perception of aspect-dependent objects by echolocating bats and dolphins; however, the importance of perceived spatial features as opposed to acoustic cues in discriminating and recognizing targets is not fully understood. In this study, two bottlenose dolphins (*Tursiops truncatus*) learned a two-alternative forced choice task requiring discrimination of a two-cylinder "dipole" target with 12-cm center-to-center spacing (S+) from one of three S− targets: a "monopole" target, or one of two dipoles with 6- or 18-cm spacing. Targets rotated in the horizontal plane on a trial-to-trial basis so that echo timing and azimuth changed depending on the dipoles' orientations relative to the dolphin. Target cylinders were identical in composition and size so that acoustic cues other than the echo timing and azimuthal separation were not available, and random positional roving reduced the possibility of any single acoustic feature indicating the S+. After achieving consistently high performance with a limited set of target rotations, testing was repeated using novel rotation angles. Results—including first trial data with novel rotations—support dolphins forming a distal, spatial representation of target shape instead of relying solely on proximal, auditory cues, a capability likely necessary in cluttered aquatic environments. [Work funded by ONR.]

1:40

4pAB2. Bottlenose dolphins reliably discriminate single- versus two-highlight echoes independent of highlight timing and amplitude. Katie A. Christman (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, katie.christman@nmmf.org), Jason Mulsow (Code 56710, NIWC Pacific, San Diego, CA), Siena Merk (Conservation Biology, National Marine Mammal Foundation, San Diego, CA), Dorian Houser (National Marine Mammal Foundation, San Diego, CA), and James Finneran (Code 56710, NIWC Pacific, San Diego, CA)

Previous experiments have demonstrated that big brown bats deconstruct echo-highlight timing onto a range axis to perceive the target shape. These experiments have also suggested that manipulation of echo amplitude changes the perceived target range; increased delay of auditory neural responses with decreasing echo amplitude leads to an apparent increase in target range (i.e., an amplitude-latency trading effect). These findings motivated the current series of experiments with bottlenose dolphins. Three dolphins were trained in a two-alternative forced-choice task comparing phantom targets with mean simulated ranges of 5 m. The S+ was a two-highlight echo with fixed highlight delays, and the S− was a single-highlight echo that varied in delay and amplitude. In the previous bat experiments, more errors ("error peaks") occurred when the S− delay was equal to one of the two S+ highlight delays, and shifts in the error peaks were noted when the amplitude of the S− echo varied. Unlike the bat experiments, dolphin performance was above 90% correct independent of S− and S+ highlight timing overlap and no error peaks arose out of S− amplitude manipulations. Dolphins apparently use different processes to determine target spatial features, with perceived range unaffected by changes in echo amplitude. [Work funded by ONR.]

2:00

4pAB3. Changes in dolphin echo-delay jitter detection thresholds when the number of jittering echoes is limited. Madilyn R. Pardini (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, madilyn.pardini@nmmpfoundation.org), Katie A. Christman (Biologic and Bioacoustic Res., National Marine Mammal Foundation, San Diego, CA), Austin O'Kelley (National Marine Mammal Foundation, La Jolla, CA), Jason Mulsow (Code 56710, NIWC Pacific, San Diego, CA), Dorian Houser (National Marine Mammal Foundation, San Diego, CA), and James Finneran (Code 56710, NIWC Pacific, San Diego, CA)

Dolphins use echo-delay to determine the range and movement of a target. Echo-delay "jitter" tasks require the dolphin to discriminate between simulated echoes with fixed delay and those with delay "jitter" (delay increases and/or decreases on successive echoes). While there have been several studies looking at dolphin echo-delay jitter-detection thresholds, results have varied based on the duration of time that the jittering echo was available. Additionally, previous experiments suggest that dolphins integrate information across multiple echoes to detect objects, but little is known about how echo-delay jitter detection may be affected when the number of echoes is limited. In this experiment a bottlenose dolphin (*Tursiops truncatus*) was trained to echolocate, listen to returning electronic "phantom" echoes, and produce a conditioned acoustic response if jitter was detected. Jitter detection thresholds were measured when the dolphin had access to 1, 3, 5, 9, 17, 33, or an unlimited number of jittering echoes. Thresholds decreased with increasing number of echoes, with the largest decrease observed between five and nine echoes (thresholds of 46 versus 1.6 μs, respectively), and then plateaued as the number of jittering echoes increased above nine. [Work supported by ONR.]

2:20

4pAB4. Characterizing sei whale (*Balaenoptera borealis*) movement patterns in the Gulf of Maine. Emma VerGow (Univ. of New Hampshire, 8 College Rd., Durham, NH 03824, emma.vergow@unh.edu), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), and S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada)

Sei whales are difficult to visually monitor due to their cryptic presence and lack of breaching; therefore, passive acoustic monitoring is now being used to expand this effort. The fine-scale temporal and spatial movements of sei whales in the Northwest Atlantic are unknown. The present study characterizes sei whale movement patterns related to migration dynamics in the Gulf of Maine. Five bottom-mounted Acoustic Long-Term Observatories landers, each equipped with a tetrahedral array of omni-directional hydrophones, were deployed throughout the Gulf of Maine. A combination of manual analysis and automated detections was used to assess sei whale daily presence and seasonality. Horizontal bearing estimation of detected vocalizations was used to examine sei whale directional movement patterns among sites. The absolute minimum number of animals present was calculated by totaling the number of directions that a species' calls arrive over a short period of time (15 min). Daily abundance was defined by the highest absolute minimum number of any 15-min time period throughout the course of the day. Presence/absence data suggests that sei whales are present in spring and fall from 2021–2024 at varying sites. This effort will contribute to the improvement of regional soundscape modeling, automated detector accuracy, and density estimation methods.

2:40–3:00 Break

3:00

4pAB5. Multi-sensor tagging and acoustic analysis for assessment of impact of seismic sparker surveys on leatherback turtle behavior at a nearshore foraging ground. Erin M. Fischell (Acbotics Res., LLC, 82 Technol. Park Dr., East Falmouth, MA 02536, efischell@acbotics.com), Oscar A. Viquez, Sam Fladung (Acbotics Res., LLC, Falmouth, MA), Farrell Davis, Ryan Munnelly, Victoria Oriole (Coonamessett Farm Foundation, East Falmouth, MA), Kate Choate, Heather Haas, Joshua Hatch (Northeast Fisheries Sci. Ctr., Woods Hole, MA), Rick Rogers (Azura, Woods Hole, MA), and Samir Patel (Coonamessett Farm Foundation, East Falmouth, MA)

High-resolution geophysical surveys are used in offshore wind development to provide data on seafloor composition. Many of these surveys use seismic sparkers, impulsive acoustic sound sources that generate signals to penetrate the seafloor. As the number of offshore wind projects grows, the impact of sparkers on marine wildlife has become a concern as the broadband sound overlaps with marine wildlife hearing. One approach to assess impact is to deploy animal-borne, multi-sensor tags to track the movement and behavior of animals in response to sparkers. A novel multi-sensor tag was built that included a camera along with an AcSense-Mini, a 1-channel acoustic recorder with IMU, pressure and temperature sensors. The tag was deployed on leatherback turtles in the vicinity of a ship-towed sparker during the fall of 2023. Data from the turtle-borne tags were analyzed for behavioral responses to the sparker; a key part of the analysis was the navigational reconstruction of the turtle track using frequency-dependent sound pressure levels with the IMU data. Findings include the acoustic sound source characteristics, two case studies of turtle response to the closest point of approach to the sparker, and limitations on turtle trajectory reconstruction to inform future design [Work funded by Bureau of Ocean Energy Management.]

3:20

4pAB6. Investigating sound transmission into a large aquarium shark tank. Madilyn K. Randall (Phys. and Astronomy, Brigham Young Univ., 217 Wymount Terrace, Provo, UT 84604, mkmoore6@byu.edu), Benjamin L. White, Molly Boseman, Trigg Randall, Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Michael Ashcroft, and Ari Fustukjian (Loveland Living Planet Aquarium, Draper, UT)

Sharks primarily hear frequencies ranging from 100–400 Hz. This study aims to compare sound exposure in these frequencies experienced by sharks in the wild and those in a large aquarium tank. In the tank, sharks are exposed to noise from filtration systems and sound transmitted from public spaces. While sound levels in public spaces are easy to measure, less has been reported regarding sound levels in tanks. To understand the relationship between sound levels in public areas and the shark tank, simultaneous acoustical measurements were taken at the Loveland Living Planet Aquarium in Utah. This shark tank has public spaces along two long walls and a tunnel underneath one side of the tank. In these public spaces, chirps and crowd noise are played through loudspeakers. Sound levels outside and inside the tank are compared. The correlation between these levels is important to understand in order to make informed decisions when setting noise

level regulations in viewing areas. The spectral levels measured in the tank are also compared to natural ocean soundscapes, to explore how tank noise differs from what sharks experience in the wild. [Undergraduate research supported by the College of Computational, Mathematical, and Physical Sciences, Brigham Young University.]

3:40

4pAB7. Accuracy improvement in few-shot bird call detection by automatic identification of the frequency range. En-Kuan Zhu (Elec. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu 300044, Taiwan, zhunkuan1598@gmail.com), Sheng-Lun Kao, and Yi-Wen Liu (Elec. Eng., National Tsing Hua Univ., Hsinchu, Taiwan)

Objective: This study aims to address the limitations in few-shot bioacoustic event detection, such as the scarcity of labeled data and the selection of appropriate frequency ranges. Data collection: We collected 17 bioacoustic recordings of varying lengths from a university campus and a popular hiking trail in Hsinchu, Taiwan, with 8 bird species subjectively identified. The total duration is 1 h and 12 min, with a total of 1389 positive events. Methods: Our approach was built on the 2023 DCASE Task 5 baseline, the Prototypical Network (PN), enhanced with an Automatic Frequency Range Identification (AFRI) method which computes the low- and high-frequency bounds by comparing the Mel-spectral power density difference between positive events and negative events. Results: Using PN alone, we achieved an *F*-score of 27.6%. Integrating AFRI with PN improved the *F*-score to 35.1%, demonstrating the effectiveness of the AFRI approach. Keywords: prototypical network, few-shot sound event detection

4:00

4pAB8. Threatened and endangered species in southern California and their potential sensitivity to rocket noise: A project overview. Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmoats@byu.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall, Megan R. McCullah-Boozer, Rachel H. Budge (Biology, California State Univ. Bakersfield, Bakersfield, CA), and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The global cadence of rocket launches is increasing. This trend has understandably raised questions about the potential negative impacts the noise from these launches might have on natural resources and ecosystems. Researchers at Brigham Young University and California State University Bakersfield are currently working together on a project that attempts to answer these questions about the threatened and endangered species living at Vandenberg Space Force Base. The base is home to several species of concern including the California least tern, snowy plover, California red-legged frog, tidewater goby, and unarmored three-spined stickleback. To answer questions about these species' sensitivity to rocket noise, continuous recordings at over 30 different locations have been taken for 9 months. These locations include areas deemed ecologically sensitive by base ecologists and are varying distances from launch pads. Already this project has yielded some anecdotal results including instances of bird vocalization stopping during launch noise as well as animal movement correlating with launch events. Presented are a detailed project overview, data collection methods, preliminary observations and results, and plans for more rigorous acoustic and statistical analysis. [Work supported by USSF through USACE.]

Session 4pBAa

Biomedical Acoustics: Bubbles and Ultrasound – Physiological Considerations II

Virginie Papadopoulou, Cochair
Biomedical Engineering, The University of North Carolina at Chapel Hill, 116 Manning Drive, 9004 Mary Ellen Jones Building, CB 7575, Chapel Hill, NC 27599-7575

Nicholas Ovenden, Cochair
Department of Mathematics, University College London, London WC1E 6BT, United Kingdom

Athanasios Athanassiadis, Cochair
Heidelberg University, Im Neuenheimer Feld 225, IMSEAM—AG Fischer, Heidelberg 69120, Germany

Invited Paper

1:00

4pBAa1. Model-dependent modulation of radiotherapeutic efficacy with oxygen microbubbles and ultrasound. Phillip Durham (Univ. of North Carolina at Chapel Hill, 116 Manning Dr., Chapel Hill, NC 27599, pgdurham@email.unc.edu), Quezia Lacerda (Thomas Jefferson Univ., Philadelphia, PA), Mark Borden (Biomedical Eng., Univ. of Colorado, Boulder, CO), John Eisenbrey (Thomas Jefferson Univ., Philadelphia, PA), Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), and Virginie Papadopoulou (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Oxygenation of hypoxic tumors during radiation therapy can increase tumor damage and improve radiotherapeutic efficacy. Lipid-shelled microbubbles can be used with ultrasound for a variety of enhanced therapeutic applications, including drug delivery. Further, the bubbles themselves can be used as carriers for therapeutic gas, such as oxygen. Upon ultrasound activation, oxygen microbubbles (OMBs) release oxygen, improving the oxygenation status of the tumor microenvironment. This process enhances the generation of reactive oxygen species (ROS) during radiotherapy, leading to increased DNA damage and tumor cell death. In developing and evaluating OMBs for the modulation of radiotherapeutic efficacy, we observed model-dependent differences in response to radiation after treatment with oxygen microbubbles. Curiously, in an immune-competent rat fibrosarcoma allograft model, treatment with OMBs and ultrasound was radioprotective, whereas a study in immunocompromised mice bearing CAL27 xenograft tumors oppositely demonstrated radiosensitization and improved anti-tumor efficacy. Some evidence implicates vasodynamic effects on tumor perfusion in the acute radiotherapy window, though additional studies comparing immune competent and deficient mice of the same tumor model also demonstrate differences in outcome following ultrasound and OMB-enhanced radiation therapy.

Contributed Papers

1:20

4pBAa2. Bubble-enhanced ultrasound and radiation in healthy brain tissue: Exploring the parameter space for a novel neurosurgical tool. Stecia-Marie Fletcher (Radiology, Brigham and Women's Hospital/Harvard Med. School, 221 Longwood Ave., EBRC 515b, Boston, MA 02115, sfletcher4@bwh.harvard.edu), Tiago Oliveira (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA), Michael Lavelle (Radiation Oncology, Dana Farber Cancer Inst., Boston, MA), Yongzhi Zhang, Chanikarn Power (Radiology, Brigham and Women's Hospital/Harvard Med. School, Boston, MA), Ross Berbeco (Radiation Oncology, Dana Farber Cancer Inst., Boston, MA), and Nathan J. McDannold (Radiology, Brigham and Women's Hospital, Boston, MA)

Combined microbubble-mediated focused ultrasound (μB-FUS) and radiation therapy (RT) have been used to treat solid tumors. We have observed that μB-FUS+RT also leads to incision-less, targeted ablation in healthy brain tissue. This novel neurosurgical approach may minimize radiation doses (20–70 Gy) needed for radiosurgery and improve the ability to tailor surgery compared to μB-FUS ablation. Here, we studied the effect of μB-FUS and RT on lesioning in healthy rats. The striatum was targeted with pulsed FUS (273 kHz; $\sim$5 $\times$ 5 mm^2; Definity μBs, 20 μl/kg). Acoustic pressures were modulated using a μB emissions-based controller (max 183 kPa). T1- and T2-weighted 7T MRI were used to confirm FUS effects. $\sim$1 h after FUS, RT (3 $\times$ 3 mm^2) was delivered to the target. Lesion progression was monitored using T2-w MRI. 7 treatment groups (n = 4 locations/group) were tested: (1) μB-FUS-only, (2–4) RT-only (8, 12, and 15 Gy), and (5–7) μB-FUS+RT. 72 h after μB-FUS+RT, lesions were seen at all targets for 12 and 15 Gy and 2/4 targets for 8 Gy. Lesions were well colocalized with the overlap of the two treatment modalities, measuring $\sim$3 $\times$ 3 mm^2, and persisted until day 21. Edema in the μB-FUS-only group resolved within 7 days, and no T2 effects were observed in the RT-only groups. Preliminary analyses indicate that mild edema on T2-w MRI immediately following μB-FUS may be predictive of successful μB-FUS+RT treatment outcomes. Ongoing work is aimed at understanding the effect of μB-FUS parameters, and the mechanisms of cell death and lesion progression.

1:40

4pBAa3. Ultrasound-targeted microbubble cavitation-induced Ca^{2+} influx via Piezo1 causes transient gene expression changes. Anurag N. Paranjape (Heart, Lung, Blood and Vascular Medicine Inst., Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15217, anuragnparanjape@gmail.com), Shuai Yuan, Adam C. Straub (Dept. of Pharmacology & Chemical Biology, Univ. of Pittsburgh, Pittsburgh, PA), Xucai Chen, and Flordeliza S. Villanueva (Heart, Lung, Blood and Vascular Medicine Inst., Univ. of Pittsburgh, Pittsburgh, PA)

Background: Ultrasound-targeted microbubble cavitation (UTMC) is a viable approach to overcome challenges in efficient targeted drug delivery by inducing transient endothelial hyperpermeability. However, a deeper understanding of molecular mechanisms and bioeffects of UTMC is essential for clinical translation. Methods and results: Using human coronary artery endothelial monolayers on transwells, we found that UTMC (f 1 MHz; PNP 250 kPa; pulse length 10 μs; pulse interval 10 ms; treatment duration 10 s) caused Ca^{2+} influx, which was required for inter-endothelial gap formation and hyperpermeability. Inhibitor (GsMTx4) and siRNA studies revealed that a significant fraction of Ca^{2+} influx occurred via the mechanosensitive channel Piezo1. To determine gene expression changes following Ca^{2+} influx via Piezo1, we performed RNA-seq in GsMTx4-treated cells post-UTMC. Pathway analysis predicted activation of multiple hyperpermeability- and inflammation-related pathways, including HIF1α, STAT3, ERK/MAPK, Notch, CXCR4, NO, RHO GTPase, and IL-4, 7, 8, 9, 13 signaling, at 3 h post-UTMC, returning to baseline by 24 h. A preliminary study using specific inhibitor WZ811 suggested that CXCR4 might play a key role in regulating UTMC-induced hyperpermeability. Conclusions: Our findings provide important new insights into the mechanisms underlying UTMC-induced hyperpermeability and potential bioeffects, paving the way for optimizing this technology for safe and effective clinical application.

2:00

4pBAa4. Optimizing ultrasound parameters for enhanced sonodynamic therapy: A physical approach to tumor treatment. Giovanni Durando (INRiM, National Inst. of Metrological Res., str. delle cacce 91, Torino 10135, Italy, g.durando@inrim.it), Fabio Saba, Alessandro Schiavi (INRiM, National Inst. of Metrological Res., Torino, Italy), Loredana Serpe, Federica Foglietta, and Roberto Canaparo (Dept. of Drug Sci. and Technol., Univ. of Torino, Torino, Italy)

Sonodynamic therapy (SDT) leverages the energy of ultrasound (US) waves to activate sonosensitizers such as IR-780, offering a localized and targeted approach to anticancer treatment. This study focuses on the physical parameters of ultrasound application, including frequency (1.505 MHz), intensity (0.63–1.66 W/cm^2), and exposure duration (3 min), tailored for both two-dimensional (2-D) and three-dimensional (3-D) pancreatic cancer cell models. The high penetrative capacity of the US ensures precise energy delivery to tumor sites while limiting damage to surrounding healthy tissues. In 3-D spheroid cultures, SDT effectively reduced tumor volume and disrupted structural integrity, driven by mechanical stress and cavitation effects induced by ultrasonic energy. Optimized power settings and duty cycles were essential for enhancing therapeutic outcomes while avoiding thermal damage. The interplay between ultrasound-induced pressure variations and sonosensitizer activation facilitated the generation of reactive oxygen species (ROS), causing localized cytotoxic effects within the tumor microenvironment. These findings demonstrate the potential of ultrasound as a dual-purpose tool for therapy and drug delivery in oncology. By exploiting its non-invasive nature, deep tissue penetration, and high spatial selectivity, SDT provides a promising platform for addressing treatment-resistant tumors, particularly pancreatic cancer. Further refinement of ultrasonic parameters will support its translation into clinical practice.

2:20

4pBAa5. Exploring the role of sonoluminescence in sonodynamic therapy. James McLeod (Dept. of Eng. Sci., Univ. of Oxford, Botnar Inst., Old Rd., Oxford OX3 7LD, United Kingdom, james.mcleod@eng.ox.ac.uk), Luca Bau, Stephanie Walton, Jovana Katrinka Mavrak (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Richard Lane (School of Phys. and Astronomy, Univ. of Glasgow, Glasgow, United Kingdom), John Callan, Anthony McHale (School of Pharmacy and Pharmaceutical Sci., Ulster Univ., Coleraine, United Kingdom), Daniele Faccio (School of Phys. and Astronomy, Univ. of Glasgow, Glasgow, United Kingdom), and Eleanor P. Stride (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Sonodynamic therapy, using ultrasound in combination with microbubbles and drugs such as Rose Bengal to achieve a therapeutic outcome, has been demonstrated as an effective method for delivering targeted cancer treatment. One proposed explanation for how sonodynamic therapy achieves its therapeutic effect is that bubbles cavitating under ultrasonic stimulation produce light, sonoluminescence, which then stimulates photoactive drugs. These drugs then produce reactive oxygen species (ROS) leading to localized cell death. Through repeated measurements using multiple techniques, we have shown that the actual amount of light produced by bubble cavitation is several orders of magnitude lower than the level required to produce a therapeutic effect in photodynamic therapy. Additionally, measurement of the levels of ROS produced during established treatment regimens is insufficient to cause significant effects on cell viability. It is therefore unlikely that sonoluminescence and subsequent ROS generation play any significant role in the therapeutic effect of sonodynamic therapy.

2:40

4pBAa6. Dose-dependent safety and renal ultrasound contrast kinetics of oxygen microbubbles in healthy dogs. Jacob A. Mattern (Div. of Pharmacoengineering and Molecular Pharmaceutics, Eshelman School of Pharmacy, Univ. of North Carolina at Chapel Hill, 9001 Mary Ellen Jones Bldg., 116 Manning Dr., Chapel Hill, NC 27599, jmattern@unc.edu), Phillip Durham (Joint Dept. of Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Kenta Kakiuchi (Biomedical Eng. Program, Univ. of Colorado at Boulder, Boulder, CO), Melissa Caughey (Joint Dept. of Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Joshua B. Currens, Katherine M. Eltz (Joint Dept. of Biomedical Eng. & Dept. of Radiology, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Jocelyn Burkit, Gabriela S. Seiler (Dept. of Molecular Biomedical Sci., College of Veterinary Medicine, North Carolina State Univ., Raleigh, NC), Susan LaRue (Dept. of Environ. and Radiological Health Sci., Flint Animal Cancer Ctr., Colorado State Univ., Fort Collins, CO), Michael W. Nolan (Dept. of Clinical Sci., College of Veterinary Medicine, North Carolina State Univ., Raleigh, NC), Mark Borden (Biomedical Eng., Univ. of Colorado, Boulder, CO), Paul A. Dayton (Joint Dept. of Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Virginie Papadopoulou (Joint Dept. of Biomedical Eng. & Dept. of Radiology, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Oxygen microbubbles (OMB) are lipid-shelled ultrasound contrast agents with an oxygen gas core that show promise as theranostic agents for diagnostic imaging and enhancing radiotherapy efficacy in preclinical rodent models. By improving oxygenation in hypoxic tumor regions, OMBs may increase radiosensitivity. However, their safety, tolerability, and potential to act as vascular imaging contrast agents have yet to be assessed in larger animals. In this study, four healthy beagle dogs received intravenous OMBs at escalating doses 1, 10, and 100 μl/kg at 1-week intervals. Over a 4-week observation period, key physiological parameters, including temperature, pulse, and respiration, were monitored alongside complete blood

counts, blood biochemistry, coagulation panels, and urinalysis. Additionally, the ability of OMBs to enhance contrast during kidney ultrasound imaging was evaluated, and pharmacokinetic parameters were estimated. Across all doses, OMB administration was well tolerated, with no clinically significant changes in vital signs or laboratory values. Ultrasound imaging revealed dose-dependent increases in renal contrast enhancement. Pharmacokinetic analyses confirmed that higher doses prolonged contrast enhancement and improved image quality. This study provides an initial safety and diagnostic benchmark for OMB administration in dogs, laying the groundwork for future investigations into their theranostic potential.

3:00–3:20 Break

Invited Papers

3:20

4pBAa7. Oxygen-sensitive hemoglobin microbubbles: Advancing ultrasound imaging and targeted drug delivery. Bahareh Kian Pour, GHazal Rastegar, Teja Pathour (Bioengineering, UT Dallas, Richardson, TX), Mohammadaref Ghaderi (Bioengineering, UT Dallas, Dallas, TX), Sugandha Chaudhary, Mohammed M. Salman (Bioengineering, UT Dallas, Richardson, TX), Misun Hwang (Radiology, Childrens Hospital of Philadephia, Philadelphia, PA), and Shashank Sirsi (Bioengineering, UT Dallas, 800 W. Cambell Rd., Richardson, TX 75080, shashank.sirsi@utdallas.edu)

The use of ultrasound contrast agents for biosensing physiological parameters such as pressure, pH, and oxygen levels is an exciting, rapidly advancing field. In our recent work, we have developed oxygen-sensitive hemoglobin microbubbles (HbMBs) designed for both imaging and therapeutic applications. These innovative microbubbles feature shells made entirely of bioactive hemoglobin, which undergo structural changes in response to oxygen binding, thereby altering their acoustic properties. Our studies have demonstrated their effectiveness, showing an extended circulation half-life when pegylated and filled with PFB gas, all while maintaining their sensitivity to oxygen. Notably, these microbubbles align with the capabilities of existing ultrasound contrast agents, offering the potential to carry therapeutic payloads. Using focused ultrasound energy, they can be precisely triggered to release these payloads, unlocking new possibilities for targeted drug delivery. This presentation will highlight our recent progress in developing hemoglobin microbubbles and explore the innovative strategies we envision for their future applications.

3:40

4pBAa8. Detection of environmental pH and oxygenation through ultrasound microbubble shell stiffening using nonlinear pulse sequences. Adam Azizi, Bahareh Kian Pour (Bioengineering, Univ. of Texas at Dallas, Richardson, TX), Connor Endersley, Ali Shariq, Caroline de Gracia Lux, Jacques Lux (Radiology, UT Southwestern Medical Ctr., Dallas, TX), Shashank Sirsi (Bioengineering, UT Dallas, Richardson, TX), and Katherine G. Brown (Bioengineering, Univ. of Texas at Dallas, 800 W. Campbell Rd., Richardson, TX 75080, katherine.brown@utdallas.edu)

Ultrasound microbubble (MB) contrast agents, typically consisting of a lipid shell and an inert gas core, are commonly used as tracers to highlight the vascular system. Recently, MBs with compositions suitable to sense specific biological factors, such as pH or oxygen level, have been developed. Studies have shown that MBs can undergo changes in their shell properties, such as stiffness, in response to alterations in their local environment (e.g., pH or oxygen saturation). In this work, we evaluate the response of MBs using B-mode imaging and nonlinear imaging to detect changes in pH and oxygen levels in their surrounding environment. *In vitro* experiments were performed with lipid-shelled MB in solutions with baseline pH values of 7.4 which were adjusted to 5.5, and with hemoglobin-shelled MB in high and low oxygen conditions. The MBs were imaged with a Verasonics Vantage 256 system, equipped with a GE9LD linear transducer at 5 MHz. The results demonstrated a 1.3× to 3× increase in harmonic content with reduced oxygenation, or lower pH, attributed to changes in the MB shell stiffness. These findings also validated simulation predictions. This study demonstrates the potential of using MBs to detect bio-environmental differences and offers a promising direction for further study.

Contributed Papers

4:00

4pBAa9. Acoustic pressure reporting and mapping using antibubbles. Athanasios Athanassiadis (Heidelberg Univ. & Max Planck Inst. for Medical Res., Im Neuenheimer Feld 225, IMSEAM—AG Fischer, Heidelberg 69120, Germany, thanasi@uni-heidelberg.de), Nicolas Moreno-Gomez (Heidelberg Univ. & Max Planck Inst. for Medical Res., Heidelberg, Germany), Albert T. Poortinga (Eindhoven Univ. of Technol., Eindhoven, Netherlands), Fabian Reuter, Hendrik Reese (Otto-von-Guericke Univ., Magdeburg, Germany), Helen Jade (Heidelberg Univ., Heidelberg, Germany), Claus-Dieter Ohl (Otto-von-Guericke Univ., Magdeburg, Germany), and Peer Fischer (Heidelberg Univ. & Max Planck Inst. for Medical Res., Heidelberg, Germany)

Antibubbles are a novel carrier for precise, ultrasound-triggered delivery of fluid or nanoparticulate payloads. These structures combine the high acoustic contrast and strong acoustic response of conventional microbubbles with a significantly higher payload capacity thanks to their large fluid core. The release characteristics of antibubbles can be precisely tuned during fabrication to enable release across a range of pressures spanning a few kPa to several hundred kPa. Moreover, antibubbles can release these payloads either all at once, or incrementally across multiple ultrasonic pulses. This talk will illustrate how such characteristics make antibubbles unique reporters for ultrasonic pressure, and how the fabrication and design determine the antibubble's response. We use high-speed microscopy to highlight the relationship between the antibubble and its released payload. Finally, it will be shown that by properly tailoring the payload, formulation, and supporting matrix, spatial maps of complex acoustic pressure fields can be produced and read out optically, providing real-time diagnostic information about spatially structured ultrasonic fields.

4:20

4pBAa10. Evaluating *in vivo* tissue stiffness of capillaries using Microbubbles Under an Ultrasound Field. Sae Jang (Cardiology, Univ. of Pittsburgh, 200 Lothrop St., Scaife 969, Pittsburgh, PA 15206, jangsk@upmc.edu), Cheng Chen, Xucai Chen (Univ. of Pittsburgh, Pittsburgh, PA), Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada), Spandan Maiti (Univ. of Pittsburgh, Pittsburgh, PA), and Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA)

Introduction: We developed a new technique to study *in vivo* biomechanical properties of capillaries using microbubbles (MBs), ultrasound, and high-speed microscopic imaging. We hypothesized that diabetes stiffens capillaries and that MB acoustic behaviors within capillaries will change as wall stiffness changes. Methods: Type 1 diabetes was induced in rats using streptozotocin. Cremaster muscle was externalized in anesthetized rats. Definity® was intra-arterially injected (40–60 μl boluses). Capillaries were imaged microscopically at 8–12 Mfps, during delivery of one 6–15 cycle ultrasound pulse [f = 1 MHz, PNP = 0.5–2.0 MPa]. Capillary and MB diameters were obtained from image analysis and power spectra were derived. Stress–strain curves were generated using normal stress exerted by MB (derived from the linearized Euler's equation) and circumferential vessel Green strain. Results: MBs in diabetic capillaries (n = 3) displayed greater subharmonic/fundamental power ratio (p = 0.03) and ultraharmonic/fundamental power ratio (p = 0.01) compared with MBs in healthy capillaries (n = 28). Preliminary elastic modulus analysis (n = 9 control and n = 2 diabetic) showed estimated elastic moduli of 1.7 MPa for healthy and 2.2 MPa for diabetic capillaries. Conclusions: Our preliminary data suggest that the MB frequency spectrum under the US may differ in diabetic capillaries, which may provide a basis for a novel approach to diagnose microvascular disease.

4:40

4pBAa11. Cavitation bubble nuclei in healthy versus cancerous 3-D cell cultures. Ferdousi Sabera Rawnaque (Graduate Program in Acoust., Penn State Univ., 210 East Hamilton Ave., Apt. 31, State College, PA 16801, fmr5186@psu.edu) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The safety of biomedical ultrasound depends on controlling cavitation bubbles *in vivo*, yet the spatial location of bubble nuclei in tissues (intracellular versus extracellular) and their behavior in healthy versus cancerous cellular environments remain unexplored. Here we evaluated the spatial location of bubble nuclei and cavitation thresholds in healthy (L6) and cancerous (L8) rat musculoskeletal myoblasts (n = 5 each), cultured in 3-D using PureCol® EZ-Gel scaffolds. A 3.68-MHz focused ultrasound transducer (f# = 1) induced low-density cavitation with pulse durations (PD) of 10–100 μs at 1 Hz PRF and pressures up to p- = 13 MPa. Brightfield microscopy and high-speed photography (20 000 fps) monitored bubble location. Cavitation bubbles ($r \approx 5$ μm) appeared stochastically and exclusively in the extracellular space of the healthy and cancerous 3-D cultures despite significant radiation force. In cancerous L8 cells, no cavitation occurred at PD = 50 μs with pressures up to p- = 13 MPa over 15 pulses; however, PD = 100 μs induced extracellular cavitation at $p \approx 12.3$ MPa. In contrast, extracellular cavitation was detected in healthy L6 cultures at PD = 50 μs at $p \approx 11.8$ MPa, suggesting the potential influence of cell stiffness on cavitation thresholds. Future work will computationally investigate the effects of scaffold and tissue stiffness on bubble formation. [Work supported by NSF CAREER 1943937.]

THURSDAY AFTERNOON, 22 MAY 2025 BALCONY L, 1:00 P.M. TO 5:40 P.M.

Session 4pBAb

Biomedical Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Wave Propagation and Aberration in Complex Media: From Theory to Applications II

Vera A. Khokhlova, Cochair
University of Washington/Moscow State University, Physics Faculty, Moscow State University, Moscow, 119991, Russian Federation

T. Douglas Mast, Cochair
Biomedical Engineering, University of Cincinnati, 3938 Cardiovascular Research Center, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Contributed Papers

1:00

4pBAb1. Comparison of diffusion coefficient measurement through Coherent or Incoherent Backscattering measurement, for the characterization of complex media. François Legrand (MAE, North Carolina State Univ., 1840 Entrepreneur Dr., Eng. Bldg. III, Raleigh, NC 27606, flegran@ncsu.edu), Parniyan Norouzzadeh, and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

This study compares methods to calculate the diffusion coefficient (D) in complex media using Coherent or Incoherent Backscattering (CBS and IBS). In heterogeneous media like lung or bone tissue, conventional ultrasound imaging is inadequate for diagnostics. Quantitative Ultrasound (QUS), including D measurement, offers insights into the "randomness" of such media. Changes in D reflect multiple scattering effects and are associated with an increased intensity of IBS and a reduced intensity of CBS over time. The study evaluates D measurement methods with varying air inclusion proportions in soft tissues, both numerically and experimentally. The three approaches include: (1) traditional CBS with plane wave beamforming for emission and reception, (2) enhanced CBS (eCBS), which leverages phase coherence between backscattered waves in different directions to

improve CBS measurements through amplitude and phase data, and (3) IBS analysis through matrix treatment to separate the incoherent intensity from the CBS peak using a virtual antireciprocity. Numerical simulations demonstrate that eCBS provides more stable and accurate diffusion measurements compared to CBS or IBS. Pending experiments on water-filled sponge phantoms and rat lungs with fibrosis seem to confirm the reliability of eCBS. This method holds promise for improved diagnostics in complex media, particularly where conventional imaging is insufficient.

1:20

4pBAb2. Estimating vascular density by using quantitative ultrasound in microbubble enhanced vessel networks. Parniyan Norouzzadeh (MAE, North Carolina State Univ., 1840 Entrepreneur Dr., Raleigh, NC 27606, pnorouz@ncsu.edu), François Legrand (MAE, North Carolina State Univ., Raleigh, NC), Jadyn Cook, Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

This study investigates the use of quantitative ultrasound techniques for characterizing tumor-related angiogenesis. Neo-vessel density serves as an established biomarker of breast cancer malignancy, making it crucial to develop methods for its quantification. Our approach involves analyzing scattering from ultrasound contrast agents within vascular networks. We assess whether parameters derived from plane wave imaging and full synthetic aperture data acquisition effectively estimate vessel density. Healthy rat kidneys were used as a model for vascular networks. Following anesthesia, microbubbles were injected through the tail vein at an infusion rate of 1.5e8 bubbles/min. Ultrasound data acquisition was performed using a Verasonics L-11-5v linear array transducer connected to a Verasonics 256 Vantage NXT scanner. A full synthetic aperture approach was applied, utilizing all 128 transducer elements to transmit two-cycle Gaussian pulses at a central frequency of 7.6 MHz. Backscattered signals were recorded across all elements, producing a 3-D Inter-Element Response Matrix (IRM). The Diffusion constant and multiple scattering-related parameters were calculated by separating coherent and incoherent backscattered intensities and analyzing the temporal growth of the diffusive halo. Acoustic angiography images were used for comparison, and light sheet imaging served as the ground truth for validating vascular density measurements.

1:40

4pBAb3. Differentiable beamforming in ultrasound autofocusing with large aperture array. Minho Song (Radiology, Stanford Univ., 3165 Porter Dr., Palo Alto, CA 94304, minhos@stanford.edu), Walter A. Simson, Josquin Foiret, Jeremy Dahl, and Katherine Ferrara (Radiology, Stanford Univ., Palo Alto, CA)

Phase aberration in ultrasound imaging, often caused by heterogeneous sound speed distribution in tissue, introduces errors in estimating relative time delays during conventional delay-and-sum beamforming, resulting in degraded image quality. Differentiable beamforming in ultrasound autofocusing (DBUA) has recently emerged as a promising method to adaptively estimate spatial sound speed distributions in tissue by minimizing a loss function based on focusing criterion, such as the common midpoint phase error. However, implementing DBUA for large aperture arrays with extended fields of view remains computationally demanding due to increased element numbers and longer acquisition samples, even with graphical processing units (GPUs) for parallel computing. This study investigates refinement techniques for DBUA using a 384-element extended aperture phased array (88 mm aperture, 0.23 mm interelement pitch (0.37λ), 2.5 MHz operating frequency), processed on an Nvidia Titan RTX GPU. Specifically, we optimize control parameters, including the f-number, learning rate, and the size and distribution of the patches (i.e., selective regions for focusing criterion calculations). The results demonstrate over a 30% improvement in computational efficiency while achieving comparable imaging quality, with 1% accuracy in sound speed estimation in phantom studies, advancing DBUA toward practical usage in abdominal imaging. [Work supported by NIH R01CA258807, R01EB033967, and R01CA271309.]

2:00

4pBAb4. Wavefield correlation imaging in arbitrary media with inherent aberration correction. Scott J. Schoen (Radiology, Harvard Med. School and Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, sschoenjr@mgh.harvard.edu), Brian Lause (GE HealthCare, Waukesha, WI), Marko Jakovljevic (Radiology, Massachusetts General Hospital, Boston, MA), Rimon Tadross, Mike Washburn (GE HealthCare, Waukesha, WI), and Anthony E. Samir (Radiology, Harvard Med. School and Massachusetts General Hospital, Boston, MA)

Ultrasound imaging can suffer from aberration—distortions caused by acoustic propagation effects not accounted for by the beamforming algorithm. These can be corrected using known medium properties, but for delay-and-sum (DAS) algorithms, delay corrections must be computed for every pixel and element, which becomes computationally expensive, especially in volumetric imaging. These delays also need to be recomputed for each new geometrical registration of the probe. Recently, wavefield correlation imaging (WCI) algorithms, first developed for seismic applications, have been adapted for ultrasound. In this work, we integrate the heterogeneous angular spectrum method (HASM)—a spectral propagation technique designed for passive acoustic mapping that accounts for spatially varying media—into WCI to enable aberration correction during image formation. This method reduces computational complexity by roughly two orders of magnitude compared to corrected DAS. Numerical plane wave imaging simulations in anatomically relevant acoustic environments show a 66% improvement in resolution (CI: 54%–79%, $p < 0.01$) compared to the uncorrected case. Heterogeneous WCI thus shows appreciable promise for enhancing imaging in aberration-challenged scenarios, particularly in point-of-care devices with limited computational resources.

2:20

4pBAb5. A forward model for estimating phase aberration in ultrasound imaging. Yuan Xu (Dept. of Phys., Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, yxu@ryerson.ca)

Phase aberration in ultrasound images is caused by inaccurate information in the sound speed distribution in the medium and can result in image distortion, such as shape change and position shifting of the imaged objects. Various methods, including cross-correlation-based methods, have been applied to the distorted images to estimate phase aberration. In this paper, we first propose that the position shifting induced by the phase aberration causes the estimated phase aberration to be inaccurate. Then, we propose an equation to relate the estimated phase aberration to the true one as the basis of a forward model for estimating phase aberration. Finally, we show that the estimated phase aberration can be predicted or modeled from the true phase aberration. Considering phase aberration as a function of the array element position, the theory shows that both the constant term and linear term of the true phase aberration will be canceled in the estimated phase aberration as they result solely in predictable position shifting. Field II simulations and data from tissue mimicking phantom were used to validate the proposed theory. [This work was supported by the Natural Sciences and Engineering Research Council of Canada (NSERC), the Canada Foundation for Innovation (CFI), and Toronto Metropolitan University.]

2:40

4pBAb6. Desktop 3-D-printed metamaterials for MHz-range ultrasound devices. Alireza Tadibi (Mech. & Mater. Eng., Univ. of Cincinnati, 230 Ludlow Ave., Apt. 4, Cincinnati, OH 45220, tadibiaa@mail.uc.edu), Yehia Zakaria, and Ahmed Allam (Mech. & Mater. Eng., Univ. of Cincinnati, Cincinnati, OH)

Ultrasonic metamaterials enable precise control of acoustic waves, offering new capabilities in diagnostic and therapeutic ultrasound applications. Yet, realizing metamaterials for practical ultrasound frequencies requires increasingly fine fabrication resolutions limiting existing solutions to low frequencies. In this work, we leverage common Fused Deposition Modeling (FDM) 3-D printing to build ultrasonic metamaterials that can operate at frequencies up to 5 MHz. To overcome the limited resolution of FDM, we utilize the micro-scale air gaps that arise during the printing of

solid structures as our repeating unit cell. The gaps' shape and size are engineered by controlling 3-D-printing parameters such as infill density, infill pattern, and layer height. A numerical analysis of the repeating unit cell predicts the ultrasonic behavior of these materials, enabling optimization for biomedical devices. Experimental validations demonstrate that the phase velocity of Polylactic Acid (PLA)-based metamaterials can be continuously programmed within 20% of near-solid PLA by adjusting the geometry of the unit cell. Moreover, the FDM-printed metamaterials exhibit lower attenuation compared to polymers fabricated using state-of-the-art light-based 3-D-printing techniques. We envision the developed materials to have direct applications in building ultrasonic devices such as low-cost ultrasound imagers, patient-tailored aberration correction, and focused ultrasound treatments.

3:00–3:20 Break

3:20

4pBAb7. Tissue heterogeneity and weak nonlinearity using a phased array transducer for clinical transrectal treatment. Jorge Torres (LabTAU, INSERM, 151 Cr Albert Thomas, Lyon 69003, France, jorge.torres@inserm.fr), Virgil Accary, Victor Delattre, Thomas Payen, and Cyril Lafon (LabTAU, INSERM, Lyon, France)

Phase aberration correction is critical for optimizing the therapeutic efficacy of HIFU in various clinical applications. While much research has focused on transcranial focal therapy, transrectal treatment of prostate cancer presents unique challenges. In this context, the acoustic beam must traverse coupling liquid, rectum, and fatty tissue before reaching the prostate, which can introduce phase distortions that affect treatment precision. Using the Focal One® system (EDAP TMS, France) as a research platform, we evaluated and refined methods to predict and compensate for these effects in controlled *in vitro* settings. To address the complexities beyond the assumptions of Rayleigh integrals, we developed a comprehensive numerical framework that includes: (i) acoustic holography to accurately reconstruct the transducer's vibration patterns; (ii) a scaling factor for peak pressure estimation from electrical power; and (iii) simulations incorporating heterogeneous media and weak nonlinearity to improve focal shift predictions. This work complements existing phase aberration correction strategies and offers a computational–experimental toolkit for enhancing HIFU protocol customization. These findings contribute to advancing precision in prostate cancer treatments, underscoring the importance of integrating physics-informed corrections in clinical practice.

3:40

4pBAb8. Leveraging the parametric array effect for transcranial focused ultrasound interventions. Pradosh Pritam Dash (Mech. Eng., Georgia Inst. of Technol., Molecular Sci. and Eng. (MOSE) Bldg., Rm. 4135, 901 Atlantic Dr. NW, Atlanta, GA 30318, ppdash@gatech.edu) and Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA)

Slightly different high-frequency beams can interact nonlinearly to generate a low-frequency "difference frequency," a phenomenon known as the parametric array effect. While this physical effect can offer sub-wavelength targeting in the brain, creating it through the skull during trans skull focused ultrasound (FUS) procedures has remained unexplored. Here, we show focused ultrasonic fields can produce different frequencies in brain tissue by exploiting the skull's inherent nonlinearity. We find that the drop in difference-frequency amplitude —particularly around 100 kHz— can be reliably detected through the skull and serve as an active acoustic feedback to perform precise acoustic hologram-to-skull registration. Furthermore, by examining changes in the difference-frequency signal, we propose a noninvasive strategy to identify ventricular enlargement in hydrocephalus, where excess cerebrospinal fluid can be distinguished from surrounding brain tissue. This approach has the potential to guide early diagnostic assessments and real-time monitoring of draining excess cerebrospinal fluid from the brain. More broadly, our findings underscore the potential of the parametric array effect in guiding transcranial FUS interventions. [This work is funded by the National Science Foundation CMMI Award 1933158.]

4:00

4pBAb9. Simulating performance of a through-transmit aberration correction method for transcranial focused ultrasound. Cooper L. Donovan (Vanderbilt Univ., 2301 Vanderbilt Pl., PMB 350938, Nashville, TN 37235, cooper.l.donovan@vanderbilt.edu) and Charles F. Caskey (Radiology, Vanderbilt Univ. Medical Ctr., Nashville, TN)

Transcranial focused ultrasound (tFUS) is a promising neuromodulation tool but typically demands subject-specific aberration correction based on pre-acquired CT images. A method that does not rely on CT is desirable. Riis *et al*. developed a method, called relativistic through-transmit (RTT), that corrects by comparing through-skull signals to free-field signals from two opposing transducer arrays positioned on either side of the head. RTT showed promise in deep-brain regions, but its robustness under noise, steering, and positioning uncertainties is not well characterized. To evaluate RTT's performance, we performed 2-D k-wave simulations modeling commercially available transducers in a through-transmit arrangement across a non-human primate skull. To avoid instability in published iterative phase correction methods, we corrected by pseudoinverse. To quantify efficacy, we measured corrected through-skull pressure and focal position. In an ideal case (no uncertainties, etc.), RTT restored 102% of the intended pressure at spatial peak (2.3 mm focal shift) and 92% at target. Performance varied between targets, restoring 56%–78% pressure when steering 10 mm off-axis. In 20 simulations where we applied random transducer position uncertainties up to 4 mm, RTT delivered 81%–108% pressure (mean = 95%, s.d. = 7%). This work provides the first theoretical analysis of a novel aberration correction method and the groundwork for methodological improvements.

4:20

4pBAb10. Simulating focused ultrasound propagation through heterogeneous biomedical materials with volume-surface integral equation methods and hierarchical matrix compression. Alberto Almuna Morales (Inst. for Mathematical and Computational Eng., Pontificia Universidad Católica de Chile, Av Vicuña Mackenna 4860, Santiago, Región Metropolitana 7820436, Chile, alberto.almuna@uc.cl), Danilo Aballay (Inst. for Mathematical and Computational Eng., Pontificia Universidad Católica de Chile, Santiago, Región Metropolitana, Chile), Pierre Gélat (Dept. of Surgical Biotechnology, Univ. College London, London, United Kingdom), and Elwin van 't Wout (Inst. for Mathematical and Computational Eng., Pontificia Universidad Católica de Chile, Santiago, Región Metropolitana, Chile)

Focused ultrasound has gained importance in cancer therapy and neuromodulation because it offers a non-invasive treatment that can reach malignant tissue at high precision. Fast and accurate simulation tools are essential to improve safety guidelines and patient-specific treatment planning. The goal is guiding sufficient acoustic energy toward the focus to achieve ablation while sparing healthy tissue in the beam path. We have already achieved realistic simulations with our open-source OptimUS library using the Boundary Element Method (BEM). The BEM is a powerful algorithm to simulate high-frequency acoustics in unbounded domains that scatter at objects with high material contrasts, like soft tissue and bone, and only needs surface meshes at material interfaces. However, the BEM is limited to scatterers with constant density and speed of sound. To investigate the influence of bone heterogeneities on the focus aberrations, we developed a Volume Integral Equation and coupled it with the BEM. To improve computational efficiency and reduce memory footprint, we developed a hierarchical matrix compression technique for the dense system matrices. Our innovations allow for simulating ultrasound scattering at ribcage models for transcostal ultrasound and transmission through skull slabs for transcranial ultrasound at operating frequencies and with material data taken from biomedical images.

4:40

4pBAb11. Ultrasonic evaluation of internal features in additively manufactured SS316L: Impact of process-induced microstructure heterogeneity. Harshith Kumar Adepu (Mech. Eng., Purdue Univ., 500 Central Dr., B36, West Lafayette, IN 47906, adepu@purdue.edu), Jacey Birkenmeyer, Partha Pratim Pandit (Mech. Eng., Purdue Univ., West Lafayette, IN), and Luz D. Sotelo (Purdue Univ., West Lafayette, IN)

Process parameter selection in additive manufacturing (AM) is not standardized. Linear energy density (ED_L = Power/Scanning speed), while commonly used, is non-unique which leads to different thermal histories and microstructures. This variability can significantly impact the performance of AM components, particularly those with internal structures. Ultrasonic nondestructive evaluation (NDE) methods are sensitive to microstructure and material properties. In this study, we assess the ability of ultrasonic NDE to resolve internal structures in AM components and evaluate the influence of microstructural heterogeneity. A systematic protocol was developed to map the viable parameter space for laser powder bed fusion (LPBF) of SS316L using a Lumex Avance 25 system. Process parameters were varied by adjusting scanning speed and laser power while maintaining a constant linear energy density to induce distinct microstructures. Samples were fabricated in two groups: solid specimens and specimens with internal structures, with each internal structure sample having a corresponding solid counterpart. Ultrasonic measurements were compared to intended geometries and X-ray CT scans to evaluate the effects of process-induced microstructural heterogeneity on feature resolution. This work highlights the potential and limitations of using ultrasound to characterize internal features in AM components, providing insights into the impact of microstructure heterogeneity on ultrasonic wave propagation.

5:00

4pBAb12. Quantifying the effect of cooling conditions on 3-D printed PLA meso-structure and acoustic properties. Partha Pratim Pandit (Mech. Eng., Purdue Univ., 585 Purdue Mall, West Lafayette, IN 47907, pandit18@purdue.edu), Anna Keim, Harshith Kumar Adepu, Monique McClain, and Luz D. Sotelo (Mech. Eng., Purdue Univ., West Lafayette, IN)

One of the most popular additive manufacturing (AM) techniques for producing thermoplastic polymers is fused filament fabrication (FFF). Nonetheless, there are still a lot of issues with the printed parts' quality. Most of the research being done to improve the quality of parts is limited to process parameter optimization and part quality analysis utilizing imaging techniques. These attempts are insufficient to define and forecast local material properties within the parts because of the inconsistent quality among printers and printed parts. Additionally, imaging techniques reveal how print parameters affect part quality; yet the majority of research in this field has been print-focused, leaving a knowledge gap about the connection between real printer operating conditions and final part quality. Polylactic acid (PLA) cubes with different fan activation layers were printed in the present study. The cubes underwent observable and controllable deformations as a result of shifting *in situ* cooling conditions. The mechanical characteristics and defect content were measured using ultrasonic and X-ray CT nondestructive evaluation techniques, respectively. We used this information to better understand how cooling history and material properties are related.

5:20

4pBAb13. Development of a dynamic acoustic phantom for simulating human tissue. amirhossein Yazdkhasti (Orthopaedics, Cedars-Sinai Medical Ctr., 8700 Beverly Blvd, West Hollywood, CA 90069, a.h.yazdkhasti@gmail.com), Joseph H. Schwab, and Hamid Ghaednia (Orthopaedics, Cedars-Sinai Medical Ctr., Los Angeles, CA)

Human body phantoms are crucial for developing and optimizing medical devices. In acoustic research, phantoms are more essential due to the complexity of acoustic propagation patterns and the invasive nature of measurements. This study presents the design, fabrication, and evaluation of a series of acoustic phantoms that replicate the temporal and acoustic properties of human tissue. These phantoms are developed to study the propagation patterns of acoustic waves through complex anatomical structures and the interactions between medical implants and external acoustic stimulation. The fabrication process begins with the segmentation of magnetic resonance imaging (MRI) data to model anatomical structures, followed by the 3-D printing of bones and the molding of soft tissues. The synthetic tissues are molded using gelatin-based materials with acoustic properties similar to different tissues. The final phantom assembly integrates the 3-D-printed bones with the molded tissue components. As a case study, a lower-leg phantom is fabricated. The static acoustic performance is assessed using surface-mounted transducers positioned at various locations, while dynamic performance is evaluated under mechanical loading applied to the superficial posterior compartment using a mechanical testing system (MTS). Finite element simulations are conducted to model acoustic wave propagation and validate the experimental results. The findings highlight the efficacy of the proposed phantoms in understanding acoustic behavior in biological tissues.

Session 4pED

Education in Acoustics and Musical Acoustics: Acoustics Education Research for Newbies: The Science of Teaching and Learning

Kimberly A. Riegel, Cochair
Physics, Farmingdale State College, 2350 Broadhollow Road, Farmingdale, NY 10803

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—1:15

Invited Papers

1:20

4pED1. I think this is working! Now what? From a classroom strategy to rigorous research. Robert E. Krakehl (Phys., Manhasset Secondary School, SUNY Farmingdale, 200 Memorial Pl, Manhasset, NY 11030-2320, Rkrakehl@gmail.com)

As educators, we often develop innovative strategies and witness positive changes in our classrooms. But how can we move beyond anecdotal evidence to rigorously evaluate and share what's working? This session bridges the gap between teaching practice and research, providing educators with a roadmap to turn classroom experiences into meaningful studies. Participants will explore practical steps to design research grounded in their teaching, including formulating research questions, collecting and analyzing data, and navigating the Institutional Review Board (IRB) process. With a focus on balancing practicality and academic rigor, this session highlights ways to document impact, contribute to the broader field of education, and inspire continuous improvement. Whether you are new to research or looking to refine your approach, this talk will equip you with the tools to transform classroom successes into evidence-based contributions that advance both practice and pedagogy.

1:40

4pED2. The productive classroom: Enhancing teaching and learning. Robert E. Krakehl (Phys., Manhasset Secondary School, 200 Memorial Pl, Manhasset, NY 11030-2320, Rkrakehl@gmail.com)

Educators are constantly striving to improve instruction and engage students more effectively. With countless innovative approaches available, the challenge lies in navigating and implementing the wealth of resources that can support these efforts. This session focuses on building a productive classroom by introducing practical, evidence-based tools for teaching and learning. Grounded in cognitive science, classroom management strategies, and over a decade of experience across diverse educational settings, this talk explores actionable ways to bridge research and practice. Participants will learn how to identify, adapt, and integrate these resources to enhance student outcomes and foster dynamic learning environments. By connecting theoretical insights with real-world applications, this session empowers educators to create classrooms where students thrive.

2:00

4pED3. Review of the current state of pedagogical research in acoustics. Kimberly A. Riegel (Phys., Farmingdale State College, 2350 Broadhollow Rd., Farmingdale, NY 10803, riegelk@farmingdale.edu)

Humans have pursued new and innovative methods for teaching for most of human history. While early teaching was largely informal and unstructured, the past 150 years have seen the field become more formalized, with the establishment of journals dedicated to reporting educational research findings. The creation of the Bureau of Education in 1867, which later evolved into the Department of Education, marked a significant push toward formalizing the evaluation of educational methods. Numerous educational research organizations and journals were founded in the early 20th century, each contributing to the systematic study of teaching practices. Additionally, discipline-specific journals and organizations, such as the American Association of Physics Teachers (AAPT), the American Society for Engineering Education (ASEE), and the National Association of Biology Teachers (NABT), have emerged to focus on methods within their respective fields. Research on acoustics education is still spread across various disciplines and journals. Now the Acoustical Society of America (ASA) is creating a more formal technical interest in the effectiveness of acoustic education to consolidate some of the work being done. This presentation will explore the current state of acoustics education research across multiple disciplines, providing a survey of key studies and accepted metrics used to evaluate teaching methods.

2:20

4pED4. Surveying physics education research for impacts on acoustics education. Jack A. Dostal (Dept. of Phys., Wake Forest Univ., P.O. Box 7507, Winston Salem, NC 27109, dostalja@wfu.edu)

Physics education research (PER) is a broad, robust field. While not an exhaustive list, PER addresses individual concepts in physics (including waves and acoustics), curricular interventions and assessments, methods to gather and analyze data, active research in the classroom, and studies into the theory and process of learning. PER can help us to understand why specific concepts in physics are difficult for students to understand, and what we might be able to do about it. Surveys, interviews, and in-class activities help us to learn why certain concepts are difficult for students. Research-based curricula can provide ways to address those concepts. There are also more theoretical articles that study the nature of student thinking without being tied to any one physics topic. In this talk, I will survey physics education research, identifying some useful articles and curricula with acoustics and sound in mind. These and many other "points of entry" can provide impact in your own classroom and may be helpful in conducting your own studies with students.

2:40

4pED5. What can acoustics education research learn from physics education research? Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

As the field of acoustics education research becomes established, we can benefit from examining other discipline-based education research fields for examples and guidance. Notably, physics education research (PER) is a well-established field with a rich history to draw from. PER has demonstrated that student-centered learning environments and active learning strategies are more effective than traditional lecture-based content delivery. These active learning strategies include interactive engagement techniques, such as group discussions, problem-solving sessions, and hands-on activities, all of which can also be used in acoustics education. Additionally, PER has developed concept inventories and attitude surveys to measure aspects of student learning and understanding. Acoustics education research could follow the example of PER by adopting these proven methodologies. Furthermore, given the highly interdisciplinary nature of acoustics, exploring cross-discipline approaches could provide unique advantages in understanding how students learn acoustics. By drawing on the insights and methodologies from PER and embracing the interdisciplinary nature of acoustics, we can advance acoustics education research and develop more effective teaching strategies that enhance student learning and engagement.

3:00

4pED6. Techniques for performing acoustics education research. Gordon P. Ramsey (Phys. Dept., Loyola Univ. Chicago, Loyola University Chicago, Chicago, IL 60660, gramsey@luc.edu)

The ultimate goal of acoustics education research (AER) is to improve instruction and learning in acoustics at all levels. The outcomes should include an understanding and appreciation of acoustics at the fundamental level, a deeper comprehension at the intermediate level and preparation for acoustics careers at the ultimate level. Unlike other acoustics research involving nature and phenomena, education research has its unique methods of dealing with humans. Physics education research (PER) has been an active research area for over 30 years and has yielded valuable techniques to improve the way physics and related courses are taught. These techniques can be used by AER to achieve the goals stated above. I will discuss the approaches used by PER researchers to define goals, design a program, gather necessary materials, perform and publish the research, and implement the changes discovered as applied to the AER community.

3:20–3:40 Break

3:40

4pED7. A PhD student's perspective of educational research in acoustics. Noah J. Parker (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, np.acoustics@gmail.com) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Doctoral research in acoustics typically involves experimental, computational, and theoretical analysis of technical problems. Research focused on understanding how students learn acoustics (acoustics education research) requires a familiarity with an entirely different knowledge and skill set. This presentation introduces education research from the perspective of an acoustics graduate student who is pursuing a dissertation involving acoustics education research. Key topics will include: the distinctions between quantitative, qualitative, and mixed methods approaches; navigating the Institutional Review Board (IRB) process; and the design of a study using focus groups to explore how undergraduate students engage with interactive animations as learning tools. This talk will emphasize the importance of moving beyond descriptive accounts of teaching practices to employing research methods to rigorously evaluate learning outcomes.

Contributed Papers

4:00

4pED8. Build-your-own, hands-on workshops: Teaching acoustics, engineering, and programming with something to take home. Heui Young Park (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hkp5188@psu.edu)

Students studying acoustics often come from various backgrounds, from engineering to music. While courses in acoustics can help teach fundamental concepts to students regardless of their background, learning tangible, applicable skills can be difficult, especially for those without prior exposure to specific skills and tools. At Penn State, a series of student-run hands-on, build-your-own workshops were implemented to help all students learn translatable skills such as computer-aided design (CAD), rapid prototyping, electronics, microcontrollers, and programming. For example, a set of workshops focused on building a mini dodecahedron speaker. Students learned to use a collaborative, online-based tool for CAD, and the basics of

3-D printing. Students also learned to solder and to calculate resistance in series and parallel. Each student can take home the finished product, so that students can not only learn the skills associated with the workshops but also have something physical to take home. The workshops can be designed to be low-cost and could also extend to teaching other related skills such as acoustic measurements and speaker characterization. These hands-on workshops implement active learning methods and universal design for learning and have potential applications to not just undergraduate or graduate students, but to students at all levels and backgrounds.

4:20

4pED9. Engineering education research within acoustics courses for undergraduate architectural engineering students. Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, vigeant@engr.psu.edu) and Karen High (Dept. of Eng. & Sci. Education, Clemson Univ., Clemson, SC)

Highlights of two engineering education research projects with undergraduate architectural engineering (AE) students will be presented. Study 1 was conducted in a third-year AE course, AE309, Introduction to Architectural Acoustics. Weekly reading quizzes and three peer-review homework assignments were incorporated into the course with the hypothesis that these elements would help support students' learning. This hypothesis was evaluated through the collection of survey data, which revealed that about 60% of the students felt that the reading quizzes increased their preparedness for lectures and 35% perceived the peer-review assignments to increase their understanding of the assigned concepts. Study 2 was conducted in an upper-level AE course, AE458, Advanced Architectural Acoustics and Noise Control. The theoretical foundation of Study 2 consists of Vygotsky's social constructivist theory of learning and a student engagement framework. Inclusive teaching strategies, including required regular practice and collaborative learning activities, were incorporated into the course. The impact of these strategies on cognitive belonging and engagement was explored using a mixed-methods approach. Details of the experimental design will be presented to introduce educational research methods for those new to the field. Study 1 was supported by Penn State's Leonhard Center; Study 2 was supported by NSF Award 2407013.

4:40–5:10 Panel Discussion

THURSDAY AFTERNOON, 22 MAY 2025 GALERIE 6, 1:20 P.M. TO 4:00 P.M.

Session 4pMU

Musical Acoustics and Computational Acoustics: String Instruments II

Vasileios Chatziioannou, Cochair
Department of Music Acoustics, University of Music and Performing Arts Vienna,Anton-von-Webern-Platz 1, Vienna 1030, Austria

Mark Rau, Cochair
Music and Theater Arts & Electrical Engineering and Computer Science, Massachusetts Institute of Technology, 77 Massachusetts Ave., Bldg. W18-Room, Cambridge, MA 02139

Montserrat Pàmies-Vilà, Cochair
Department of Music Acoustics - Wiener Klangstil (IWK), University of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, mdw - Inst. 22, Vienna, 1030, Austria

Contributed Papers

1:20

4pMU1. Between harpsichord and grand piano paradigms—Acoustical differences of modern fretted zithers. Simon Linke (Hamburg Univ. of Appl. Sci., Finkenau 35, Hamburg 22081, Germany, simon.linke@haw-hamburg.de), Jonathan G. Fiegl (Anton Bruckner Privatuniversität, Nürnberg, Germany), Patrick Kontopidis, Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Hamburg, Germany), and Robert Mores (Hamburg Univ. of Appl. Sci., Hamburg, Deutschland, Germany)

Fretted zithers, also known as concert zithers, originated in the Alpine region during the mid-19th century. As a relatively young instrument, their construction is still evolving. This study examines the acoustic properties of two modern zithers, which, while having similar shapes, have very different underlying designs. The bending wave propagation on both instruments is measured using a miniature impact hammer to excite the instruments at the bridge and a laser Doppler vibrometer to measure time-dependent velocity responses. The detected acoustical differences between the two zithers are compared to similar measurements from grand piano and harpsichord soundboards to judge their musical meaning. The traditionally constructed zither shows some similarities to the vibrational behavior of a grand piano, which may reflect their shared origins in the Romantic period and, thus, similar musical aesthetics. Conversely, the second zither exhibited some acoustic parallels to harpsichords, possibly reflecting the needs of modern musicians moving away from traditional zither music.

1:40

4pMU2. Acoustic comparison of period instruments on piano. Masanobu Miura (Collection for Organology, Kunitachi College of Music, 5-5-1, Kashiwa-cho, Tachikawa, Tokyo 1908520, Japan, miura.masanobu@kunitachi.ac.jp)

Here introduces an acoustic comparison of period instruments on piano. The Collection for Organology, Kunitachi College of Music (COKCM), has been collecting various kinds of musical instruments, in particular, period instruments. The COKCM focuses on collecting and investigating academic research concerning musical instruments. One of the most important efforts is to collect the old piano, or pianoforte, which is explicitly distinguished from the contemporary piano. Since Western music in the Baroque and Roman eras are along with the era of the birth and development of the piano, the music of the period is frequently required to be played using the piano of the period. The sound of the old piano is, however, rarely discussed in the field of acoustic research. This research focuses on the difference in acoustics of the old piano. Several types of old pianos, such as John Broadwood (1820), Pleyel & Comp. (1848), and two kinds of S. Erard (1850) and Johann Schanz (1820) were used in a recording experiment. The sound is recorded and plotted in a two-dimensional plane by using MDS. The effect of the so-called "second soundboard," located on the strings in the piano is discussed from the spectral viewpoint.

2:00

4pMU3. Spectral analysis of a chord plucked on an acoustic guitar and comparison to theoretical predictions based on the Fourier transform equations. Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu) and William Ramone (Phys., Skidmore College, Tampa, FL)

Theoretically, the acoustics of string instruments are analyzed by calculating the harmonics of a vibrating string through solutions to the wave equation and through Fourier transforms. But the reality of observing the spectrums predicted directly from the recording of a plucked string is generally unsatisfactory and is often ignored. Students who are trying to grasp the physical concepts that are predicted are often confused by the spectra produced in a typical recording. This paper looks at the spectral analysis produced by a chord plucked on an acoustic guitar and its comparison to theoretical predictions based on the Fourier Transform Equations. What can be determined by observing the spectral data? Can we gain information about the chord played and perhaps how it was played by examining the spectral data? The spectral analysis of an A minor chord played on an acoustic guitar is used as an example of Fourier's Theorem and verification of the Fourier Transform Equation applied to plucked strings. While the original objective of the study was to compare the spectrums produced in various room environments, upon analyzing the data, it was realized that the harmonic spectrum for each string is clearly shown for each note of the chord.

2:20–2:40 Break

2:40

4pMU4. Experimental measurements of the directivity of string instruments played using pizzicato. Laura Austin (Grove City College, 100 Campus Dr., Grove City, PA 16127, AustinLJ22@gcc.edu), Hanna Pavill (Brigham Young Univ., Provo, UT), Dallin T. Harwood (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Micah Shepherd (Brigham Young Univ., Provo, UT)

The directivity of musical instruments plays an important role in the sound that is produced when a musician plays an instrument. Previous studies have measured the directivity of string instruments when they are bowed. However, there has not been a thorough investigation of the directivity of these instruments when they are plucked. High-resolution directivity data for pizzicato (plucked) notes of the violin, viola, cello, and double bass were collected using a multi-capture directivity measurement system (DMS). The measured data were then processed to create balloon plots of the directivity of the fundamental frequency and strongest harmonics of each of the open strings. During this presentation, the pizzicato directivity data will be presented and compared between instruments (such as between violin and viola) to illustrate important similarities and differences in the directivities of the instruments when they are played using pizzicato. The data collected for pizzicato notes will also be compared to equivalent data from a previous study that looked at the directivity of the same instruments when they are bowed.

3:00

4pMU5. Influence of pitch, dynamics, and vibrato on the emotional characteristics of violin sounds. Wenyi SONG (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong NA, Hong Kong, wsongak@cse.ust.hk), Man Hei LAW (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Anh Dung Dinh (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, New Territories, Hong Kong), and Andrew B. Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

This study explores how pitch, dynamics, vibrato levels, and their interactions influence the emotional perception of violin tones. Using 32 violin notes, 62 participants provided valence and arousal ratings on a 9-point Likert scale and binary responses for 16 emotional categories. Results show that pitch was the most influential factor, with higher pitches increasing arousal. Vibrato levels had multifaceted effects, enhancing arousal while reducing valence. Dynamics played a less prominent role but revealed that louder sounds increased valence and unexpectedly decreased arousal. Interaction effects demonstrated that vibrato extent's influence on arousal was moderated by both pitch and dynamics. Additionally, an analysis of low-level acoustic features, such as loudness and vibrato rate, revealed significant correlations with emotional responses. For instance, higher vibrato rates were associated with increased arousal, while attack time had subtle effects on valence. Observations across the 16 emotional categories highlighted distinct quadrant-specific trends. For example, high-arousal emotions were primarily influenced by loudness and vibrato, while low-valence emotions were more sensitive to attack and decay times. These findings provide nuanced insights into the interplay of acoustic features in shaping emotional experiences, contributing to research in music cognition and emotional processing.

3:20

4pMU6. Measuring the frequency response of a violin—What Does it Mean? Chris Rogers (Mech. Eng., Tufts Univ., Medford, MA, chris.rogers@tufts.edu)

Over the last few decades, a group of luthiers and physicists (and a programmer or two) have been working on better characterizing the violins using a number of advanced tools to try and gain insight into what makes an excellent violin. Radiation and mobility measurements, as well as full modal analyses, have been coupled with hearing tests to better understand the instrument. My role in all this has been to help develop an analysis toolkit (freeware)—and my goal with this talk is to have more of a discussion on how best to interpret these results. We will play (in real time) with convolving the frequency response function with the signal from an electric violin and compare it to how the original instrument sounds, and then remove parts of the frequency response function and compare that to the original instrument with various modes suppressed.

3:40

4pMU7. On the use of tonewood leftovers for elastic parameter estimation of soundboards. Sebastian Duran (Industrial Eng., Univ. of Bologna, Via Zamboni 33, Bologna 40126, Italy, sebastian.duran2@unibo.it), Henna Tahvanainen (Industrial Eng., Univ. of Bologna, Bologna, Italy), Ludovico Ausiello (School of Elec. and Mech. Eng., Univ. of Portsmouth, Southampton, United Kingdom), and Michele Ducceschi (Industrial Eng., Univ. of Bologna, Bologna, Italy)

Estimating the elastic properties of tonewood is an integral part of the instrument making process. Luthiers can use experience-informed methods, such as tapping or bending the actual instrument soundboards, or more state-of-the-art techniques involving measurement equipment and computational methods. These methods often require rectangular pieces, and for that purpose, leftover tonewood can be used. This paper investigates the

feasibility of using leftover pieces in the workshop to estimate the elastic parameters of instrument soundboards. The verification is done by comparing the vibrational patterns obtained by the numerical models computed with the estimated elastic parameters to those of the actual soundboards. Measurements are performed by two different methods that represent different ends of the technology involved in the measurement: experimental modal analysis with a laser Doppler vibrometer and Chladni pattern detection with an inexpensive exciter.

THURSDAY AFTERNOON, 22 MAY 2025 GALERIE 5, 1:20 P.M. TO 4:20 P.M.

Session 4pNS

Noise: Exposure Response and Community Tolerance Level

Ken Kaliski, Chair
RSG, 55 Railroad Row, White River Junction, VT 05001

Invited Papers

1:20

4pNS1. Abstract withdrawn.

1:40

4pNS2. Modeling community noise tolerance with non-Gaussian distributions. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil) and Chris L. Pettit (Aerosp. Eng., U.S. Naval Acad., Annapolis, MD)

Previously, a hierarchical (multi-level) model was shown to predict exposure-annoyance curves in excellent qualitative agreement with noise survey data aggregated from multiple communities D. K. Wilson *et al*. [J. Acoust. Soc. Am. 142(5), 2905–2918 (2017)]. The hierarchical model includes variations in exposure and annoyance occurring at the individual and community levels. The community-level annoyance variations in the hierarchical model are conceptually similar to the community tolerance level (CTL) concept proposed by Fidell *et al*. [J. Acoust. Soc. Am. 130(2), 791–806 (2011)]. Wilson *et al*. modeled the exposure and tolerance variations in the hierarchical model using Gaussian distributions. However, recent research indicates that noise exposure can have significantly non-Gaussian behavior; in particular, noise-level distributions tend to be "heavy-tailed," indicating that loud events occur more often than would be predicted by a Gaussian distribution. Similarly, human tolerance to noise could have significant non-Gaussian behavior. This presentation examines the impact on the exposure-annoyance curves of extending the previous hierarchical model to non-Gaussian distributions, specifically to the family of Lévy-alpha stable distributions.

2:00

4pNS3. Updating ISO 1996-1: Standardizing on the Community Tolerance Level for evaluating long-term noise annoyance. Ken Kaliski (RSG, 55 Railroad Row, White River Junction, VT 05001, ken.kaliski@rsginc.com), Stephen Keith, David Michaud (Health Canada, Government of Canada, Ottawa, ON, Canada), and Douglas M. manvell (DMdB Sound Advice, Charlottenlund, Denmark)

The ISO 1996-1 standard, "Acoustics—Description, measurement and assessment of environmental noise, Part 1: Basic quantities and assessment procedures" provides, in part, methodologies used to evaluate long-term noise annoyance. Prior to 2016, this was done using a sigmoid fit to the Schultz curve where the day-night sound level (Ldn) was adjusted to account for the effects of sound characteristics on the percentage of highly annoyed. As part of the ongoing ISO 1996-1 revision, the ISO working group is considering focusing noise annoyance assessment on the Community Tolerance Level (CTL). This is a minor modification in that the new sigmoid curve is based on loudness, but it explicitly suggests that community-related parameters are important to annoyance. These approaches make it easier to compare different communities' tolerance to noise using a single number descriptor. It will also discuss approaches other than CTL and where these may be appropriate. This paper will describe our consideration of this approach, the benefits of using CTL for standardized comparisons, level adjustments, and other sound sources and characteristics

2:20

4pNS4. Burden of disease from road traffic noise and associations with socioeconomic marginalization in Toronto, Canada. Tor H. Oiamo (Dept. of Geography and Environ. Studies, Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, tor.oiamo@torontomu.ca), Anette K. Bølling, and Gunn Marit Aasvang (Air and Noise, Norwegian Inst. of Public Health, Oslo, Norway)

This paper examines the burden of disease from traffic noise as disability-adjusted life years (DALYs) in Toronto, Canada, and its association with four dimensions of socioeconomic marginalization based on the Ontario Marginalization Index. Previous research found that 7% of all Canadians and up to 36% of residents in certain areas of Toronto are highly annoyed by traffic noise. Exposure to traffic noise has also been associated with ischemic heart disease in Toronto and other urban areas. However, it is not clear how the disease burden due to traffic noise in Toronto compares to other cities and how it is related to socioeconomic indicators. Building facade noise levels from the US FHWA Traffic Noise Model were applied to estimate attributable DALYs from high annoyance, sleep disturbance and ischemic heart disease. A total of 25 725 DALYs at a rate of 942 per 100 000 were estimated to be lost annually to traffic noise exposures in Toronto. We also found statistically significant correlations between the estimated DALYs and socioeconomic marginalization at the dissemination area level. The highest correlations were observed for residential instability ($r = 0.44$) and ethnic concentration ($r = 0.22$). The estimated burden of disease rate for traffic noise in Toronto is among the highest for cities currently reported in the literature.

2:40–3:00 Break

Contributed Papers

3:00

4pNS5. Thurstone's law of comparative judgment applied to 5-point verbal annoyance responses from community noise studies. Matthew Boucher (NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, matthew.a.boucher@nasa.gov)

Studies on community annoyance with aircraft noise often use a 5-point verbal scale in which the categories are assumed to be equally spaced (ISO/TS 15666). Responses are dichotomized into highly annoyed (4 and 5) or not highly annoyed (1, 2, and 3) responses, and generalized linear models are used to find relationships between the noise dose and annoyance response. A Thurstone approach avoids dichotomization of responses and interprets each response as a comparison between the selected verbal response and the boundaries of the 5-point verbal categories. From this, standard implementations of the Law of Comparative Judgment yield a latent annoyance scale. This approach gives a finite value to the percent highly annoyed for each response, even for values less than 4, which may be a modeling advantage for datasets with a low prevalence of annoyance. Thurstone analysis is also used to evaluate the assumption of equally spaced categories. Applying the Law of Comparative Judgment to 5-point verbal annoyance responses may be advantageous because it considers the psychological difference between perception (an internal, private judgment) and the external perceptual response.

3:20

4pNS6. Concepts for the incorporation of number effects into the assessment of annoyance. Andrew Christian (Appl. Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov)

Recent psychoacoustic results from both laboratory and community studies have suggested that the tradeoff in annoyance between a small number of loud noise events and a large number of quieter events may not scale linearly with acoustic energy as is implied by most noise metrics in use today. This presentation discusses several mathematical constructions that could be used to introduce curvature into the tradeoffs made by noise metrics. Broadly, these strategies can be broken into two classes: The first uses the absolute number of events to introduce nonlinearity into a metric's response. This could be used to not only penalize large numbers of events but also protect against a small number of extremely loud events being incorrectly compared to many moderate ones. The second class takes the time structure of the events into consideration—are the events evenly spread out over an entire day, or are they clumped together? Noise events that are clustered in time may lead to a sense of "relentlessness" in the mind of the observer, and a penalty based on this effect would induce similar curvature into the tradeoff function for certain schedules of events.

3:40

4pNS7. The sound of pain: Exploring health impacts of noise exposure. Joanne Solet (Harvard Med. School, The Cambridge Hospital 1493 Cambridge St., Cambridge, MA 02139, Joanne_Solet@HMS.Harvard.edu)

Noise exposure has emerged as a significant environmental and public health challenge, influencing health far beyond auditory impacts. This research review explores relationships between noise and pain perception, including physiological, neurological, and psychological mechanisms. Chronic noise exposure elevates pro-inflammatory markers including IL-6 and TNF-alpha, driving neuropathic pain and systemic sensitization. Through disruption of restorative sleep, noise impairs emotion regulation, elevates inflammatory markers, and amplifies pain sensitivity, contributing to chronic pain syndromes. Interdisciplinary strategies, including public health campaigns and legislation aimed at integrating noise control and prevention into clinical, housing and work environments, offer opportunities for protecting health and well-being. This research review offers actionable insights to motivate leaders in acoustics, pain medicine, sleep health, and occupational and environmental medicine to address this growing challenge of noise-related health impacts.

4:00

4pNS8. Nearfield directivity analysis during the NG-19 launch. Carson F. Cunningham (Phys., Brigham Young Univ., Provo, UT 84602, carsonfcunningham@gmail.com), Micah Shepherd (Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In August 2023, the Antares 230 launched successfully for the NG-19 resupply mission to the International Space Station. Acoustic measurements were taken at various locations around the launch pad, ranging from 60 to 200 m away from the vehicle. The analysis focused on azimuthal and polar angles to investigate the vehicle's sound directivity during the launch. Spectral data were evaluated as functions of frequency, angular position around the pad, and orientation relative to the vehicle. A spatio-spectral analysis was conducted to interpret the data effectively. Initial findings reveal that maximum sound levels are associated with wider angles relative to the plume for stations closer to the source. The peak frequency at all stations was observed to be between 20 and 60 Hz, which is common for vehicles of this size. Although proximity to the rocket complicates distinguishing between angles, making directivity analysis challenging, a spatio-spectral analysis best reveals the spectral features of the noise.

Session 4pPAa

Physical Acoustics and Computational Acoustics: Infrasound II

Philip S. Blom, Cochair
Earth & Environmental Sciences, Los Alamos National Laboratory, P.O. Box 1663, M/S F665, Los Alamos, NM 87545

David N. Green, Cochair
AWE Blacknest, AWE Blacknest, Brimpton, RG7 4RS, United Kingdom

Stephen Arrowsmith, Cochair
Southern Methodist University, 819 Lake Terrace Drive, Dallas, TX 75218

Contributed Papers

1:00

4pPAa1. Thermospheric infrasound from the 2024-Sep-18 Toropets ammunition depot explosions: signal variability and source characterization. David N. Green (AWE Blacknest, Brimpton RG7 4RS, United Kingdom, dgreen@blacknest.gov.uk) and Alexandra Nippress (AWE Blacknest, Reading, United Kingdom)

Infrasound in the rarefied thermosphere (altitudes >85 km) exhibits waveform stretching and wavefront steepening due to non-linear propagation effects. Such characteristics are a function of both the acoustic pressure amplitudes and the thermospheric propagation path length. If the propagation path can be accurately estimated, the extent of waveform stretching can be used to evaluate near-source pressure amplitudes, providing information on the acoustic source size. On 2024-September-18 a series of explosions occurred at the Toropets ammunition depot, in Russia, reportedly due to a Ukrainian drone attack. Recordings of >20 thermospheric phases were recorded at a distance of 339 km on the IS43 six-element microbarometer array near Dubna, Russia. Across a period of 4 h the group velocity (celerity) of these arrivals increases from 238 to 252 m/s, likely indicating a continuous reduction in the propagation path turning height. We compare these observations with numerical propagation modeling results using state-of-the-art meteorological specifications and consider how the observations can assist in better characterizing the propagation path. The implications for source size estimation and associated uncertainties will be discussed. [UK Ministry of Defence © Crown Owned Copyright 2025/AWE.]

1:20

4pPAa2. A survey of signals and noise at high-frequency infrasound arrays in Nevada and South Korea. Jonathan Y. Reiter (Roy M. Huffington Dept. of Earth Sci., Southern Methodist Univ., 3225 Daniel Ave. 221, Dallas, TX 75206, jojoyehreiter@gmail.com), Alexandra Reyes, Junghyun Park, Jacob Clarke, and Stephen Arrowsmith (Roy M. Huffington Dept. of Earth Sci., Southern Methodist Univ., Dallas, TX)

The development of the International Monitoring System infrasound network, combined with additional deployments of regional infrasound arrays, has led to enhancements in our understanding of acoustic signals and noise in the 0.1–5 Hz range over the last two decades. The goal of this study is to explore the properties of signals and noise at higher frequencies: from high-frequency (HF) infrasound to low-frequency (LF) audible (1–30 Hz). By embedding two new HF infrasound arrays within existing regional arrays in Nevada and South Korea, we are exploring the signals observed at these frequencies and assessing how such HF arrays can complement traditional regional infrasound arrays. In addition to observing regional infrasound signals that vary seasonally due to the stratospheric winds, repeating local sources are identified at each array, including mining explosions and operational signals such as machinery. Notable detections from the OSIRIS-REx reentry, missile and satellite launches, and the 2024 M_W 7.5 Noto earthquake are highlighted. In addition to cataloging signals, we extend noise models to the HF infrasound/LF audio range, assessing coherent and incoherent noise contributions. We find that the new HF arrays extend the capability of traditional regional arrays by detecting new local signals and providing new information on regional signals, and they open the understanding to study HF infrasound propagation at local distances.

1:40

4pPAa3. Studying infrasound propagation in the middle atmosphere with Upper Atmosphere Icosahedral Non-hydrostatic model: parameterization and characterization of gravity waves with the Multi-Scale Gravity Wave Model. Samuel Kristoffersen (CEA, DAM, DIF, Chem. du Ru, Bruyeres-le-Chatel F-91680, France, samuel.kristoffersen@unb.ca), Constantino Listowski (CEA, DAM, DIF, Arpajon, France), Georg-Sebastian Voelker (Leibniz Inst. for Baltic Sea Res., Rostock, Germany), Ulrich Achatz (IAP, Goethe Univ., Frankfurt, Germany), Julien Vergoz, and Alexis Le Pichon (CEA, DAM, DIF, Arpajon, France)

Infrasound signals are used to monitor various anthropogenic and natural sources. To determine precise source locations and energy, an accurate wind and temperature model from the surface up to the lower thermosphere is necessary, hence operational NWP products are of great importance for routine infrasound monitoring activities. However, many of these models focus on tropospheric conditions, and the middle and upper atmosphere, where the relevant infrasound waveguides for long-range propagation are found, is not well represented. UA-ICON is an upper-atmosphere version of the ICOsahedral Non-hydrostatic weather and climate model (ICON) that provides modeled atmospheric parameters up to the thermosphere. From an infrasound perspective, small-scale perturbations—most notably gravity waves—can have a large impact on propagation due to the effects on both the background winds and temperatures, hence on the acoustic waveguides, but also due to the small perturbations they produce, which cause partial reflections of acoustic waves. Therefore, the transient-3-D Multi-Scale Gravity Wave Model (MSGWaM) was used within UA-ICON to produce accurate background conditions and predict the global gravity wave activity. We will present the methodology used to generate the wind and temperature gravity wave perturbation profiles, and analysis of infrasound propagation using these gravity wave realizations.

2:00

4pPAa4. Fast-Field Program modeling to validate long range detection. Michael J. White (U.S. Army Engineer Res. and Development Ctr., US Army ERDC/CERL, 2902 Farber Dr., Champaign, IL 61822-1072, michael.j.white@usace.army.mil), Richard D. Costley, and Sarah McComas (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

A large explosion was detonated at Fort Johnson, in west-central Louisiana, as part of a test to support explosives storage safety. The explosion was detected 256 km away with a five-element infrasound array at the U.S. Army Engineer Research and Development Center (ERDC) campus in Vicksburg, MS. The initial acoustic arrival was followed by a much larger arrival approximately 100 s later. Ground-to-Space (G2S) atmospheric profiles (Hetzer *et al.*, 2019) and a ray-tracing model were used to investigate the paths and times of arrival. The G2S profiles were then input into a Fast-Field Program (FFP) model to determine arrival times for comparison. The FFP method will be reviewed and the modeling results presented, along with the measurements. [C. H. Hetze *et al.*, The NCPA-G2S request system (2019). https://g2s.ncpa.olemiss.edu. Accessed 2024-07-29.]

2:20

4pPAa5. Seismo-acoustic detection and localization of iceberg collapse in Atka Bay, Antarctica. Gil Averbuch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #11, Woods Hole, MA 02543-1050, gil.averbuch@whoi.edu), Aymeric Houstin, Daniel P. Zitterbart, Julien Bonnel (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), John A. Collins (Dept. of Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), and Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrecht, Netherlands)

On January 12, 2022, shortly after 9:12 PM UTC, scientists working at the Neumayer-III scientific station in Atka Bay, western Antarctica, observed an iceberg collapse at the edge of the ice-covered bay. The iceberg had a triangular shape where each of its edges was $\sim$300 m long. The collapse generated seismic waves propagating through the sea ice and ice shelf and acoustic waves propagating in the ocean and atmosphere. These waves were recorded by a collection of seismic and acoustic instruments deployed on land, on the ice shelf, and underwater. In this study, we analyze the observed seismo-acoustic wavefield recorded in all three media. A combination of the seismic and underwater acoustic observations suggests at least two phases of the collapse separated by 16 s. Seismic observations show that the first arrival corresponds to a longitudinal P wave followed by a dispersed surface wave and multipath coda waves. Array processing results of the infrasound data show two overlapping arrivals, possibly a combination of direct propagation and low-altitude refraction or reflection. Finally, two localization techniques placed the iceberg within less than 300 m from its ground-truth location. These unique observations highlight the added value of continuous seismo-acoustic monitoring for studying local ice-related dynamics.

2:40

4pPAa6. Numerical developments for estimating the sensitivity of infrasound to atmospheric parameters: Applications to atmospheric structure imaging. Solène Gerier (ISAE-Supaero, 10 Ave. Marc Pélegrin, Toulouse 31400, France, solene.gerier@isae-supaero.fr), Roland Martin (Géosciences Environnement Toulouse - Observatoire Midi-Pyrénées, Toulouse, France), and Raphael F. Garcia (ISAE-Supaero, Toulouse, France)

Infrasound, low-frequency acoustic waves that can propagate over hundreds of kilometers, can become a key tool to study the Earth's atmosphere. As infrasound propagates, it incorporates information about the atmospheric structure, including temperature and wind variations. As a result, this information can be used to reinforce the numerical weather prediction models, allowing a better representation of the vertical structure of winds and atmospheric temperature. In particular, this study focuses on atmospheric imaging through the development of numerical tools to characterize the sensitivity of infrasound waveforms to the atmospheric domain. To this end, full waveform inversion methods, typically used in seismology, are exploited. These methods estimate the sensitivity of waveforms to geophysical domain parameters using the adjoint method. By adapting this method to infrasound waves, it is shown, for the first time, an estimate of infrasound waveform sensitivities to wind and wave speed variations. The method is applied to the real case of an explosion at the Hukkakero site (Finland), recorded on a CTBT infrasound network. This study also addresses the influence of the source on sensitivity kernels. Given encouraging results on sensitivity, this method is integrated into an inverse problem framework to recover the atmospheric structure in synthetic cases.

3:00–3:20 Break

3:20

4pPAa7. Infrasound uncertainty propagation: Ensuring traceability from the laboratory to the field. Samuel Kristoffersen (CEA, DAM, DIF, Chem. du Ru, Bruyeres-le-Chatel F-91680, France, samuel.kristoffersen@unb.ca), Severine Demeyer (LNE, Trappes, France), Paul Vincent (CEA, DAM, DIF, Bruyeres-le-Chatel, France), Dominique Rodrigues (LNE, Trappes, France), Alexis Le Pichon (CEA, DAM, DIF, Arpajon, France), Michaela Schwardt (BGR, Hannover, Germany), Nicolas Fischer (LNE, Trappes, France), and Franck Larsonnier (CEA, DAM, DIF, Bruyeres-le-Chatel, France)

Confidence in the quality of infrasound measurements is at the heart of the operational requirements linked to the detection and assessment of geophysical and industrial events. The entire measurement process, from laboratory calibration to the field, must be considered to estimate the confidence level of the measurement through the associated uncertainty. As part of the European Infra-AUV (metrology for low-frequency sound and vibration) project, a field calibration campaign was performed to complete the calibration chain to the sensors in the field. This paper presents a methodology to obtain traceable measurements of the infrasound wave parameters, taking into account the entire traceable calibration chain and other uncertainty sources arising from a thorough analysis of the measurement process. We present an *in-situ* calibration method of infrasound sensors used in conjunction with the Gabrielson on-site calibration method to provide field traceability of measurements under varying environmental conditions. In the context of this application, the resulting uncertainties in the back azimuth, between 0.05° and 7°, and trace velocity between 0.2 and 60 m/s (from high to low frequency), were predominantly due to the Gabrielson phase uncertainty. The amplitude uncertainty of approximately 0.2 dB also had significant contributions from the temperature and pressure susceptibilities.

3:40

4pPAa8. Capturing atmospheric dynamics with cross-correlation functions: Insights from infrasound propagation during the April 8, 2024 Solar Eclipse. Ketan Singha Roy (Earth Sci., Southern Methodist Univ., 3225 Daniel Ave. Heroy 262, Dallas, TX 75205, ksingharoy@smu.edu), Stephen Arrowsmith, Brian Stump, Chris Hayward, and Junghyun Park (Earth Sci., Southern Methodist Univ., Dallas, TX)

The Earth's atmosphere is a highly dynamic system influenced by periodic and transient phenomena, such as diurnal cycles, seasonal changes, thunderstorms, and solar eclipses. This variability is especially pronounced in the Atmospheric Boundary Layer (ABL), where rapid fluctuations in temperature and wind speed/direction significantly affect the propagation of infrasound waves. Traditional numerical models often fail to fully capture this complexity due to the ABL's inherent spatiotemporal variability. In this study, we employed cross-correlation functions (CCFs) to analyze the atmospheric response to infrasound waves. A 24-h experiment conducted at Southern Methodist University, Dallas, Texas, from April 8 to 9, 2024, during diverse atmospheric conditions—including a total solar eclipse—generated a unique dataset. Chirp signals emitted by a fixed infrasound source were recorded by an array of sensors at various distances, allowing detailed CCF analysis of source-receiver signals. The results revealed significant changes in signal travel time and amplitude, particularly during the solar eclipse, highlighting the ABL's transient response. These findings demonstrate that CCFs effectively capture the dynamic evolution of the atmosphere, providing insights that surpass the capabilities of numerical models. This study underscores the importance of experimental techniques in advancing infrasound propagation research and atmospheric studies.

4:00

4pPAa9. Development of synthetic infrasound signature models from window of opportunity observations. Aprameya Satish (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, aprameya.satish@gtri.gatech.edu), Sounak Das (Georgia Inst. of Technol., Atlanta, GA), Nicholas Breen, Michael Fitzpatrick, and Alessio Medda (Georgia Tech Res. Inst., Smyrna, GA)

Between May 21 and May 22, 2024, researchers from Georgia Tech Research Institute deployed an infrasound sensing array at Cabrillo National Monument and by the shoreline at Torrey Pines, San Diego, California to measure the infrasound signatures of sources of interest such as rotorcrafts, jets, and large boats. This source library was developed to facilitate aircraft detection and classification using infrasound measurements. Pictures of potential targets with the corresponding time-stamps were taken for visual identification and data annotation. In post-processing, the recorded time-series from the infrasound array were associated with the target pictures and delay-and-sum beamforming was used to maximize the signature signal-to-noise ratio (SNR) across the observation window. Using the spectral characteristics from the annotated data, a source signature model capable of generating the time-series for an acoustic source in motion relative to a receiver was developed and tested. Synthetic acoustic data constructed using the source signature model were used to augment the sparse measurement dataset, with the goal of improving target classification performance. The effectiveness of this approach was assessed by comparing the results of classifiers trained with and without dataset augmentation to identify the class of the source signature, i.e., the type of vehicle producing that signature.

4:20

4pPAa10. Modeling and analysis of anomalous infrasonic arrivals in the stratospheric shadow zone. Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., P.O. Box 1663, M/S F665, Los Alamos, NM 87545, pblom@lanl.gov)

Infrasound propagation through the lower- and middle atmosphere is strongly impacted by the stratospheric winds associated with the circumpolar vortex. Combined temperature and wind gradients in the lower stratosphere can produce sufficient refraction to form an acoustic waveguide which is typically characterized by direct ensonification 180–240 km from the source. Between a few tens of km and the inner edge of this ensonification region, an acoustic shadow zone is predicted into which geometry ray paths do not enter. However, in several cases, anomalous infrasonic signals have been observed in this region. Such arrivals have been attributed to scattering from wind shear and other fine-scale structures in the lower stratosphere. Observations of such arrivals were made during the recent Large Surface Explosion Coupling Experiment (LSECE) for stations 140 km downwind of a pair of 1-ton equivalent TNT chemical surface explosions. These arrivals exhibit slower celerities than observed stratospheric refracted paths and apparent velocities that increase later in the arrival wavetrain. Analysis of these arrivals and simulation using ray tracing with imposed reflection layers will be presented.

4:40

4pPAa11. Fundamental frequency normalization for target identification and classification. Lorin Hendricks (GSL, ERDC - USACE, 3909 Halls Ferry Rd., Vicksburg, MS 39180, Lorin.Hendricks@usace.army.mil)

During operation, most types of vehicles produce fundamental frequencies in the infrasound (below 20 Hz) and low-end audible ranges (20–400 Hz). These fundamental frequencies are then repeated as harmonics at integer multiples of the fundamental frequency. These frequencies are generally tied to the revolutions per minute (RPM) of the engine and are mostly produced by the ignition of the cylinders in an internal combustion engine and the air impact of the blades from propellers or rotors. However, the constant variation of the environment, the engine's RPM, and velocity changes (and resulting Doppler shift) during normal operation result in difficulty in identifying and classifying based on acoustical signals. Fortunately, the ratio of the first few harmonics remains relatively stable. Additionally, atmospheric absorption is roughly proportional to frequency allowing these first few harmonics to propagate much farther. This paper will demonstrate techniques for the identification of tonal sources by utilizing an ambient estimation, refinement of the fundamental frequency estimation by combining harmonics, and normalization of the spectrum for future identification and classification. [Permission to publish was granted by the Director, Geotechnical & Structures Laboratory, U.S. Army Engineer Research and Development Center.]

5:00

4pPAa12. A new self-calibrating infrasound sensor design. Carrick Talmadge, Hank Buchanan (NCPA, Univ. of Mississippi, Oxford, MS), Jonathan Parsons, David Harris, Chad Williams (Hyperion Technol. Group, Inc., Tupelo, MS), and JaqWan Works (Hyperion Technol. Group, Inc., 545 Commerce St., Tupelo, MS 38801, jworks@hyperiontg.com)

We will present the results of testing a new self-calibrating infrasound sensor design, developed by Hyperion. This design integrates a sound source and system controller logic into the body of a Hyperion infrasound sensor. Both analog and digital versions of this design were constructed and tested. The calibrator produces a sinusoidal pressure signal with amplitude and frequency specified by the user into the back volume of the infrasound sensor. The calibration controller corrects for static pressure and temperature effects on the pressure source, and the system can be operated fully remotely. Results of testing of this system will be presented along with a discussion of possible improvements of this design, including improvements in methodology which would make this system more directly traceable in a metrological sense.

Session 4pPAb

Physical Acoustics: General Topics in Physical Acoustics I

Carl R. Hart, Cochair
U.S. Army Engineer Research and Development Center, Cold Regions Research and Engineering Laboratory, 72 Lyme Road, Hanover, NH 03755

Olivia J. Cook, Cochair
Penn State, 212 Earth Engineering Sciences, University Park, PA 16801

Contributed Papers

1:00

4pPAb1. Mapping thermal phase transformations in NiTi using immersion ultrasonic testing. Olivia J. Cook (Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci., University Park, PA 16801, ojc3@psu.edu) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Nickel-titanium (NiTi) shape memory alloys display reversible, thermoelastic martensitic transformations in response to applied stress or cooling. Methods such as digital image correlation (DIC) have been used to map the transformations under loading, providing spatially resolved information about strain accumulation. However, thermal transformations are monitored using differential scanning calorimetry (DSC), which probes only a small volume and cannot investigate spatial evolution. This study proposes a novel approach using ultrasonic testing to monitor the R-phase transformation *in situ* in Ni-rich NiTi through measurements of longitudinal wave speed and attenuation in a temperature-controlled immersion water bath. By mapping the spatial evolution of acoustic properties during transformation, this technique aims to track transformation fronts and identify local variations in behavior on the bulk scale. This methodology could extend beyond shape memory alloys to other materials systems where understanding thermal phase transformations is crucial for performance optimization, or where the temperature dependence of acoustic properties can reveal information about microstructure.

1:20

4pPAb2. Anisotropy of acoustic attenuation in KBr, NaBr, and RbBr crystals. Farkhad Akhmedzhanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-Plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, 33 Durmon yuli St., Tashkent 100125, Uzbekistan, akhmedzhanov.f@gmail.com), Nodir Makharov, and Jakhongir Kurbanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-Plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, Tashkent, Uzbekistan)

The anisotropy of the attenuation coefficient of acoustic waves, effective elastic constants, and effective anharmonicity constants in KBr, NaBr, and RbBr crystals has been investigated. It is shown that in these crystals, practically the same dependence of the attenuation coefficient of longitudinal acoustic waves on the direction of propagation in the (110) plane is observed. It is established that in the studied crystals, the effective anharmonicity constants determining the magnitude and anisotropy of attenuation of longitudinal and transverse acoustic waves along the [100] direction change linearly with a change in the radius of the K, Na, and Rb cations.

1:40

4pPAb3. Dirac-like spectrum for surface waves in an ultrasonic crystal. Nicholas Gangemi (Physical Acoust., U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, nicholas.gangemi@nrl.navy.mil), Caleb F. Sieck (Code 7160, U.S. Naval Res. Lab., Washington, DC), Joseph Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), Jeffrey Baldwin, Steven Liskey, Aaron Edmunds, William Wilson, Douglas Photiadis, and Bernard Matis (Physical Acoust., U.S. Naval Res. Lab., Washington, DC)

Absent of dissipation, wave energy transport occurs with the group velocity, and in this talk we demonstrate how to exploit this principle to resolve the Dirac cone at the Brillouin-zone edge for ultrasonic surface waves in a hexagonal lattice of resonant cavities. Wave propagation measured along varying real-space lattice angles resolves dispersion along the Dirac cone in reciprocal space where the band structure gradient is oriented along the angle for which energy propagation is measured. Through the Dirac point, we measure group and phase velocities, and the Dirac-point frequency, in agreement with finite-element modeling. The results afford the study of Dirac cone evolution in the development of alternative acoustic systems, and demonstrate how one-dimensional line-scan measurements can resolve targeted reciprocal space dispersion about a specific high-symmetry point for a two-dimensional crystal.

2:00

4pPAb4. Anisotropy of acoustic and acousto-optic parameters in paratellurite crystals. Farkhad Akhmedzhanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-Plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, 33 Durmon yuli St., Tashkent 100125, Uzbekistan, akhmedzhanov.f@gmail.com) and Ulugbek Saidvaliev (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-Plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, Tashkent, Uzbekistan)

The anisotropy of the velocity, the acoustic attenuation coefficient, and the acousto-optic quality factor in paratellurite crystals during the propagation of acoustic waves along and near the symmetry axes of the second and fourth order has been studied in detail. It is shown that for small deviations of the acoustic wave vector direction (no more than 4 deg), the use of approximation calculation formulas yields values of acoustic and acousto-optic parameters that coincide with the experimental results. For angles equal to and greater than 4 deg, the values of these parameters must be determined using standard formulas. The results obtained can be useful in developing acousto-optic devices used to convert or control signals in optical and fiber-optic communication lines.

2:20

4pPAb5. Characterization of graphite nodules in nodular cast iron via multimodal ultrasonic scattering measurements using a rosette configuration. Olivia J. Cook (Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci., University Park, PA 16801, ojc3@psu.edu), Andrew J. Gavens (Naval Nuclear Lab., Niskayuna, NY), Andrea P. Arguelles, and Christopher M. Kube (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Nodular cast iron's complex microstructure is characterized by spherical graphite nodules embedded within a polycrystalline matrix, presenting unique opportunities for ultrasonic scattering measurements due to microstructural heterogeneity. This work focuses on two grades of cast iron (65-45-12 and 80-55-06) tested in an ultrasonic immersion setup using a custom 3-D-printed "rosette" fixture. Four spherically focused 10 MHz transducers were placed to collect different scattering modes, utilizing longitudinal and transverse waves in both in-plane and out-of-plane configurations. Cylindrical specimens (diameter 52 mm and thickness 12.6 mm) were raster scanned in 0.25 mm steps with the ultrasonic rosette, followed by analysis of time-dependent spatial variance over the entire sample area and within local regions. Microscopy provided quantitative mapping of nodule properties on top sample surfaces for correlation with scattered wave measurements. Bulk analysis showed the spatial variance of scattered signals revealed significant sensitivity to microstructural variations between grades despite subtle variations in nodule properties. Local analysis demonstrated a spatial correlation between ultrasonic scattering and nodule size distribution, with locations containing larger nodules and wider size distribution having higher spatial variance in ultrasonic scattering. These results demonstrate the potential of multimodal ultrasonic measurements for quantitative microstructural characterization at both bulk and local scales.

2:40–3:00 Break

3:00

4pPAb6. Non-destructive testing of lithium-ion batteries via analysis of bending modes. Simon Montoya-Bedoya (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 2515 Speedway, Austin, TX 78712, simonmontoyabedoya@utexas.edu), Tyler McGee, Amelie D. Nguyen, Ofodike A. Ezekoye, and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Lithium-ion batteries are crucial for portable electronics, electromobility, and stationary energy storage, playing a critical role in global decarbonization goals. Tracking battery performance during their lifetime ensures reliability, as various degradation mechanisms affect their operation. These changes may alter mechanical properties and thus understanding the complex relationship between electrochemistry, heat transfer, and mechanical properties therefore remains a key research challenge. Elastodynamic inspection methods, such as ultrasonic and vibrational analysis, have shown promise in detecting mechanical changes under varying states of charge (SOC) and state of health (SOH). Recent research has demonstrated the shift in the fundamental resonance frequency is a reliable metric of the SOC and SOH of Nickel-Manganese-Cobalt (NMC) pouch cells. This study presents an analysis of flexural modes for NMC cells at 0% and 100% SOC over 80 charge–discharge cycles. We employ spatial filtering to extract and enhance the response of the first three modes. We observe a correlation between the resonance frequency and the SOC/SOH for all the modes explored. The trends in resonance frequency and quality factor versus cycle from the data are explored and we propose model-based methods to extract insights regarding the evolution of mechanical properties that exploit higher modes as a function of charge level and aging.

3:20

4pPAb7. Porosimeter measurements with a sample-based calibration. Cody M. Best (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@erdc.dren.mil), and Christopher J. Donnelly (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

Porosity is a fundamental transport property that describes the relative volume of voids contained within a material. For an air-saturated material, the voids accessible from the exterior are known as the open porosity. Measurement methods based on gas expansion provide direct, nondestructive measurement of open porosity with high accuracy. All gas expansion methods rely on Boyle's Law for an isothermal process. A system, based on prior work, was designed, analyzed, and used to measure the open porosity of consolidated steel ball bearings, glass beads, filter foams, and sintered plastics. Compared to mass density or water imbibition estimates, the average relative error of measured open porosity is less than one percent. Crucial to obtaining this accuracy is calibrating the instrument with precisely measured aluminum cylinders, which permits an adjustment to the effective residual volume contained within the test chamber.

3:40

4pPAb8. Unconsolidated granular materials sound absorption model using sound propagation theories in porous media. Yousif Badri (Acoust. and Vib. Res. Ctr., Mech. and Mechatronics Eng. Dept., The Univ. of Auckland, 20 Symonds St., Auckland 1011, New Zealand, ybad223@aucklanduni.ac.nz), George Dodd, Andrew Hall (Acoust. and Vib. Res. Ctr., Mech. and Mechatronics Eng. Dept., The Univ. of Auckland, Auckland, New Zealand), John E Cater (Mech. Eng. Dept., Univ. of Canterbury, Christchurch, New Zealand), Gian Schmid (Acoust. and Vib. Res. Ctr., Mech. and Mechatronics Eng. Dept., The Univ. of Auckland, Auckland, New Zealand), and Grant Emms (Marshall Day Acoust., Auckland, New Zealand)

The Sound Absorption Coefficient (SAC) of unconsolidated Granular Material (GM) is usually characterized by utilizing empirical models based on airflow resistivity correlations or more comprehensive models like Johnson–Champoux–Allard (JCA). These models rely on porous media geometrical parameters, which are obtained through inverse characterization methods using SAC experimental results. This presents challenges when GM's SAC estimates are required in the absence of experimental results. Therefore, this paper reports a sound absorption model for GMs based on three porous media absorption models. Each model corresponds to GMs with a mean particle diameter (De) range, related to a specific SAC frequency profile that transitions through three stages, influenced by particle size changes, up to 2 kHz. Model parameters were determined based on granular frame geometrical properties as a function of airflow resistivity and porosity of packed polydisperse Discrete-Element spherical particles. SAC estimates for various GMs were validated against impedance tube measurements. JCA-based absorption model was found suitable for larger-particle GMs (De $>$ 2 mm), exhibiting absorption peaks at frequencies above 1 kHz (Stage 1). Meanwhile, empirical-based absorption model characterized the frequency-independent SAC plateau observed in GMs with De $<$ 2 mm (Stage 2). A newly developed coupled fluid-solid powders absorption model characterized SAC for 100-micron-sized particles (Stage 3).

4:00

4pPAb9. The vibroacoustic response of snow based on Biot theory. Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@erdc.dren.mil) and Zoe R. Courville (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

The theory of poroelasticity is relevant to understanding the vibrational and acoustic response of snow to incident sound. As a porous, elastic medium snow can support two types of dilatational (compression) waves and distortion (shear) wave. A model system relevant for impedance tube measurements of snow is derived. Parameterizations for the adiabatic bulk modulus and shear modulus are based on empirical data. These parameterizations serve as a key to interpreting past snow acoustic studies that utilized an

impedance tube, since measurements of elastic constants are typically omitted. Consequently, so long as the air temperature, pressure, snow density, thickness, and static airflow resistivity are known, then the only free parameter (assuming lossless bulk and shear moduli) in the poroelastic model is tortuosity. Historical specific acoustic impedance measurements will be examined in the context of model predictions, along with a couple of recent studies. Preliminary results suggest that good agreement can be obtained between model predictions and empirical data by simply optimizing the value of tortuosity.

4:20

4pPAb10. Acoustic energy harvesting using 3-D-printed acoustic metamaterials. Justin An (Mech. Eng., Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Bldg. 42, Rm. 212-F, Washington, DC 20008, justin.an@udc.edu), Wagdy Mahmoud (Elec. Eng., Univ. of the District of Columbia, Washington, DC), and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, acoustic energy harvesting using 3-D-printed acoustic metamaterials of sonic crystals is investigated. Of particular interest is to leverage acoustic energy harvesting from urban ambient noise in the mid-range frequencies, spanning from 100 to 500 Hz. This frequency band aligns well with common environmental and urban noise frequencies, ensuring a consistent and reliable energy source, thus converting it into electrical energy through the piezoelectric effect, where certain materials generate an electric charge in response to mechanical stress or vibrations. Numerical models were developed to study the transmission loss characteristics of the acoustic metamaterials to aid in the design of the acoustic metamaterials. The numerical models are designed with the intent to aid in the positioning of a piezoelectric device on the elastic resonant energy region of the acoustic metamaterial. Impedance tube experiments are performed to investigate the output power of 3-D-printed polylactic acid (PLA), thermoplastic polyurethane (TPU), and polyethylene terephthalate glycol (PETG) acoustic metamaterials. In order to optimize the output power, an assessment was conducted by systematically altering the load resistance within the range of 0–1000 MΩ.

4:40

4pPAb11. 3-D cellular structures for thermoacoustic energy control. Elio Di Giulio (Industrial Eng., Univ. of Naples Federico II, Piazzale Tecchio, 80, Naples 80125, Italy, elio.digiulio@unina.it)

Porous stacks are essential in acoustics and thermoacoustics by enabling viscous and thermal energy exchange across a complex pore network. Optimizing these structures can enhance heat transfer and promote oscillatory gas flow with minimal dissipation. The sound power variation in an infinitesimal channel segment is influenced by acoustic pressure, volume velocity, temperature gradients, and microstructural properties, governed by key parameters $r\nu$ and $1/r\kappa$. Complex geometries, such as tetrakaidecahedral (Kelvin cell) structures, allow for independent control over thermal and viscous responses. Fabricated via additive manufacturing, these structures have been analyzed with advanced low-frequency acoustic methods, revealing that membrane thickness and opening ratio significantly impact thermoviscous behavior. This innovative geometry provides enhanced control over viscous and thermal responses, offering a new approach for optimizing thermoacoustic devices by decoupling these effects and expanding the functional versatility of porous stacks.

5:00

4pPAb12. Simulation and measurement of a Helmholtz resonance based acoustic metamaterial for military vehicle application. Michael D. Clasen (Mech. Eng., Inst. for Mechatronics, Holstenhofweg 85, Hamburg 22043, Germany, clasenm@hsu-hh.de), Clement Passemard, and Sachau Delf (Mech. Eng., Inst. for Mechatronics, Hamburg, Germany)

In the military vehicle, automotive sector, the acoustic emission of internal combustion engines has a significant impact on the environment. For example, engine fans or exhaust streams exhibit low-frequency spectra with local maxima. These correspond either to the driving speed or to the idling mode, which is related to the finite amount of pistons in the engine. In the latter case, a stationary signal of the idling mode is analyzed for its local frequency peaks. In this approach, an acoustic metamaterial based on Helmholtz-resonance is developed for the passive reduction of these narrowband frequencies. A pre-dimensioning in the geometric design of cavities and necks is carried out, while a downscaling of the size is necessary for laboratory purposes. A simulation model is developed to investigate the sound pressure distribution as well as the sound power transmission loss. The results are compared with experiments using a 3-D-printed metamaterial prototype inserted into an acoustic impedance tube. The results of this study show the potential for optimizing sound emission in noise-sensitive regions as well as in tactical, and military environments where a mission vehicle should not be detected.

Session 4pPPa

Psychological and Physiological Acoustics: Virtual Thunder: Top Presentations from the P&P Trainee Lightning Round Competition

Gregory M. Ellis, Cochair
Audiology and Speech Pathology, Walter Reed National Medical Military Center, 4494 Palmer Road N, Bethesda, MD 20814

Varsha H. Rallapalli, Cochair
Communication Sciences & Disorders, University of South Florida, 4202 Fowler Avenue, PCD 1017, Tampa, FL 33620

William M. Whitmer, Cochair
Hearing Sciences - Scottish Section, Level 3, New Lister Building, Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom

Chair's Introduction—12:55

Invited Papers

1:00

4pPPa1. On the sound pressure distribution in the inner ear during bone conduction stimulation. Simon Kersten (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstraße 5, Aachen 52074, Germany, simon.kersten@akustik.rwth-aachen.de), Henning Taschke (Formerly at: Inst. of Commun. Acoust., Ruhr Univ. Bochum, Bochum, Germany), and Michael Vorlaender (Inst. for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

During air conduction (AC), intracochlear sound pressures in the scala vestibuli and scala tympani differ significantly, enabling the determination of differential pressure as a key indicator for auditory perception. In bone conduction (BC), however, these pressures are of similar magnitude and exhibit variability due to the motion of the inner ear and interactions among different BC mechanisms. Using a finite element model, we investigated intracochlear sound pressure distributions during BC and AC, separately analyzing rigid body motion and oval window (OW) input. Our results reveal distinct pressure patterns and directionality for rigid body motion, fundamentally differing from those associated with (AC-related) OW input. Basic acoustic principles further clarify the fluid dynamics underlying these distributions. The findings underscore the need to superimpose pressure contributions from various BC mechanisms in a location-dependent manner to fully understand the inner ear's response. These results provide insight into the variability of measurement results published in the literature, and they highlight the challenges of isolating the pressure driving the basilar membrane traveling wave using intracochlear probes. Further investigations are needed to reconcile theoretical models for AC and BC stimulation and to improve the interpretation of experimental data.

1:20

4pPPa2. Two cues are better than one: Adding interaural time differences improves lateralization for bilateral cochlear-implant listeners. Paul G. Mayo (Hearing and Speech Sci., Univ. of Maryland-College Park, 0100 LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, paulmayo@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Bilateral cochlear-implant (BI-CI) listeners show limited sensitivity to interaural time differences (ITDs), the dominant localization cue for acoustic-hearing listeners, and mainly rely on interaural level differences (ILDs) for localization. Studies utilizing bilaterally synchronized direct stimulation capable of conveying ITDs have investigated sensitivity to ITDs or ILDs in isolation. It is, however, unclear how these two cues interact in controlled and bilaterally synchronized electrical stimulation. Therefore, this study performed an ITD-ILD cue-weighting lateralization experiment with BI-CI listeners using direct stimulation of single electrodes. Preliminary results show that BI-CI listeners display sensitivity only to ILDs with unsynchronized stimulation and equal sensitivity to both ITDs and ILDs with synchronized stimulation. Additionally, providing ITDs and ILDs together via bilaterally synchronized stimulation resulted in increased lateralization ranges and slopes, thus improved spatial hearing acuity compared to either cue alone. These results suggest providing both ITDs and ILDs via bilaterally synchronized sound processors has the potential to improve spatial hearing in BI-CI listeners. The data have implications for clinical sound processor design and stimulation strategies.

1:40

4pPPa3. Diotic narrowband noises can be perceived as intracranially off center in listeners with symmetrical audiometric thresholds. Obada J. AlQasem (Hearing and Speech Sci., Univ. of Maryland-College Park, 8700 Baltimore Ave., College Park, MD 20740, obada.j.oj@gmail.com) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Localizing sound sources in the horizontal plane depends on the interaural time (ITD) and level (ILD) differences. It is assumed that individuals with normal and symmetrical hearing thresholds perceive stimuli with zero ILD and ITD as fused and centered auditory images, which is not always true. This study aimed to explore ILD lateralization biases in individuals with normal hearing and explain whether they can be explained by an interaural asymmetry in monaural loudness. An ILD lateralization task was performed where the ILDs ranged between $\pm$20 dB. The stimulus was a 1-equivalent-rectangular-bandwidth narrowband noise centered at 250–8000 Hz or a wideband noise. Preliminary results showed that the participants exhibited ILD lateralization biases across all test frequencies. These biases were not caused by headphone placement, as the transducers were reversed between testing blocks. Biases were consistent across three separate testing days. A loudness balancing task showed that participants demonstrated loudness perception imbalances across testing frequencies, which were also consistent across three separate testing days. Audiometry showed $\leq$10 dB hearing thresholds asymmetry. ILD lateralization biases in individuals with normal hearing were not explained by interaural asymmetries in loudness perception or hearing thresholds.

2:00

4pPPa4. Language Environment Analysis precision in real-world and controlled environments. Rachael Pennock (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., 1-451, Evanston, IL 60208, rachaelpennock2028@u.northwestern.edu), Varsha H. Rallapalli (Commun. Sci. & Disord., Univ. of South Florida, Evanston, IL), and Pamela E. Souza (Commun. Sci. & Disord., Northwestern Univ., Evanston, IL)

Background: Adults with hearing loss experience individual auditory ecologies, encountering unique environments and listening demands. A clinician's subjective understanding of each patient's auditory ecology informs clinical decision-making. As subjective interpretation is subject to error, objective measurements would improve accurate auditory ecology characterization. The Language Environment Analysis (LENA) system is a promising tool to evaluate older listeners' environments to aid clinical decision-making. However, most LENA research has focused on adult-child conversations in quiet. This project aims to determine LENA's accuracy in capturing real-world acoustic environments. Methods: Measurements were gathered of an adult female in real-world and controlled (sound-treated booth) speech-in-noise environments. The LENA and a Brüel & Kjær sound level meter were opposite the talker. Overall levels and signal-to-noise ratios (SNR) were estimated by analyzing the intensities of segment categories over varying time periods. Results: Similar noise levels and SNRs were found for analysis windows from 5 to 15 min. For initial tests in real-world environments, noise levels varied between 61 and 86 dB(C), and SNR varied between $-$0.9 and $+$3.0 dB. Differences between the approaches may be related in part to technical parameters. Implications: Although they do not reach laboratory precision, LENA-informed measures could provide insight into individuals' auditory ecologies.

2:20

4pPPa5. Perceptual categorization of and adaptation to human voice and musical instruments: A passive-listening study. Zi Gao (Otolaryngol., The Ohio State Univ., 75 E River Rd., Minneapolis, MN 55455, gao00196@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

The human voice is a highly social and relevant auditory stimulus. Previous studies have observed a wide range of voice-sensitive effects, but less is known about the role of attention and context when categorizing sounds as either voice or non-voice. To address this gap, the current study adopted electroencephalography (EEG) passive listening tasks. In Experiment 1, vowel utterances (/a/, /o/, /u/, and /i/) and instrumental tones (bassoon, horn, saxophone, and viola) were presented with equal probability in a random sequence, and different brain responses to the two categories were observed. In Experiment 2, mismatch negativity was observed for rare instrumental tones (viola) embedded in a random sequence of four different vowels, but not vice versa, suggesting that categorization of voice and non-voice could require little to no attention but may be modulated by stimulus familiarity. In Experiment 3, ambiguous voice-instrument morphs were presented in either vocal or instrumental contexts. Logistic regression models performed above chance in predicting the type of context (voice or instrument) from the responses to ambiguous morphs. The results suggest that neural signatures of both perceptual categorization (voice/non-voice) and context effects can be observed in EEG responses under passive listening conditions. [Work supported by NIH grant R01 DC012262.]

2:40

4pPPa6. Behavioral, computational, and electrophysiological measures of temporal envelope processing. Zenzele Thomas (Medical Eng., Univ. of South Florida, 4202 E Fowler Ave., PCD 1017, Tampa, FL 33612, thomasz16@usf.edu), Nathan C. Higgins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), David A. Eddins (Commun. Sci. and Disord., Univ. of Central Florida, Orlando, FL), and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Listening to speech in noisy environments poses significant challenges for individuals with hearing loss, potentially due to poor representation of the temporal envelope. We explored the peripheral and central roles of temporal envelope processing using a gaps-in-noise task, computational modeling of the auditory periphery, and electrophysiology. Three behavioral experiments were conducted, using a 3-alternative-forced-choice procedure and a 3-down-1-up adaptive tracking method to estimate gap thresholds. Experiments 1 and 2 tracked gap detection (in ms) for various signal bandwidths, masker bandwidths, signal-to-noise ratios (SNRs), and gap modulation depths. Experiment 3 fixed the gap duration and tracked gap modulation depth (in dB) at various SNRs. Results showed that poorer SNRs were associated with poorer gap thresholds in all three experiments. In experiment 2, increased masker bandwidths also led to poorer gap thresholds. Modeling these results indicated that gap thresholds in noise can be attributed to mostly peripheral processes; however, the results of experiment 2 were more consistent with central factors. Participants also completed a passive EEG task using a standard acoustic change complex paradigm. The present study provides a comprehensive assessment of peripheral and central factors associated with temporal envelope processing which may have implications for amplification strategies. [Work supported by NIH.]

Session 4pPPb

Psychological and Physiological Acoustics and Signal Processing in Acoustics: Cadenza Machine Learning Challenge (CAD2): Improving Music for People with Hearing Loss

Trevor J. Cox, Cochair
Acoustics Research Centre, University of Salford, Newton Building, Salford M5 4WT, United Kingdom

William M. Whitmer, Cochair
Hearing Sciences - Scottish Section, Level 3, New Lister Building, Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom

Alinka Greasley, Cochair
Music, University of Leeds, UWoodhouse Lane, Leeds LS2 9JT, United Kingdom

Chair's Introduction—3:15

Invited Paper

3:20

4pPPb1. The second Cadenza machine learning challenge (CAD2): Improving music for people with hearing loss. Gerardo Roa (Comput. Sci., Univ. of Sheffield, Sheffield, United Kingdom), Scott Bannister (Music, Univ. of Leeds, Leeds, United Kingdom), Jennifer L. Firth (Univ. of Nottingham, Nottingham, United Kingdom), Simone Graetzer, Rebecca Vos (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom), Jon P. Barker (Comput. Sci., Univ. of Sheffield, Sheffield, United Kingdom), Trevor J. Cox (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., Salford, Greater Manchester M5 4WT, United Kingdom, t.j.cox@salford.ac.uk), Bruno Fazenda (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), Alinka Greasley (Music, Univ. of Leeds, Leeds, United Kingdom), and William M. Whitmer (Hearing Sci. - Scottish Section, Glasgow, United Kingdom)

The Cadenza machine learning challenges are improving the processing of music in hearing aids and consumer devices for those with hearing loss. There are two tasks in the current round (CAD2), which is organized within the IEEE SPS program. The tasks are motivated by the problem people with hearing loss can have when trying to hear out lyrics and instruments. Task 1 is to improve lyric intelligibility for pop/rock music without compromising audio quality. The objective audio quality is evaluated using HAAQI, the Hearing Aid Audio Quality Index, and intelligibility by a system based on the Whisper ASR (automatic-speech recognition). Subjective audio quality and intelligibility are also evaluated via a panel of listeners with hearing loss. Task 2 is to rebalance instruments within small classical ensembles (duets to quintets of strings and woodwinds) to enable personalized remixing. The objective metric is HAAQI. Both tasks start with stereo recordings. We will present the challenge design, a summary of the approaches taken by challenge entrants, and an overview of the objective and perceptual evaluation of the systems entered. We will also discuss how CAD2 will inform the third Cadenza challenge, launching later in 2025.

Contributed Papers

3:40

4pPPb2. Rebalancing classical music for hearing impaired listeners. Alexander Levinson (Aalborg Univ., København, Denmark) and Frej S. Lorenzen (Aalborg Univ., A. C. meyersvaenget 15, Copenhagen 2450, Denmark, fslo21@student.aau.dk)

The current quality of music listening for individuals with hearing impairments remains unsatisfactory, particularly in complex auditory environments like classical music. The Cadenza Challenge (CAD2) Task 2 aims to improve the music enhancement process for hearing-impaired listeners by rebalancing instrument levels guided by personalized audiograms. This paper addresses specific auditory challenges, such as lack of clarity and distortions, by exploring machine learning approaches and digital signal processing techniques. We implement and fine-tune the baseline ConvTasNet models on the "Real Data for Tuning" dataset, and experiment with Non-negative Matrix Factorization (NMF). Evaluations were performed using the Hearing Aid Audio Quality Index (HAAQI) on the synthesized validation set of the pre-trained model, which scores slightly lower than expected in the case of the fine-tuned model, possibly due to being tuned for real data. NMF scores are even lower as of now, but future improvements could be made in terms of using score-informed source separation techniques.

4:00

4pPPb3. Audiogram guided EQ for instrument level classical music balancing in hearing impaired listeners. Parakrant Sarkar (School of Creative Media, City Univ. of Hong Kong, Run Run Shaw Creative Media Ctr., Level 7, 18 Tat Hong Ave., Kowloon Tong, Hong Kong, Hong Kong, parakrant.sarkar@my.cityu.edu.hk) and PerMagnus Lindborg (School of Creative Media, City Univ. of Hong Kong, Hong Kong, Hong Kong)

The CAD2 Task 2 Challenge aims to create more inclusive audio experiences by rebalancing classical music for listeners with hearing loss. This rebalancing works at the instrument level within a small classical ensemble. We propose a novel enhancement pipeline that uses audiogram-based equalization (EQ), integrated with a deep-learning model for refined frequency shaping, to suit the individual hearing profiles of listeners. This will be useful for adjusting the frequency bands to suit the individual hearing profiles of listeners. This personalized frequency shaping helps maintain clarity and musicality across a range of various instruments in an ensemble. We measure the efficacy of our method with the Hearing-Aid Audio Quality Index (HAAQI), which records the enhancements in perceptual ability in a setting when hearing is impaired. We compared our method with the existing baseline model and observed that our method consistently improves both balance and intelligibility. This ensures that critical elements of the ensemble such as strings, winds, and percussion are clearly audible. Also, a personalized instrument level EQ helps in more richer listening experience for the listeners with hearing loss.

4:20

4pPPb4. Personalized music enhancement using signal adaptive dynamic equalization. Parakrant Sarkar (School of Creative Media, City Univ. of Hong Kong, Run Run Shaw Creative Media Ctr., Level 7, 18 Tat Hong Ave., Kowloon Tong, Hong Kong, Hong Kong, parakrant.sarkar@my.cityu.edu.hk) and PerMagnus Lindborg (School of Creative Media, City Univ. of Hong Kong, Hong Kong, Hong Kong)

The CAD2 Task 1 Challenge aims to enhance lyrics intelligibility for individuals with hearing loss while maintaining the audio quality. To do this, we need adaptive methods that change how sound is processed based on the characteristics of each listener. One of the popular techniques, Equalization (EQ) is widely used to adjust the balance of frequencies in audio; however, static EQ is often insufficient for tailoring music to specific auditory needs. We propose a dynamic equalization (DEQ) system that adapts in real time to listener-specific audiograms, dynamically modifying frequency bands to optimize intelligibility and tonal balance. Unlike conventional EQ, our approach uses dynamic thresholds and Signal-to-Audiogram Ratio (SAR) to adjust gains across frequency bands based on both the listener's hearing profile and the signal's energy. These adjustments are further refined with smoothing techniques to ensure a natural listening experience. Our system integrates into the baseline music enhancement pipeline, applying personalized adjustments to the demixed vocal and accompaniment components. We evaluate of proposed system using the objective metrics: CTW (correct transcribed word ratio) that is based on Whisper and audio quality (HAQQI). The results demonstrate that the proposed system delivers personalized and high-quality music experiences for listeners with hearing loss.

4:40

4pPPb5. Exploring music audio assessment with large language models and HAAQI-Net. Dyah A. M. G. Wisnu (Res. Ctr. for Information Technol. Innovation, Academia Sinica, 128 Academia Rd., Section 2, Nankang, Taipei 115, Taiwan, dyahayumgw@citi.sinica.edu.tw), Stefano Rini (Inst. of Communications Eng., National Yang Ming Chiao Tung Univ., Hsinchu, Taiwan), Ryandhimas Zezario (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan), Hsin-Min Wang (Inst. of Information Sci., Academia Sinica, Taipei, Taiwan), and Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan)

Large Language Models (LLMs) demonstrate impressive multimodal capabilities, sparking interest in applications beyond traditional text and image processing. This study examines the feasibility of using LLMs, specifically Gemini, for music audio quality assessment. Initial experiments reveal Gemini's inconsistent performance in scoring and evaluating music quality, with limited reliability across trials. To benchmark Gemini's performance, we compare it to HAAQI-Net, a Non-intrusive Neural Music Audio Quality Assessment Model initially designed for hearing aids but adaptable for normal-hearing listeners. HAAQI-Net, a proven metric for music quality evaluation, serves as the reference model. Unlike LLM-based approaches requiring extensive fine-tuning, HAAQI-Net offers greater training flexibility while maintaining consistent, reliable predictions. Using HAAQI as the benchmark, Gemini's scores show weak alignment, with a Linear Correlation Coefficient (LCC) of 0.11, Spearman's Rank Correlation Coefficient (SRCC) of 0.08, and a Mean Squared Error (MSE) of 0.15. In contrast, HAAQI-Net demonstrates strong alignment with HAAQI, achieving an LCC of 0.93, SRCC of 0.95, and an MSE of 0.01. These findings highlight HAAQI-Net's superior reliability and effectiveness, making it a more robust model for music quality evaluation than LLM-based approaches.

Session 4pSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Active and Tunable Acoustic Metamaterials

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St, Ann Arbor, MI 48109

Serife Tol, Cochair
Mechanical Engineering, University of Michigan, 2350 Hayward Street, Ann Arbor, MI 48109

Chair's Introduction—12:55

Invited Papers

1:00

4pSA1. Time-modulated phononic crystal of oscillating bimetallic rods. Arkadii Krokhin (Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, arkady@unt.edu), Matthew Li, Dmitrii Shymkiv (Phys., Univ. of North Texas, Denton, TX), and Ying Wu (Comput., Elec., and Mathematical Sci. and Eng. Div., King Abdullah Univ. of Sci. and Technol., Thuwal, Saudi Arabia)

Mathematical treatment of space- and time-modulated structures is similar. However, the practical realization of time dependence on elastic properties is a much more difficult problem than the fabrication of a phononic crystal where elastic properties are space periodic. The most common method of temporal modulation is applying AC voltage to a piezoelectric constituent. This method allows high-frequency modulation, but the depth of modulation is relatively weak. Here, we propose a mechanical method of temporal modulation which requires a complicated engineering scheme but provides deep modulation. The scatterers of a phononic crystal are bimetallic rods driven by an external force to oscillate along their axes in a solid (or fluid) matrix. Due to mechanical oscillations, a propagating sound wave suffers time-dependent scattering. High elastic contrast between the components of the bimetallic rods provides deep time modulation and high contrast between the metals and the background matrix provides deep space modulation. The band structure of a mechanically time-modulated phononic crystal is calculated for aluminum-copper rods in an epoxy matrix. Mix band gaps with complex values of ω and k are predicted and analytical properties of the dispersion relation in complex ω-k plane are studied. [This work is supported by the NSF Grant No. 1741677 and by the AFOSR grant FA9550-23-1-0630.]

1:20

4pSA2. Broadband sound absorption at the sub-wavelength scale with noncausal active absorbers. Chen Shen (Dept. of Mech. Eng., Rowan Univ., 201 Mullica Hill Rd., Glassboro, NJ 08028, shenc@rowan.edu), Kangkang Wang (Inst. of Acoust., Nanjing Univ., Nanjing, China), Sipei Zhao (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Botany, New South Wales, Australia), Haishan Zou, Jing Lu (Inst. of Acoust., Nanjing Univ., Nanjing, China), and Andrea Alu (Photonics Initiative, Adv. Sci. Res. Ctr., City Univ. of New York, New York, NY)

Sound absorption is of vital importance in many applications. In recent years, metamaterial-based absorbers have shown great promise in noise reduction for various setups. However, it has been shown that there is a fundamental tradeoff between the thickness and bandwidth for passive absorbers based on causality. To relax this constraint, here we demonstrate an active absorber that specifically leverages the *a priori* information for broadband sound absorption. A general relation between the minimum footprint of the absorber and the absorption bandwidth is established under the noncausal limit. Theoretical and experimental results confirm the validity of the proposed approach, in which an active absorber with an adjustable absorption spectrum is demonstrated. The absorber uses a feedforward mechanism to induce a noncausal response and further reduces its thickness compared to passive absorbers. The results showcased a scenario where noncausal information can be important in designing active acoustic elements.

1:40

4pSA3. Realizing a tunneling-like phenomenon in nonreciprocal active acoustic metamaterials. Joe Tan (Univ. of Southampton, Southampton, United Kingdom), Sayan Jana (Mech. Eng., Tel Aviv Univ., Tel Aviv, Israel), Felix Langfeldt (Univ. of Southampton, Southampton, United Kingdom), and Lea Beilkin (Mech. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, leabeilkin@tauex.tau.ac.il)

We demonstrate a tunneling-like phenomenon recently reported in lattices with nonreciprocal couplings, realized here in an active acoustic metamaterial. The system consists of a waveguide with acoustic actuators and sensors embedded in its walls. Nonreciprocity, leading to asymmetric sound transmission in opposite directions, is created using actuators and sensors operating in a real-time feedback loop process. This nonreciprocal sound transmission is realized in an inner portion of the waveguide, thereby forming an artificial non-Hermitian interface that separates two Hermitian sections. This configuration supports the non-Hermitian skin effect, where modes accumulate at one boundary. In our acoustic implementation, we investigate the propagation of a sound wave along the waveguide as it encounters the artificial interface. While the skin mode accumulation acts as a barrier, inhibiting wave penetration into the interface, under specific conditions, the wave tunnels to the other side, creating a quiet zone within the In our realization the cross-section remains unobstructed, akin to a plain waveguide. This experimental demonstration underscores the applicability of the tunneling phenomenon across different physical systems and its potential applications in acoustic wave control.

2:00

4pSA4. Some non-Hermitian and modulated acoustic topological crystals. Baile Zhang (Nanyang Technolog. Univ., SPMS-PAP-0506, 21 Nanyang Link, Singapore 637371, Singapore, blzhang@ntu.edu.sg)

Topological acoustics was introduced over a decade ago, with recent research increasingly focusing on non-Hermitian and modulated acoustic crystals. In this talk, we introduce some of our recent advancements in this direction. First, we explore a one-dimensional (1-D) acoustic crystal with disorder applied to the nearest-neighbor couplings. Such a disorder can induce non-Hermitian point-gap topology and localize all waves at a boundary, which challenges the conventional picture of Anderson localization. Second, we discuss the gap closure in such a non-Hermitian crystal and demonstrate the non-Hermitian edge burst. Finally, we illustrate the concept of delicate topology and demonstrate it with returning Thouless pumping in a 1-D acoustic crystal.

Contributed Papers

2:20

4pSA5. Reciprocity and propagation in space–time acoustic materials. John D. Smith (DSTL, DSTL Porton Down, Salisbury SP4 0JQ, United Kingdom, jdsmith@dstl.gov.uk)

Generalized reciprocity relations for space–time-dependent acoustic materials are considered. The form of the resulting propagation equations depends on the original constitutive equations and thermodynamic relationships. A specific example of a space–time acoustic fluid is developed and it is found that in this case, the propagation is reciprocal, in the sense that the Green's function is self-adjoint. The physical behavior is different from conventional pressure-acoustics however and, for the spatially varying case, propagating waves must have associated vorticity even in the absence of viscosity.

2:40–3:00 Break

3:00

4pSA6. Modeling phi-bits in acoustic metamaterial for quantum-inspired computation. Abrar Nur E Faiaz (Mech. Eng., Wayne State Univ., 4206 Saint Antoine, Detroit, MI 48201, hr1642@wayne.edu), Akinsanmi S Ige (Dept. of Mater. Sci. and Eng., The Univ. of Arizona, Tucson, AZ), Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., Detroit, MI), Jake Balla (Dept. of Comput. Sci., The Univ. of Arizona, Tucson, AZ), M Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI), Pierre A. Deymier (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ), Keith Runge (New Frontiers of Sound Sci. and Technol. Ctr., The Univ. of Arizona, Tucson, AZ), and Joshua A. Levine (Dept. of Comput. Sci., The Univ. of Arizona, Tucson, AZ)

Logical phi-bits, classical analogs of qubits, are nonlinear modes arising in externally driven coupled finite-length acoustic waveguides. Logical phi-bits exhibit geometric phases that, by tuning the driving conditions, can be exploited to realize quantum-like operations in classical acoustic systems. In this work, we model the dynamics of phi-bits using a nonlinear three-chain mass-spring system to investigate the possible origins of the behavior of the phi-bit geometric phase observed in physical experiments. The model accounts for nonlinearity, dissipation, and boundary constraints. We examine how driving frequencies and nonlinear spring interactions shape phi-bit behaviors. Nonlinearities up to the fifth order significantly influence the geometric phase, with higher-order terms revealing interdependencies that expand the range of observable phenomena. Dissipation results in phi-bits states with complex amplitudes and may play a critical role in determining the behavior of the geometric phase. Finally, the effect of boundary conditions at the ends of the finite waveguides is also reported. This work contributes to the broader understanding of the complex behavior of the phi-bit geometric phase and more generally of acoustic analogues of quantum systems. [Funding: NSF grant # 2204382, 2204400, and 2242925.]

3:20

4pSA7. Flexural wave reflections from time modulated boundaries. Kayla Cecil (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, kncecil97@utexas.edu), Blaine T. Gilbert (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Benjamin M. Goldsberry (Appl. Res. Labs. at The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Traditional analysis of flexural wave scattering assumes that the scatter properties are time invariant and therefore the scattered wave field is at the same frequency as the incident wave and only depends on the local impedance contrast. Recent interest in time-modulated systems enables unprecedented control over wave behavior, offering opportunities for frequency and wavenumber conversions that are not possible in passive media. The control of scattered flexural waves in beams and plates with time-varying boundary conditions is of particular interest for vibration control. In this work, we present a semi-analytical model to study flexural wave reflections from time-varying boundary conditions. This model is based on Euler–Bernoulli beam theory in which expressions for reflected flexural waves are derived for various beam boundary conditions. The time dependence of the boundary conditions is incorporated by representing the solution as a Fourier series expansion to consider scattering into harmonics related to the modulation frequency and wavenumber. The semi-analytical model is compared to a realistic implementation that utilizes resonator termination with PZT patches. Numerical and experimental work is carried out to validate the model.

3:40

4pSA8. Coupled-mode theory for guided acoustic waves with spatiotemporally modulated boundaries. Blaine T. Gilbert (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Eng. Teaching Ctr. II - ETC, 204 E Dean Keeton St., Austin, TX 78712, blaine.gilbert@utexas.edu), Kayla Cecil (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Benjamin M. Goldsberry (Appl. Res. Labs. at The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Recent research in optical, electromagnetic, and acoustical metamaterials has shown that spatiotemporal modulation (STM) of bulk material properties and interfaces between materials can increase the control of propagating waves and wave scattering at boundaries. Specifically, STM of bulk properties has been used to enable nonreciprocal wave propagation in unbounded systems and to couple modes at different frequencies in bounded systems through frequency and wavenumber conversion. This work considers the case of mode coupling in an acoustic waveguide using STM boundaries. As a first step in the design of STM boundaries, we investigate the effect of STM on one boundary of a two-dimensional acoustic waveguide that is coupled to a fluid half-space on its other boundary. The analysis first considers guided wave modes when the waveguide has spatially varying boundaries using coupled-mode theory. Temporal modulation is then introduced by expanding the solution for propagating and leaky modes using Fourier series expansion to consider scattering into harmonics of the modulation frequency and wavenumber. An analysis of mode conversion efficiency based on the modulation amplitude and frequency is presented and discussed.

4:00

4pSA9. Modeling active response in a leady wave antenna. Abigail D. Willson (Acoust., Penn State Univ., P.O. Box 30, M.S. 3220B, State College, PA 16804, adw5@psu.edu), Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., State College, PA), and Amanda Hanford (Penn State Univ., State College, PA)

A leaky-wave antenna (LWA) is a passive acoustic metamaterial comprised of an array of unit cells. The array acts as a beam-steering device in which the output radiation direction is dependent on the input frequency. The passive LWA only functions in a frequency range set by the dimensions and material properties of the array. In order to control the frequency range for a constant or slow changing input, tunable material properties can be introduced. If the input signal changes often or rapidly over time, then active properties with quick responses to inputs are used instead. Presented is a technique for modeling LWA in the time-domain with active elements. The model is used to explore the transient and steady-state of both the unit-cell and full system outputs of the system.

4:20

4pSA10. Synchronization driven acoustic nonreciprocity. Alexander K. Stoychev (Dept. of Mech. and Process Eng., ETH Zurich, Sonneggstrasse 3, ML J 41, Zurich 8092, Switzerland, astoychev@student.ethz.ch), Xinxin Guo (Dept. of Mech. and Process Eng., ETH Zurich, Zurich, Switzerland), Ulrich Kuhl (Institut de Physique de Nice, Université Côte d'Azur et CNRS, Nice, France), and Nicolas Noiray (Dept. of Mech. and Process Eng., ETH Zurich, Zürich, Switzerland)

This contribution investigates the scattering properties of aeroacoustic limit cycles and their reciprocity-breaking interactions. Specifically, a mean tangential flow grazing an aperture is employed to generate a self-sustained cavity mode (a whistle), providing a framework for analyzing the acoustic scattering behavior of a self-excited, sub-wavelength structure, referred to as a meta-atom. The nonlinear interactions within a system of coupled meta-atoms are then harnessed to control the wave propagation in an acoustic waveguide, demonstrating phenomena such as synchronization-driven superradiance, unidirectional transmission and unidirectional transparency. Furthermore, using the mean flow as a control signal enables the management of the wave propagation path across multiple waveguides connected to the meta-atom assembly. Additionally, the experimentally observed phenomena have been successfully described by a minimal model of a nonlinear saturable gain oscillator. The model's autonomous and non-autonomous dynamics, as well as its synchronization properties, are shown to align with experimental measurements.

Session 4pSC

Speech Communication: Speech Production Poster Session II

Marie Bissell, Chair
Univ. of Texas at Arlington, Arlington, TX 76019

All posters will be on display from 1:20 p.m. to 4:20 p.m. Authors of odd numbered papers will be at their posters from 1:20 p.m. to 2:50 p.m. and authors of even numbered

Contributed Papers

4pSC1. Acoustic cues at the intersection of prosodic position and phonological contrast. Chiara Repetti-Ludlow (Carnegie Mellon Univ., 350 Lamont Pl, Pittsburgh, PA 15232, chiara.repetti.ludlow@gmail.com)

In speech, phonemes are distinguished by acoustic cues that exist in a dynamic context, varying based on factors such as phonological contrast and prosodic position. However, few studies have considered this interaction in depth. To bridge this gap, the present research considers Italian stop voicing and stop gemination in sentence-medial and sentence-final positions. Prior research on Italian has shown that durational and voice quality cues are relevant to these contrasts and that these cues also vary at prosodic boundaries. Italian therefore provides an ideal test case for exploring how acoustic cues vary based on contrast, prosodic position, and their interaction. 25 speakers of Italian participated in a production study, reading target words in frame sentences. Acoustic information (e.g., stop duration, H1-H2) was extracted using Praat, and regression models and decoding methods were used to assess acoustic cue variation. Results reveal that some cues are informative both for voicing and gemination, while simultaneously being impacted at prosodic boundaries. Despite this, both contrasts remain robust across phrasal positions. This demonstrates that the same cue can vary based on several factors, but this does not necessarily result in neutralization as other cues take on a compensatory role.

4pSC2. The effects of lexical stress levels and information structure on the spectral characteristics of American English full vowels. Mariko Sugahara (Dept. of English, Doshisha Univ., Kamigyo, Kyoto 602-8580, Japan, msugahar@mail.doshisha.ac.jp)

This study investigates how lexical stress levels ([StrLev]) and information structure, i.e., focus versus post-focus given, affect the spectral characteristics of four full-vowel categories (/æ, ɛ, ɔ, oʊ/), and whether the vowel categories ([VCat]) interact with the two factors. The stress levels considered are primary stress (P) as *local*, secondary stress (S) as *localization*, and unstressed unreduced (U) as *locality*. Speech data were collected from 25 American English native speakers, divided into two based on the presence/absence of post-focus pitch accent (the deaccenting versus the hyper-accenting group). Vowel dispersion, measured as Euclidean distances from the F1–F2 center, was analyzed by ANOVA separately for the two groups. Regardless of the groups, significant main effects of [StrLev] and [InfoStruc] were obtained: vowel dispersion was greatest for P and smallest for U, and post-focus vowels were less dispersed (more reduced) than focused ones. This indicates that the post-focus reduction is partly independent of deaccenting. However, a significant interaction of [StrLev] × [InfoStruc] was only present for the deaccenting group. A significant interaction of [VCat] × [StrLev] was obtained for both groups: spectral reduction induced by U was more drastic for /æ/ than for the other vowels, which indicates that stress-related reduction is not uniform across vowels.

4pSC3. An articulatory and spectral study of "er-hua" rhymes in Beijing Mandarin. Shuying WANG (Linguist and Translation, City Univ. of Hong Kong, Hong Kong 518057, Hong Kong, swang747-c@my.cityu.edu.hk) and Wai Sum Lee (Linguist and Translation, City Univ. of Hong Kong, Hong Kong, Hong Kong)

This study presents the ultrasound and formant data of the so-called "er-hua" rhymes in Beijing Mandarin that contain the vowels [i, y, u, a, ɿ, ʅ, ɤ] with the rhotacized suffix "-er." The ultrasound imaging reveals two distinct tongue gestures, "retroflex" and "bunched," during the "er"-suffixation, which is regardless of the vowel type of the rhymes. Both types of tongue gestures result in a large fall in F3, although the extent of change is less pronounced for the "bunched" than the "retroflex" rhymes and for the rhymes that contain the back vowels than the front vowels. Besides, a schwa-like sound may appear at the end of the rhymes after the "er"-suffixation. The schwa is either a retroflex [əʵ] or bunched [ɚ], with tip curling gesture for the former but not the latter. To explore the articulation-acoustics relationship during the "er"-suffixation, a "mixed effects linear regression model" (Hussain and Mielke, 2021) will be performed to examine the parallel between the variations in the articulatory parameters (blade anteriority, dorsum height and dorsum concavity) and the formant frequencies of the different types of vowels in the "er"-suffixed rhymes with the "retroflex" and "bunched" tongue gestures.

4pSC4. Mapping the Turkish vowel space: A comprehensive vowel production study. Zuheyra Tokac (Linguist, Northwestern Univ., Evanston, IL 60208, zuheyratokac@u.northwestern.edu)

Turkish vowel production studies to date have either small numbers of speakers or involve uncontrolled production stimuli that do not account for allophonic, contextual, or coarticulatory variation. This study aims to provide a comprehensive description of the Turkish vowel space, offering a thorough examination of proposed Turkish vowel allophones in the literature and an exhaustive analysis of vowel-to-vowel coarticulation in Turkish. Twenty-six monolingually raised adult native Turkish speakers (17 females) produced 1600 Turkish vowel tokens in controlled phonetic contexts, forming minimal pairs across syllable-initial, medial, and final positions in monosyllabic and polysyllabic words and nonwords, within carrier sentences and read passages. I analyze formant values, duration, and spectral change to describe the canonical vowel space of Turkish and discuss allophonic and coarticulatory variation due to vocalic and consonantal context. I outline how the Turkish productions collected in this work will be published as a publicly available Turkish corpus, inviting further phonetic analysis, and serving as a resource for stimuli in future speech perception experiments.

4pSC5. Production of the Canadian English /ɹ/ by Mandarin-speaking second-language learners: Lip articulation and acoustic measures. Youran Lin (Commun. Sci. and Disord., Univ. of Alberta, 810-8210 111 St NW, Edmonton, AB T6G 2C7, Canada, youran1@ualberta.ca) and Daniel Aalto (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada)

Producing a speech sound in a second language (L2) can be challenging when it has a counterpart in the learner's first language (L1) but involves nuanced differences. The Canadian English /ɹ/ involves a slight lip rounding, absent in its Mandarin counterparts. This study asks whether and how Mandarin-L1 speakers' English /ɹ/ productions differ from English-L1 speakers. Forty Mandarin-L1 international students and 40 English-L1 speakers will be recruited to produce controlled non-words, words, and sentences containing /ɹ/. Their third formants (a characterizing acoustic measure of /ɹ/) and lip opening, spreading, and protrusion measures will be extracted from audio and facial kinematic recordings. Pilot analyses focused on 2 speakers' short non-word productions, paired with 3 vowels (/a i u/) in 3 positions (pre-, post-, inter-vocalic), 5 repetitions each (45 productions of each speaker). Linear regression indicated that the Mandarin-L1 speaker was not different from the English-L1 speaker in their third formants ($p > 0.05$) but produced less lip spreading and more protrusion ($p < 0.001$). Although the L2 learner achieved similar acoustic outcomes, their /ɹ/ articulation featured excessive lip rounding, a non-mandatory feature in English, indicating phonetic-level differences due to L1–L2 interaction. Full analyses will identify patterns related to L1, task complexity, and individual differences.

4pSC6. Acoustics of tongue-root feature in the Enugu Ezike variety of Igbo. Samuel K. Akinbo (Dept. of Linguist, Univ. of Toronto, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, samuel.akinbo@utoronto.ca) and Nkechi M. Ukaegbu (Linguist and Nigerian Lang., Univ. of Nigeria, Nsukka, Nsukka, Enugu, Nigeria)

We investigate the acoustic correlates of the tongue-root feature in the Enugu Ezike variety of Igbo, which has not been studied until now. This variety has 11 vowels, like many Northern Igbo varieties, unlike the Standard variety which has 8 vowels. The three additional vowels in Enugu Ezike are [ɛ] and two schwas, [ə] and [ɘ]. Based on their phonotactic distribution and alternation, the vowels, including the schwas, are classified according to tongue-root advancement and retraction (ATR and RTR). Our investigation shows that various acoustic parameters distinguish ATR vowels from their RTR counterparts. The results of the acoustic analyses indicate that the acoustic correlates in Enugu Ezike are comparable to those in the Standard variety and other varieties of Igbo. Our findings also support the hypothesis that the articulatory gestures involved in tongue-root advancement and retraction are in synergistic and antisynergistic relationships with the articulatory gestures of laryngeal constriction and movements associated with voice quality and pharyngealization. Enugu Ezike also presents novel evidence supporting the hypothesis that the features ATR and tense are distinct. Keywords: acoustics, vowels, tongue root, Igbo, African language, schwa

4pSC7. Phonetic documentation in Central African Republic: The acoustics and articulation of Ubangi labial-velars and implosives. Lorenzo Maselli (African Studies, Universiteit Gent, Blandijnberg 2, Gent, Oost-Vlaanderen 9000, Belgium, lorenzo.maselli@ugent.be)

This contribution presents the first comprehensive acoustic and articulatory analysis of labial-velar and implosive articulations in the Ubangi languages of the Central African Republic. Integrating electro-glottographic evidence, oral airflow measurements, subglottal pressure analysis, and acoustics, we investigate micro-variation in these rare sound classes, which, while relatively prominent in the Macro-Sudan Belt, remain severely underdescribed in the literature. Labial-velar and implosive consonants are hypothesized to share historical and typological ties, but a general lack of phonetic data has hindered research into these issues. Ongoing work at our university addresses key questions regarding their synchronic and diachronic behavior, including the classification of implosives as stops or sonorants, the headedness of labial-velar left edges, and the adequacy of articulatory phonology in describing these segments. Additionally, our research examines the socio-historical factors underlying the spread of these sounds across Ubangi and their integration into neighboring Bantu speech communities. These findings are presented following the most extensive phonetic documentation mission ever conducted in the Ubangi region, during which numerous under-documented sound systems were surveyed. This research provides critical insights into the linguistic diversity of one of Africa's least-studied areas and addresses broader phonological and typological questions about the behavior of some of its rarest sounds.

4pSC8. Acoustic variability underlying pathological voice quality. Jody Kreiman (Head and Neck Surgery and Linguist, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu) and Yoonjeong Lee (Viterbi School of Eng., Univ. of Southern California, Los Angeles, CA)

Pathological voice quality poses unresolved challenges for voice science. The presence of physical pathology does not necessarily imply poor voice quality, nor does a large deviation from modal phonation necessarily imply pathology. Creak, breathiness, and chaotic vibrations occur in both normal and pathological voice, and most speakers can produce a wide range of qualities, including those often labeled "pathological." Why do some voices signal a disorder, rather than being perceived as part of a quality continuum that also includes normal voices? To address this question, we derived a psychoacoustic voice space for speakers with diagnosed vocal pathologies and compared it to the corresponding space for speakers without a diagnosis. Acoustic variables were measured from recordings of sentences read by speakers with and without diagnosed pathology (e.g., mass lesions, paralyses, functional disorders, neurogenic disorders) and analyzed using principal component analysis. We hypothesized that vocal pathology disrupts the transmission of biologically important information related to a speaker's state of arousal and reproductive fitness. This information is available from the voices of speakers without pathology, and its loss in cases of pathology hypothetically undermines communicative functions that may represent a primary evolutionary basis for phonation.

4pSC9. Analyzing formant trajectories of Akuzipik vowels. Giulia M. Soldati (English, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, gmasella@gmu.edu), Matthew C. Kelley, and Sylvia Schreiner (English, George Mason Univ., Fairfax, VA)

Akuzipik, or St. Lawrence Island Yupik, is an endangered Alaska Native language spoken fluently by approximately 540 people, primarily on St. Lawrence Island, Alaska. Previous work on Akuzipik used point measurements of formants to investigate the vowel inventory. In the present study, we are examining formant trajectories and vowel inherent spectral change to identify the temporal patterns of the acoustics. Six Akuzipik speakers were previously recorded for vowel analysis as part of a larger documentation and revitalization project. Speakers read the carrier sentence *aghnat* X *atiimaat* [aʁnat X atiːmaːt] "the women said X," where X is a word starting with one of the seven target vowels. The stimuli had been selected so that there were eight unique words for each vowel. Each word began as VC, with V being the target vowel and C being equally balanced between labial, coronal, velar, and uvular obstruents. Vowels were manually annotated in Praat. The formant trajectories will be analyzed using generalized additive mixed modeling, with particular attention paid to verifying the putative vowel qualities and lengths. The results of this study will scaffold further work on the phonetics of Akuzipik and related revitalization work.

4pSC10. Accentedness in English is connected to inter-word (co)articulation and speech rate, a pilot study. Eija Aalto (Département de génie électrique, École de technologie supérieure, 1100 Notre-Dame St W, MOntreal, QC H3C 1K3, Canada, eija.aalto@etsmtl.ca), Walcir Cardoso (Education, Concordia, Montreal, QC, Canada), Lucie MENARD (Linguist, UQAM, Montréal, QC, Canada), and Catherine Laporte (Département de génie électrique, École de technologie supérieure, Montreal, QC, Canada)

Previous studies have shown that consonantal errors and difficulties in coarticulation may decrease English language learners' (ELL) speech intelligibility. The current pilot study investigates connections between inter-word (co)articulation, perceived accentedness, and speech rate. *Participants*

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were six adults (2 men) and one native speaker, with L1 of Mandarin ($n = 4$), Thai ($n = 1$), and Finnish ($n = 1$). Methods: The dataset contained 21 repeating (4x) short sentences and two reading passages with all English phonemes. Inter-word consonantal errors, speech rate (compared to the model), and accentedness (2 judges) were compared statistically (Spearman) and qualitatively. Results revealed that accentedness correlated strongly with the number of errors ($r = 0.95$, $p > 0.001$) and speech fluency ($r = 0.92$, $p > 0.01$). Qualitatively, the error types that increased with accentedness, were word-final consonant omission, voicing, consonant substitutions, and assimilations. In addition, all the ELL speakers showed variability in consonant coarticulation in repeating sentences, unlike the native speaker. The variability increased with accentedness up to 1/3rd of the sentences pronounced with varying inter-word errors. Discussion: The results of this pilot study align with earlier findings on the importance of consonants and coarticulation in ELL speech production. In addition, speech rate and variability in pronunciation may be connected to perceived accentedness.

4pSC11. Suprasegmental emphasis in Neo-Aramaic. Noah Khaloo (Linguist, Univ. of California, San Diego, 3869 Miramar St., Cr-156, San Diego, CA 92037, nkhaloo@ucsd.edu)

This study provides an acoustic analysis of so-called "emphasis" and its spreading in Urmi, a dialect of Neo-Aramaic. In Semitic and neighboring languages, emphasis takes the form of contrastive plain versus emphatic segments. In Urmi, however, emphasis is suprasegmental, resulting in minimal pair distinctions between emphatic and non-emphatic words without a segmental trigger (Garbel, 1964, 1965a/b; Khan, 2008). Previous impressionistic work has been inconclusive as to the phonetic nature of emphasis in the language, with reports of both velarization and pharyngealization, among other articulations. We performed an acoustic analysis of 284 words from a single native speaker (data from another native speaker has also been collected and will be reported). The results show that vowels in emphatic words have a lower F2, while fricatives (particularly /x/ and /s/) have a lower Center of Gravity. The frequency of F1 is not a robust correlate of emphasis. This suggests that the primary phonetic parameter of emphasis in Urmi is velarization. Other analyses will include voice quality measurements over vowels to determine whether the emphatic vowels differ from non-emphatic ones in terms of increased glottal constriction. In sum, Urmi provides evidence for a rare phonological feature: suprasegmental velarization, without a segmental trigger.

4pSC12. A comparative study of the acoustic features of the post-alveolar fricative [ʃ] in Mandarin, Cantonese, and English speakers. Shuqi Huang (The Univ. of Hong Kong, Mintian Sisha Village, Shatian Town, Dongguan, Guangdong 523000, China, huangshuqi2123@163.com)

Chinese learners of English often experience interference from their native languages when pronouncing certain fricatives in English, particularly the post-alveolar fricative [ʃ]. This study investigates the acoustic features of the voiceless post-alveolar fricative [ʃ] produced by Mandarin Chinese learners, Cantonese learners, and native English speakers. The corpus comprises recordings of English words containing [ʃ] from 20 Mandarin Chinese speakers, 20 Cantonese speakers, and 20 native English speakers. Spectral moments of [ʃ]—centroid frequency, variance, skewness, and kurtosis—were extracted using Praat to capture key acoustic properties associated with fricative production. Linear Discriminant Analysis was employed to evaluate the distinctiveness of accented productions of [ʃ] and to classify differences between Mandarin- and Cantonese-accented [ʃ] relative to native English [ʃ]. Results reveal that transfer from native languages significantly influenced [ʃ] production: Mandarin speakers' production of [ʃ] in English resembled Mandarin retroflex [ʂ], while Cantonese speakers' [ʃ] in English showed features of Cantonese [s]. Spectral parameter analysis indicated that Mandarin-accented [ʃ] was acoustically closer to native English [ʃ] than Cantonese-accented [ʃ]. These findings provide empirical evidence on the influence of L1 phonetic transfer on English fricative production and underscore the need for targeted pronunciation instruction tailored to specific L1 backgrounds.

4pSC13. An acoustic study of L2 lexical tone production by tonal L1 learners. Shuqi Huang (The Univ. of Hong Kong, Mintian Sisha Village, Shatian Town, Dongguan, Guangdong 523000, China, huangshuqi2123@163.com)

While most studies on second language (L2) tone acquisition have focused on learners with non-tonal first language (L1) background, little attention has been given to tonal L1 learners. Significant differences in tonal contrasts between Cantonese and Mandarin create challenges in L2 tone production. This study compares the tonal production of L2 Mandarin by Cantonese L1 speakers and native Mandarin speakers, as well as the tonal production of L2 Cantonese with Mandarin L1 speakers and native Cantonese speakers. The aim of this study is to investigate the differences in F0 parameters—mean F0, F0 range, duration, and the F0 turning points—and the influence of tonal L1 on L2 tone acquisition. Acoustic parameters extracted using Praat indicated that Cantonese L1 speakers had a larger tonal space than L2 speakers. The F0 values of L2 Mandarin resembled those of Cantonese L1 speakers more than native Mandarin speakers. These findings suggest assimilation processes, where L1 tonal characteristics influence L2 tonal production, consistent with the category assimilation hypothesis from the Speech Learning Model. This study highlights the impact of tonal contrasts in L2 acquisition and offers insights into cross-linguistic tone learning for phonetic and pedagogical research.

4pSC14. Exploring the speech chain experimentally: English sound identification accuracy is modulated by phonological factors across adult speakers and listeners. Daniel Aalto (Commun. Sci. and Disord., Univ. of Alberta, 8205 114 St NW, Edmonton, AB T6G2G4, Canada, aalto@ualberta.ca) and Naomi Gurevich (Commun. Sci. and Disord., Purdue Univ. Fort Wayne, Fort Wayne, IN)

Speech perception is influenced by communicator characteristics and linguistic factors. Using a multi-speaker sound identification task, this study explored the relative contributions of speaker and listener variability. 384 triphones were extracted from 21 native English speakers reading a list of phonetically rich words. The words and the resulting triphones were balanced in terms of sound frequency (five groups), context prominence (three tiers), and word stress (levels: monosyllabic, unstressed/stressed polysyllabic). The mean length of the words was 6.0 sounds (range: 3–14). A total of 22 listeners were instructed to type the consonants they identified in audio clips drawn randomly from all the speakers using a Latin square design yielding a total of 16 408 responses. The mixed effects logistic regression model showed identification accuracy was lowered by lower frequency sounds ($z = 4.0$), less prominent context ($z = 5.7$), lack of stress ($z = 3.7$), and number of sounds in the word from which the triphone was extracted ($z = 3.3$). Speaker intercept standard deviation was 87% greater than that of listeners. The results corroborate a model of sound identification that considers frequency and context prominence. The results suggest that our experimental design is sufficiently powered to further explore the components of the speech chain.

4pSC15. Validating novel measures of sibilant fricatives using phoneme goodness ratings of children's speech. Eugene Wong (Univ. of Minnesota, 164 Pillsbury Dr. SE, UMinneapolis, MN 55455, wong0703@umn.edu) and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

Spectral moments (SMs, Forrest *et al.*, 1988) are widely used to characterize fricatives. Shadle (2023) described the problems of using SMs and proposed a set of alternative parameters (APs) based on articulatory and aerodynamic models. The current study examines how well the APs from Shadle (2023) and SMs predict listener ratings of the goodness of children's word-initial /s/ and /ʃ/ productions. Seventy-six adult listeners rated a total of 1014 /s/- and /ʃ/-initial words produced by 64 children of 2.5–3.5 years of age along a visual analog scale. For each fricative, the first four SMs were calculated for the middle portion of the fricative, as were the filter parameters in Shadle (2023): frequency and amplitude of the main peak in the 3–8 kHz range, and the frequency and amplitude of the minimum in the 0.55–3 kHz range. While correlation and mixed-effect regression analyses show that both sets of measures robustly predict adult ratings, the model with APs yielded a slightly lower AIC, indicating a better model fit. Moreover, the APs better predict the narrow transcriptions of children's /s/ and /ʃ/ productions. Together, these findings provide external validation for Shadle's APs.

4pSC16. The acoustic properties of laterals in Southwestern Saudi Arabic. Saeed Alqarni (English, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, salqarn@gmu.edu) and Matthew C. Kelley (English, George Mason Univ., Fairfax, VA)

Lateral consonants exhibit varied acoustic characteristics across languages with distinct realizations ranging from clear to dark. In Arabic dialects, these variations are salient due to their interaction with emphatic consonants, which creates distinct acoustic patterns through their influence on surrounding segments. The present study examines the acoustics of the lateral consonants in Southwestern Saudi Arabic, focusing on how phonetic environment, age, and sex influence their realization. Using generalized additive mixed models, the trajectory of F2−F1 was analyzed across initial, medial and final positions adjacent to emphatic and nonemphatic sounds. By modeling dynamic patterns over time, the study captures non-linear formant dynamics while accounting for the phonetic environment as well as age and sex. The findings demonstrate positional effects, with medial laterals adjacent to emphatic sounds exhibiting lower F2−F1 differences compared to those in non-emphatic contexts. Male speakers produced lower F2−F1 values, and older speakers showed greater variability.

4pSC17. Perception of voice qualities in implicit and explicit phonetic convergence. Kyler B. Laycock (Linguist, Ohio State Univ., 1308 Carbone Dr., Columbus, OH 43224, laycock.21@osu.edu)

Talkers' speech becomes more similar to the speech of others through phonetic convergence. While the effects of phonetic convergence are well-documented for a number of acoustic and perceptual dimensions, the present study examined convergence to voice quality in a word shadowing task. This study additionally sought to understand differences in production when participants are asked to repeat the words they hear (implicit shadowing) as compared to when participants are explicitly asked to imitate speech. Convergence to voice quality was assessed acoustically through measurement of f0, the difference in amplitude between first and second harmonics (H1–H2), and cepstral peak prominence (CPP), and perceptually through an AXB task in which participants assessed the perceptual similarity of shadowing productions to the productions of model talkers. In general, both these analyses of participants' speech showed phonetic convergence toward model talkers' voice qualities. Preliminary results suggest the degree of convergence was characterized by the instructions in the shadowing task. Explicit instructions generally resulted in greater convergence but also resulted in convergence to different acoustic features as compared to implicit instructions. Complementary to studies describing quantitative differences in the magnitude of convergence across these conditions, these data also evidence qualitative differences in which acoustic dimensions are preferenced in convergence.

4pSC18. Data-canvas application for speech perception and production data. Hugh M. Birky (Otolaryngol., Vanderbilt Univ. Medical Ctr., 3000 Hillsboro Pike, Apt. 124, Nashville, TN 37215, hugh.m.birky@vumc.org), Victoria A. Sevich (Otolaryngol., Ohio State Univ., Columbus, OH), and Terrin N. Tamati (Otolaryngol., Vanderbilt Univ. Medical Ctr., Nashville, TN)

Data visualization in speech communication research often requires specialized coding skills, which can limit accessibility for students, clinicians, and researchers unfamiliar with programming. This project aimed to develop an intuitive and flexible data visualization application, DataCanvas, which allows users to explore, analyze, and visualize behavioral speech perception and production datasets without requiring coding expertise. The application was built using the Shiny framework in R and offers an interactive interface that allows users to upload datasets, choose from multiple visualization options (e.g., scatterplots, boxplots), and customize plots. This presentation will demonstrate the use of DataCanvas for a study involving speech perception and production data collected from adult cochlear implant recipients. Boxplots and scatterplots were created in DataCanvas to visualize trends in accuracy, reaction times, and sensitivity, and to explore relationships among factors. DataCanvas simplifies the data visualization process for speech communication researchers by removing coding-related obstacles and providing a highly adaptable tool for data exploration and presentation. Future development will focus on expanding the app's functionality and customization options based on feedback from researchers, students, and clinicians. Areas of suggested improvements include better data integration, statistical analyses, acoustic analyses, and more data visualization functionality.

4pSC19. Monophthongization of diphthongs in Southern American English: A perception study. Harys Dalvi (Comput. Sci., Brown Univ., Providence, RI), Rachel Meyer (Linguist, Univ. of Florida, Gainesville, FL), and Ratree Wayland (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, ratree@ufl.edu)

The monophthongization of /ai/ is more prevalent than /au/ in Southern American English due to historical, phonetic, and sociolinguistic factors. Phonetically, the tongue movement for /ai/ (low to high front) may be more prone to reduction than for /au/ (low to high back). Sociolinguistically, these changes often signal regional identity. This study investigates the monophthongization of /ai/ to /a/ and /au/ to /a/ in Southern English through two 11-step continua ([a-ai] and [a-au]) presented to listeners for identification. Preliminary data from 4 Southerners and 12 non-Southerners partially support the hypothesis that Southerners perceive /a/ more in the /a-ai/ continuum. Boundary positions for both continua were similar across groups, but Southern speakers showed narrower boundary widths for /a-au/ ($M = 0.854$) compared to /a-ai/ ($M = 1.188$), indicating clearer perceptual distinctions for /a-au/. Non-Southerners exhibited consistent boundary widths for /a-ai/ ($M = 1.318$) and /a-au/ ($M = 1.335$), suggesting stable perceptual systems unaffected by dialectal variation. These findings reveal that the monophthongization of /ai/ in Southern English reduces perceptual precision, reflecting its greater susceptibility to dialectal variation. In contrast, non-Southerners maintain uniform perceptual precision across continua, highlighting regional dialects' impact on phonetic perception.

4pSC20. Acoustic evidence of complex stop reduction in Gã children's speech. Felix Kpogo (Linguist Program, Brown Univ., 91 Waterman St., Rm. 113, Providence, RI 02912, felix_kpogo@brown.edu)

The present study examines whether 5-year-old Gã-speaking children in Ghana produce VOT and closure duration differences between reduced complex stops (i.e., [k͡p] and [g͡b]) and their simplex labials (i.e., [p], [b]) in word-initial and phrase-medial contexts. Using a picture-naming task, the study provides evidence of covert contrast for voiceless stops [k͡p] versus [p], whereas voiced stops [g͡b] versus [b] undergo complete neutralization. In the word-initial position, [p] and [k͡p] exhibit positive VOT, but [p] has significantly higher VOT than reduced [k͡p], which shows negative VOT when unreduced, suggesting that listeners fail to perceive the contrast children produce. However, no significant VOT difference is observed between [b] and [g͡b] in word-initial position. In phrase-medial contexts, closure duration for [b] and [g͡b] shows no significant difference, indicating a convergence of voiced stops across contexts. This acoustic evidence supports crosslinguistic auditory data that doubly articulated stops are susceptible to segmental reduction in children's speech (Isaiah, 2015; Nwokah, 1986; Kpogo *et al.*, 2021). Particularly, Kpogo *et al.* (2021) show that 5- to 8-year-old Gã-speaking children simplify [k͡p] and [g͡b] to [p] and [b], respectively. The present study attributes the reduction patterns to the complex articulatory demands of producing doubly articulated stops and potential perceptual challenges associated with the contrast.

4pSC21. PIN-PEN merger in perception. Irene B. Smith (Linguist, McGill Univ., 1085 ave. du Docteur-Penfield, Montreal, QC H3A1A7, Canada, irene.smith@mail.mcgill.ca) and Meghan Clayards (Linguist, McGill Univ., Montreal, QC, Canada)

Merged production of /ɪ/ and /ɛ/ before nasal consonants is well documented in Southern US English. Perception studies of this merger are more limited (cf. Austen 2020). One possible source of pre-nasal merger is anticipatory vowel nasalization. A 2AFC perception task asked US listeners from inside or outside the South, to categorize stimuli on continua from *bid* to *bed* and *bin* to *Ben*. To test the effects of consonant and vowel nasality separately, we cross-spliced stimuli in a 2 by 2 design. We ask (1) whether Southern speakers are merged in perception, as is generally assumed, and (2) whether it is vowel nasality, consonant nasality, or both that gives rise to the merger in perception. We fit a logistic regression model on the probability of /ɛ/ responses as a function of *continuum step*, *vowel nasality*, *consonant nasality*, and *subject region*, with all interactions. We found that Southern listeners had a flatter categorization function with lower accuracy at the continuum ends than non-Southern listeners whenever a nasal coda was present, regardless of vowel nasality, confirming that (1) Southern speakers are, to some degree, merged in perception, and (2) that the presence of the nasal coda, and not vowel nasality, conditions merger in perception.

4pSC22. Acoustic variation in monolingual and bilingual production: comparing human- versus machine-directed speech. Olga Dmitrieva (Linguist, Purdue Univ., West Lafayette, IN) and Daria Novoselova (Linguist, Purdue Univ., 337 S Chauncey Ave., West Lafayette, IN 47906, dnovosel@purdue.edu)

As voice technology becomes more widespread, it is increasingly important to understand how speakers modify their speech for human versus computer interactions. Previous research indicates that individuals treat computers as social actors with limited processing capabilities and adapt their speech accordingly. This study investigates how bilingual and multilingual speakers adjust their speech in human–computer interactions, focusing on acoustic differences between L1 and L2 speakers. Forty subjects (20 L1 speakers of American English and 20 L2 English speakers with Farsi as their L1) perform a gap-filling task in two settings: one with a human English teacher and the other with a virtual assistant. Both conditions are controlled using the Wizard of Oz technique, with pre-recorded videos for the teacher and an animated avatar for the assistant. We analyze acoustic-phonetic features such as vowel space, vowel duration, tense-lax vowel contrasts, and voice onset time for initial stop consonants. Based on prior research, we expect computer-directed speech to differ from human-directed speech and exhibit clear speech characteristics; bilinguals may show less pronounced clear-speech features compared to monolinguals. The findings could improve speech technologies, particularly for L2 speakers, as current systems favor native speakers.

4pSC23. An acoustic analysis of rhoticity in Nigerian English. PraiseGod Aminu (Linguist, Univ. of Pittsburgh, 616 North Highland Ave., Pittsburgh, PA 15206, praisegod.aminu@pitt.edu)

Although Nigerian English (NE) has been described as a non-rhotic English variety (Jowitt-2018), I present the first systematic acoustic analysis of rhoticity (non-prevocalic-/r/s in ca***r***, fa***r***me***r***) in NE. The dataset comprises spontaneous and elicited speech of 30 Yoruba native speakers born and raised in Lagos, Nigeria. Interviews were transcribed and force-aligned in hand-checked FAVE. A modified Praat script measured the first three formants in each V + /r/ sequence at nine equally spaced time points. For a feature intrinsic normalizing measure of rhoticity, this study focuses on the static measurement of the F3–F2 difference. The formant measurements are subjected to mixed-effects linear regression analyses in R, using lme4 package to model F2–F3 at the F3 minimum. Based on 11 823 tokens of non-prevocalic-/r/ and with higher values of F3–F2 (Hertz) indicating weaker rhoticity, findings show the following results: gender (female = 941; male = 1972), age (18–29 = 901; 30–49 = 1634; 50–63 = 2082), social class (upper-class = 703; middle-class = 1598; lower-class = 2097), and positional contexts (word-final = 964; word-initial = 1673). As there is overt prestige associated with American English (AE), I hypothesize that NE is becoming rhotic with influence from AE. Younger upper-class females lead the change, especially in elicited speech. Low F3 close to F2 serves as an accurate correlate of rhoticity in NE. Importantly, there was hyper-rhoticity (i.e.-“Africaɹ”).

4pSC24. Speech planning influences the acoustic realization of phonological alternation: Evidence from morpho-syntactically complex Mandarin Tone 3 sandhi. Chang Wang (Linguist, Univ. of Kansas, Blake Hall, Rm. 427, 1541 Lilac Ln., Lawrence, KS 66045, changwang@ku.edu) and Jie Zhang (Linguist, Univ. of Kansas, Lawrence, KS)

Production planning has been shown to affect the application rate of variable phonological alternation. This study investigates how planning could influence the acoustic realization of alternation. We ask whether the acoustic gradience in morpho-syntactically complex Mandarin T3 sandhi is a consequence of speech planning regulating how much information of the T3 allomorph can cascade to downstream acoustics. Building on the production planning hypothesis and cascadedness of speech production, we propose that the availability of T3-sandhi context constrains the cascading process of the T3 allomorph, yielding differences in the amount of T3 residual in the acoustic output. Specifically, a stronger syntactic boundary reduces the likelihood that phonological information embedded in the upcoming word is available during target word production. Consequently, more T3 allomorphic information cascades from the lexical–phonological level to the phonetic level, resulting in more T3-like acoustic output characterized by a shallower rise in f0. Results from 15 speakers’ data showed that, under stronger boundary conditions, T3 sandhi items exhibited a significantly shallower rising slope compared to weaker boundary conditions, while T2 non-sandhi items did not show this effect. This Tone-by-Boundary interaction supported our hypothesis that the acoustic difference stemmed from production planning and cascadedness, not phonetic effects of duration.

4pSC25. Morphological influence on Korean /l/ pronunciation: A comparative study of heritage and native speakers. Yun J. Kim (Linguist, Emory Univ., 532 Kilgo Circle 202C, Atlanta, GA 30322-0001, yun.kim@emory.edu)

This study examines how morphological information influences Korean /l/ pronunciation and how speakers’ dominant language affects this process. The research compared heritage Korean speakers in the US with native Korean speakers. The experiment used 45 words with (C)V/l/ita structure, including morphologically simple words, complex phrases, and words with geminate /l/. Twenty college-aged participants (10 native speakers from Junnam Korea and 10 heritage speakers from the US Southeast) recorded these words in carrier sentences. Researchers analyzed the recordings using Praat and VoiceSauce, focusing on vowel duration, liquid duration, and burst release duration. The findings revealed that word-final /l/ before vowel-initial suffixes undergo incomplete neutralization in Korean. Additionally, Korean and English speakers differ in balancing morphological information preservation with epenthesis. The study showed that bilingual speakers’ dominant language influences their /l/ realization in both languages. These insights contribute to understanding morphology’s role in language production and have applications in language education and AI systems.

4pSC26. Acoustic cues in morphological processing: The role of context in speech perception. Yun J. Kim (Linguist, Emory Univ., 532 Kilgo Circle 202C, Atlanta, GA 30322-0001, yun.kim@emory.edu)

Previous studies have shown that morphological information influences phonetic details in speech production. Seyfarth *et al.* (2018) demonstrated that English speakers use vowel and segment duration to distinguish between morphologically simple words (e.g., “booze") and complex words (e.g., “boos"). However, it remains unclear whether speakers can utilize acoustic cues signaling morphological information during speech perception. This study investigated whether native English speakers use these acoustic cues in perception. Forty sentences with embedded target words were recorded by native English speakers. Two types of stimuli were used: isolated target words (Experiment 1) and low-pass filtered sentences with intact target words (Experiment 2). Participants chose between morphologically simple and complex options for each stimulus. Results showed that participants performed at chance level with isolated words but could significantly distinguish between morphologically simple and complex words when given filtered sentence context. This suggests that speakers can utilize acoustic cues for morphological information, but only with sufficient acoustic context. These findings demonstrate the active use of acoustic signals in speech perception and morphological processing, offering insights into how children might process morphologically varying words.

4pSC27. Closure voicing in stop consonants: Crosslinguistic insights from language contact. Le Xuan Chan (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, Philadelphia, PA 19104, lxchan@sas.upenn.edu)

This paper examines closure voicing (i.e., vocal fold vibration during a consonant closure) crosslinguistically among multilingual speakers in a language contact setting – Malaysia. While previous studies have brought attention to the length and the shape of closure voicing in stop consonants crosslinguistically, less is known about how these effects interact in multilingual, language contact environments. In this paper, I examine both the shape and length of closure voicing in three typologically different languages that exist within the same speech community – Malay (true voicing), English (aspiration; variable voicing), and Mandarin (aspiration). Analyzing a corpus of 7634 wordlist tokens (Malay: 3000, English: 3500, Mandarin: 1134) from 50 early multilingual speakers, I found that Malay voiced stops display full voicing lasting throughout the consonant closure, while voicing in Malaysian English is partial and ceases before the burst release. These patterns, however, interact with different multilingual speakers – bilingual speakers of Malay and English distinguish voicing patterns between Malay /b,d,g/ and English /b,d,g/, while trilingual speakers of Malay, English, and Mandarin do not. These findings have important ramifications for crosslinguistic phonemic representations in multilingual speech, leading to sound change trajectories in language contact.

4pSC28. Effects of word position on high vowel devoicing rates and related duration patterns in Tokyo Japanese. Rion Iwasaki (Program in Speech-Language-Hearing Sci., Graduate Ctr., CUNY, 365 Fifth Ave., Rm. 7304, New York, NY 10016, riwasaki@gradcenter.cuny.edu), Kevin D. Roon (Program in Speech-Language-Hearing Sci., Graduate Ctr., CUNY, New York, NY), Jason A. Shaw (Dept. of Linguist, Yale Univ., New Haven, CT), and D. H. Whalen (Program in Speech-Language-Hearing Sci., Graduate Ctr., CUNY, New York, NY)

High vowels /i/ and /u/ are often devoiced between two voiceless obstruents in Tokyo Japanese. We examined the rate of devoicing across word positions, in either the first or second mora of the word. We also examined the durations of intervals crossing segments over a /C_1VC_2/ sequence, where both C_1 and C_2 were stops, such as a /C_1V/ mora and the interval from C_1 burst to the onset of C_2 closure. Eight speakers of Tokyo Japanese produced 24 four-mora nonce words 15 times each. Items had the structure of /C_1VC_2ateko/ or /taC_1VC_2ateko/. Eight items contained an environment conducive to high vowel devoicing. The rate of devoicing was dependent on word position. It was nearly 100% in the first mora, but was much lower in the second mora, just above 50% with large cross-speaker variability. Regardless of the word position, devoicing had a consistent shortening effect on the cross-segmental intervals. Moreover, intervals with voiced high vowels in the devoicing environment were shorter than voiced vowels in other environments, indicating that the devoicing environment has an effect on timing even when devoicing fails.

4pSC29. An exploration to the acoustics of focus in the Spanish spoken in Pasto, Colombia. Yeimy J. Roberto (Modern Lang. and Linguist, Florida State Univ., 625 University Way, Tallahassee, FL, Office 356, Tallahassee, FL 32306, yjr19@my.fsu.edu)

This study explores the acoustics of focus in the Spanish spoken in Pasto, Colombia, specifically emphasizing a narrow focus on subject and object positions. Prior research on subject and object focus in Spanish is scarce (Face, 2001, 2002; Prieto *et al*., 2018; González *et al*., 2022). An experimental study was conducted using a contextualized sentence-reading task, and recordings were analyzed for pitch accents, boundary tones, duration, and pitch range following the SpToBI (Aguilar *et al*., 2024) annotation system. Preliminary results from eight speakers indicate distinct acoustic patterns for marking focus. Focused subjects were characterized by a delayed rising pitch accent (L+>H*), accompanied by increased duration and pitch range. In contrast, focused objects often featured a rising pitch accent (L + H*) aligned with the stressed syllable or were deaccented (L*). Additionally, the verb's pitch accent played a significant role in focus marking, with falling tones signaling subject focus and rising tones signaling object focus. These findings shed light on the role of intonation in encoding sentence focus and highlight potential dialectal differences in Spanish. Future research could investigate the perceptual and processing implications of these acoustic patterns to deepen our understanding of focus marking mechanisms in speech.

4pSC30. Exploring the perception–production link in bilingual voices. Sylvia Cho (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, sylvia_cho@sfu.ca)

In language acquisition research, some studies have found a strong relationship between perception and production, while others have not found a perception–production link. While the relationship between perception and production has been explored for segmental and non-segmental contrasts, less attention has been given to exploring this relationship between voice perception and production. The goal of the present study is, therefore, to evaluate the link between the acoustic variability found in voice production and the processes of speaker identification in voice perception. To explore this, 24 filter and source-based acoustic measurements are estimated for Korean-English bilingual voices, and these features will be compared to perceptual discrimination ratings of these voices [J. Acoust. Soc. Am. 155, A274 (2024)]. The results will be reported in relation to the relationship between acoustic variability found within and across speakers and its effects on voice quality and speaker identity.

4pSC31. Quantifying the effects of the Indian English phonological system on accent perception using deep neural networks. Nitin Venkateswaran (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32605, venkateswaran.n@ufl.edu), Rachel Meyer, and Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL)

The perception of accents in speech is influenced by the speaker's native phonology, but quantifying this influence remains a challenge. This study aims to quantify how Indian English phonology affects the perception of accents in the English spoken by Hindi speakers. A Gated Recurrent Unit (GRU) based neural network model, Phonet [Vásquez-Correa *et al*., Proc. Interspeech 2019, 549–553 (2019)], is trained on corpora of spoken Indian English (IE) and General American English (AE) to learn the phonological class probabilities of speech segments of both Englishes in a joint vector space. Class probability vector representations are then generated for IE speech from a test corpus annotated with accent ratings by native speakers of AE. We analyze two key contrasts: the labiodental approximant [ʋ], an allophone of the labiovelar approximant [w] in IE, and the retroflex stop [ʈ], compared with its AE counterpart [t]. Euclidean distances between test segments and mean AE/IE baselines are calculated in the joint vector space. A multinomial logistic regression of the distances on the accent ratings shows that segments more distant from the AE baseline correlate with higher odds of strong accents, with segments more distant from the IE baseline showing lower odds. The methods used in this study have potential applications in sociophonetic research and speech acquisition/learning, providing new tools for understanding accented speech.

4pSC32. A systematic review and meta-analysis of the development of coarticulation in child speech. Atlas Boulom, Coralie Cram, Nicholas Guymon, John McGahay, Zachary Metzler, Emma Montilla, Vishwas Shetty, Jian-Leat Siah (Linguist, UCLA, Los Angeles, CA), and Margaret Cychosz (Linguist, UCLA, 335 Portola Plaza, Los Angeles, CA 90095, mcychosz4@ucla.edu)

Coarticulation, the temporal and spatial overlap of speech gestures, is known to develop throughout childhood. However, methodological differences across studies have made it challenging to characterize its developmental trajectory. This meta-analysis synthesizes findings from 21 studies

examining coarticulation in children aged 3–10 years (initial search yielded = 325 results; 199 papers underwent title-abstract screening). We normalized articulatory and acoustic measures of coarticulation from the selected works and grouped participants into 2-year age bins. Effect sizes (Hedges' *g*) will be calculated for each age bin and analyzed using mixed-effects meta-regression to assess relationships between the degree of coarticulation and age. Our findings will indicate whether coarticulatory patterns systematically change throughout childhood, and at what developmental time point this change occurs. This analysis will provide the first quantitative synthesis of developmental coarticulation research to determine how mature coarticulatory patterns emerge as children develop and refine their speech motor control. Results will have implications for models of speech motor development and clinical assessment of speech production in children.

4pSC33. Comparing the rhythmic p(roduction)-center in Mandarin and Cantonese. Bihua Chen (Indiana Univ., Ballantine Hall 704, 1020 E Kirkwood Ave., Bloomington, IN 47401, bc1@iu.edu), Yu-Jung Lin (College of the Holy Cross, Worcester, MA), and Kenneth de Jong (Indiana Univ., Bloomington, IN)

Rhythmic productions of speakers of numerous languages show that syllables with different segmental content do not exhibit isochrony of any single acoustic event. Rather, studies across languages show consistent effects of onset and coda consonant and vowel, which together suggest an alignment point with the metronome near the onset of the vowel. This alignment is affected by the repetition rate, with faster rates inducing later alignment with the metronome. Chow *et al*. [JPhon 49, 55–66] reported productions by Cantonese speakers with an apparent alignment point at the beginning of the onset consonant, suggesting that the structure of Sinitic languages may determine a different alignment point. However, Lin and de Jong [JPhon 99, 101245] found Mandarin productions to be similar to those for other languages, but differing from this Cantonese data. The current paper presents productions by Cantonese speakers gathered with the same protocol as used in Lin and de Jong, finding effects of consonant type, repetition rate, presence of onglide, and consonant duration, with an alignment pattern similar to the Mandarin study, but different from the findings in Chow *et al*. These results, together with those in Lin and de Jong, do not support language structure as a factor in rhythmic alignment.

4pSC34. Articulatory and acoustic correlates of tongue root advancement in Khalkha Mongolian. Joshua Sims (Linguist, Indiana Univ., 2461 S Woolery Mill Dr., Bloomington, IN 47403, jodasims@iu.edu)

This paper presents articulatory and acoustic data of tongue root contrasts in Khalkha Mongolian, comparing acoustic cues in F1 and F2 to measurements of tongue root position from 3-D/4-D ultrasound imaging. The primary acoustic cue of [+ ATR] across languages is decreased F1 (Stalwart, 2008), with various effects on F2 and F1 bandwidth found in the literature. Washington (2016) found that Kyrgyz and Kazakh covary [±ATR] and [±back], with [+ ATR,-back] marked by increased F1 and decreased F2, based on acoustic and ultrasound data. Khalkha Mongolian contrasts [−ATR, + back] vowels both with [+ ATR, + back], conditioned by vowel harmony; and [+ ATR, −back] vowels, conditioned by palatalized consonants. This paper compares formants of these three classes of vowels with measures of tongue root position, taken from 3-D/4-D Ultrasound images of 6 Khalkha speakers. Comparisons of formants of these three classes of vowels, correlated with tongue root measurements, make clear the acoustic correlates of tongue root advancement in vowel formants. The [+ ATR, + back] vowels are marked by decreased F1, while the [+ ATR, −back] vowels show decreased F1 and increased F2.

4pSC35. An initial acoustic investigation of the Mindat (K'Cho) phonetic inventory. Amanda Bohnert, Grayson Ziegler (Linguist, Indiana Univ., Bloomington, IN), and Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu)

Mindat (also known as K'chò or Mün) is a South Central Tibeto-Burman (Chin) language spoken in southeastern Chin State in western Burma and in diaspora communities worldwide. Little work on—and, to our knowledge, no acoustic documentation of—the Mindat sound system exists. Nolan (2002) notes a series of plain and aspirated voiceless stops that is relatively standard both cross-linguistically and Chin-internally (/p p^{h} t t^{h} c k k^{h}/), as well as voiceless sonorants (and voiced counterparts) /m̥ n̥ ŋ̊ l̥/, which are typologically rare but common in Chin. Additional sounds that are uncommon even in Chin languages include dental and velar affricates /t̪͡θ k͡x/, implosives /ɓ ɗ/, a dense fricative space /f v s∼ʃ z∼ʒ x ɣ h/, and complex onsets /pl p^{h}l/. Nolan (2002) also reports a seven-vowel system /i ɛ a ə ɨ o u/ with all monophthongs contrasting for length, six diphthongs /ɛj aj əj ɨ oj uj/, and four tones (high, low, rising, falling). Using data from one speaker, we provide an acoustic description of the Mindat phonetic inventory, including (but not limited to) measures of VOT for stops, center of gravity for fricatives/affricates, formant measures/trajectories for vowels, and pitch tracks for tones.

4pSC36. F0 and F1 interactions in a tonal language with a dense high vowel space. Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 540, Bloomington, IN 47405-2201, kberkson@indiana.edu), Amanda Bohnert, and Grayson Ziegler (Linguist, Indiana Univ., Bloomington, IN)

Several aspects of the relationship between fundamental frequency (F0) and F1 in vowels are of interest to speech scientists, including that F0 may exceed the standard F1 range, such as in sung speech of sopranos (Garnier *et al*., 2010; Schutte and Miller, 1986; Sundberg, 2009), and that intrinsic F0 tends to be higher for high vowels than for low vowels (e.g., Whalen and Levitt, 1995). This intrinsic difference holds true in Thai, a tonal language, albeit with some vowel- and tone-specific differences (Punyayodhin *et al*., 2010). Thai has only two high vowels (/i u/); however, in Lutuv (aka Lautu), a South Central Tibeto-Burman language, 10 of 14 phonemic vowels occupy the higher part of the vowel space (including /i y ɨ ʉ ɯ u/). As Lutuv is also tonal, two things are true: (1) F0 associated with high tones can be high enough to approach the expected high-vowel F1 range; and (2) small differences low in the F1 range are highly consequential due to the density of the high vowel space. Herein we investigate Lutuv high vowels produced in three different tonal ranges to investigate the interaction of F0 and F1 in a language with a dense high vowel space.

4pSC37. Articulatory and acoustic variability of glottalized sonorant codas in Hakha Lai. Grayson Ziegler (Linguist, Indiana Univ., 504 Ballantine Hall, 1020 E. Kirkwood Ave., Bloomington, IN 47405, grzieg@iu.edu)

Non-modal sonorants cross-linguistically exhibit gestural asynchrony, resulting in modal and non-modal portions. This asynchrony can exhibit substantial variation; for example, Maddieson (2002) and Bird *et al*. (2008) find variability in glottalized sonorants, observing glottalization (i.e., creak) pre-, post-, and mid-segment (i.e., [^{n}n̰ n̰^{n} ^{n}n̰^{n}]). Hakha Lai, a South-Central Tibeto-Burman language, has both onset voiceless sonorants [m̥ n̥ ŋ̊ l̥ r̥] and coda glottalized sonorants [mˀ nˀ ŋˀ lˀ rˀ]. Previous work (Ziegler *et al*., 2023; Ziegler, 2024) found that HL voiceless sonorants exhibit strong articulatory variability in the coordination of voicelessness and oral constriction despite persistent and strong acoustic cues. The articulatory and acoustic characteristics of the glottalized sonorant codas remain unexplored. Using data from 5 HL speakers (4F, 1M), this work provides an acoustic analysis of Hakha Lai glottalized sonorants to determine (a) the temporal coordination (and variability) of glottalization, (b) the acoustic cues used to contrast coda modal and glottalized sonorants (including F0, spectral tilt, harmonics-to-noise ratio, strength of glottal excitation, and vowel duration/quality), and (c) whether variability across manner of articulation (nasal versus liquid) is observed in parallel to onset voiceless sonorants.

4pSC38. Does clinical experience impact listeners' perception of disordered speech? Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., 1579 S Renwick Blvd, Bloomington, IN 47401, mjhenry5519@gmail.com), Tessa Bent, Carina Obrecht, and Palak Hemrajani (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Speaker-related variability associated with speech sound disorders (SSDs) can pose a challenge to accurate speech recognition. Listeners who have more experience with disordered speech, such as speech-language pathologists (SLPs), may have advantages in speech recognition compared to naive listeners, similar to perceptual adaptation findings with other unfamiliar variants (e.g., second language accents). However, there are conflicting findings regarding the experience benefit with SSD and few studies have examined how experience impacts intelligibility. Here, we test how clinical experience (i.e., clinical practicum hours) impacts listeners' intelligibility ratings using visual analog scaling for typically developing (TD) and SSD talkers. Listeners included speech, language and hearing sciences undergraduate and graduate students ($n = 43$). Stimuli were simple sentences produced by six children (3–6 years of age) from the Speech Exemplar and Evaluation Database (SEED, Speights *et al.*, 2020). TD talkers received similar intelligibility ratings, regardless of experience. For SSD speech, intelligibility ratings decreased with increased clinical experience. Though listeners had relatively little experience, results suggest clinical experience leads to increasingly stringent response criteria for evaluating disordered speech as typical or atypical. Future studies should include practicing SLPs with broader experience and specific clinical speciality areas (e.g., pediatric SSD versus adult cognition).

THURSDAY AFTERNOON, 22 MAY 2025 GALERIE 1, 1:00 P.M. TO 5:20 P.M.

Session 4pSPa

Signal Processing in Acoustics, Acoustical Oceanography and Computational Acoustics: Machine Learning in Underwater Acoustics III

Kendal Leftwich, Cochair
Physics, University of New Orleans, 1021 Science Building, New Orleans, LA 70148

Youngmin Choo, Cochair
Sejong University, 209 Neungdong-ro Gwangin-gu, Seoul 05006, Korea

Shaun Pies, Cochair
Physics, University of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148

Contributed Papers

1:00

4pSPa1. Phase-preserving gray box model for robust sonar signal separation. TaeHoon Her (DGIST, 333, Techno jungang-daero, Hyeonpung-eup, Dalseong-gun, Daegu, Republic of Korea, Rm. 813, Bldg. 205, Biseul Village, Daegu 42988, Korea, taehoon.her@dgist.ac.kr), Jaesok Yu (DGIST, Daegu, Korea), Jongkwon Choi, and Keunhwa Lee (Sejong Univ., Seoul, Korea)

Passive sonar systems monitor underwater acoustic signals from diverse sources, including vessels, submarines. However, separating these signals from noise and overlapping sources remains a significant challenge. Traditional spectrogram-based methods often lose critical phase information, reducing separation performance, while existing deep learning models adapted from speech signal separation lack interpretability, limiting practical application in sonar systems. This study introduces Cycle Conv-Transformer TasNet, a novel gray box model for underwater signal separation that addresses these limitations. By leveraging time series representations of signals and preserving phase information, the model achieves more robust separation. Built upon ConvTasNet, it incorporates Transformer-based self-attention to effectively capture temporal correlations. Furthermore, the model is designed as an Explanation-Producing System, providing interpretable visualizations of time-frequency regions influencing its decisions and enhancing explainability alignment with XAI principles. Additionally, cycle consistency enhances signal reconstruction reliability even with limited training data. The proposed model processes signals using a convolutional encoder, Transformer-based source separator, and decoder, enabling efficient and accurate signal separation. Experimental results demonstrate a $1.2\times$ improvement in processing speed compared to DPRNN and a 3.9 dB gain in SI-SNRi, highlighting its superior performance in both speed and separation quality.

1:20

4pSPa2. Advanced modeling of propeller tip-vortex cavitation noise using deep learning. Youngjoo Kim (Sejong Univ., 209, Neungdong-ro, Seoul, Gwangjin-gu 05006, Korea, submakim@gmail.com), Jongkwon Choi, wooyoung hong, and Keunhwa Lee (Sejong Univ., Seoul, Korea)

Propeller tip-vortex cavitation (TVC) noise significantly influences naval vessel acoustic signatures. However, existing cavitation noise models in underwater acoustic simulators fail to account for characteristics of propeller TVC noise. This study proposes an advanced method that integrates a probabilistic representation of noise occurrence with generative adversarial networks (GANs) to generate realistic propeller TVC noise. Experimental data collected under controlled cavitation tunnel conditions were used to model the stochastic occurrence of TVC events based on propeller revolutions per second (rps), while GANs generated high-frequency, impulsive waveforms. The proposed method was validated through comprehensive evaluations, including statistical analyses of waveform similarity (cross-correlation and low-dimensional embedding), time-frequency analyses using spectrograms, and auditory feature analyses based on cepstral coefficients to assess perceptual fidelity. The results demonstrated that the proposed approach effectively captures both the stochastic and acoustic properties of TVC noise, thereby improving the realism of underwater acoustic simulations and outperforming existing methods. [This work was supported by the Korea Research Institute for Defense Technology Planning and Advancement (KRIT) grant funded by the Korea government [Defense Acquisition Program Administration (DAPA)] (No. KRIT-CT-22-052, Physics-guided Intelligent Sonar Signal Detection Research Laboratory, 2025)]

1:40

4pSPa3. Machine learning methods applied to the Thames Water dataset. Ruth Willet (Penn State Univ., 201 Old Main, University Park, PA 16802, raw5930@psu.edu) and Karl Reichard (Penn State Univ., State College, PA)

The Thames Water dataset is a collection of over 38 000 microphone data files from over 18 000 unique sites across London and the UK published by the Thames Water utility company. The files are published with metadata including date and time, site id, noise and spread calculated by an internal acoustic logger, and a binary indicator of if water pipe leaks were found during the measurement period. This work explores applications of machine learning methods to this acoustic data to aid in applying acoustic prognostic health management (PHM) analysis through accurate condition indicators and fault detection. The challenge in adapting machine learning algorithms for PHM is the abundance of unbalanced data. It is statistically more likely to measure data from a properly working pipe than one that is leaking, so it can be challenging to train models that will be able to efficiently detect faults. This work plans to quantitatively compare results from several common machine learning models to adequately determine which ones can provide the most insights compared to standard acoustic signal processing techniques.

2:00

4pSPa4. Similarity metric based feature fusion for active sonar target classification. Mingu Kang (Underwater Acoust. and signal processing, Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Korea, rkd9412@sju.ac.kr) and Youngmin Choo (Sejong Univ., Seoul, Korea)

We propose a novel multi-feature fusion strategy that selectively merges hand-crafted acoustic features with low inter-feature similarity to enhance active sonar classification in data-scarce scenarios. While deep learning models typically learn latent representations automatically, the limited availability of active sonar data necessitates reliance on hand-crafted features, which inevitably cause information loss. Our method addresses this limitation by systematically measuring feature similarity through an analysis of common neighbors between two feature sets and fusing only those sets determined to be sufficiently dissimilar. Experimental results demonstrate that the selective fusion of low-similarity feature sets yields better classification performance compared to fusing highly similar features, thereby underscoring the effectiveness of our approach in maximizing complementary information from limited sonar datasets. [This work was supported by Korea Research Institute for defense Technology planning and advancement (KRIT)—Grant funded by the Korea government (DAPA (Defense Acquisition Program Administration)) (No. KRIT-CT-23-026, Integrated Underwater Surveillance Research Center for Adapting Future Technologies, 2025)]

2:20–2:40 Break

2:40

4pSPa5. A study on the weakly supervised deep learning for active sonar target classification algorithm using Meta Pseudo Labels. Yena You (Electron. and Elec. Eng., Kyungpook National Univ., 80 Daehak-ro, Buk-gu, Daegu 41566, Korea, nayena@knu.ac.kr) and Seokjin Lee (School of Electron. and Elec. Eng., Kyungpook National Univ., Daegu, Korea)

Current target classification using active sonar relies heavily on the expertise and sustained concentration of sonar operators over extended periods. Recently, neural network-based target classification methods for the active sonar system have been researched, but they suffer from difficulty in obtaining a sufficient amount of labeled data. To address this problem, we developed a weakly supervised deep learning algorithm for active sonar target classification in this paper. The proposed system consists of a Convolutional Recurrent Neural Network (CRNN) with Meta Pseudo Labels to handle both labeled and unlabeled active sonar data. To evaluate our method, a simulation for target classification was performed, and the results show that the proposed method achieves significantly improved accuracy compared to the baseline model.

3:00

4pSPa6. A study on practical adaptation of active sonar classifiers under dataset shifts. Geunhwan Kim (Dept. of Elec., Electron., and Control Eng., Changwon National University (CWNU), Changwon National Univ., 20, Changwondaehak-ro, Uichang-gu, Changwon-si, Gyeongsangnam-do, Changwon 51140, Korea, kimgw200@changwon.ac.kr) and Youngmin Choo (Sejong Univ., Seoul, Korea)

Active sonar classifiers in practical environments face significant challenges due to dataset shifts. While fine-tuning is a widely used technique to address such problems, its application to active sonar classification is hindered by the small size and limited diversity of available datasets, resulting in catastrophic forgetting and negative transfer. In this study, we introduce a mode-connectivity-based fine-tuning to overcome these challenges. This approach constructs a mode-connectivity curve between the weights of two independently pre-trained networks and optimizes the curve parameters using *in-situ* test data instead of offline training data. With the guidance of the mode-connectivity curve, this approach ensures that the fine-tuned weights remain robust to both training and test datasets, enabling effective adaptation to shifted test datasets while maintaining performance on the training dataset. Additionally, we extend this method to various architectures and datasets. Experimental results demonstrate its robustness across diverse conditions, highlighting its potential applicability to a wide range of applications.

3:20

4pSPa7. Open-source synthetic aperture sonar simulation datasets. Jason Philtron (Penn State Univ., ARL Bldg., University Park, PA 16802, jhb186@psu.edu), Daniel C. Brown, Shawn Johnson, and David Williams (Penn State Univ., University Park, PA)

There are many interesting machine learning (ML) applications in underwater acoustics. However, some ML algorithms require large amounts of training and testing data and there is a lack of open-source data for these purposes. The Point-based Sonar Signal Model (PoSSM) is a useful tool that generates synthetic time-series data appropriate for coherent signal processing applications. One underwater acoustics application is the identification of objects in synthetic aperture sonar (SAS) imagery. This paper introduces multiple datasets that provide a collection of SAS imagery from a generic SAS sonar above multiple seafloor textures and bathymetries. A variety of objects (e.g., cylinders, rocks, lobster traps) are contained in the imagery. This collection of synthetic data is suitable for training and testing ML algorithms that span a range of complexity, from constant false alarm rate (CFAR) automated detectors to convolutional neural networks (CNNs). These algorithms can perform a variety of tasks that include object detection and classification. In this paper, we test a CFAR detector on image data to estimate object locations. An example use-case of the synthetic data is shown via the training and evaluation of CNN-based classifiers for object recognition.

3:40

4pSPa8. Disentanglement learning for Synthetic Aperture Sonar imagery. Geoff Goehle (Penn State Univ., 225 Sci. Park Rd., State College, PA 16803, goehle@psu.edu) and Benjamin Cowen (Penn State Univ., State College, PA)

The performance of Convolutional Neural Net (CNN) based target recognition algorithms for Synthetic Aperture Sonar (SAS) imagery is strongly dependent on the seafloor background, with worse classification performance for samples with backgrounds outside the CNN training curriculum.

Online performance estimation seeks to identify whether new samples had representation within the training curriculum, and if so whether a given model is expected to perform well on that sample. Performance estimation is critical in underwater object classification, given the diversity of seabed textures and sparsity of training data. Disentanglement learning is a data-driven approach to learning a latent representation that is conceptually interpretable. In the context of performance estimation, we use a disentanglement learning approach based on variational autoencoders to train a latent representation that encodes separately both target class and background type. Online performance estimation is performed by determining if a given sample has a background that is likely to be in the class of backgrounds included in the training data. Results are demonstrated using in-air linear SAS scans from the airSAS dataset.

4:00–4:20 Break

4:20

4pSPa9. An analysis for consistent object classification performance on in-air synthetic aperture sonar data. Raymond M. Thompson (Elec. Eng., Penn State Univ., 1126 Stanwood St., Philadelphia, PA 19111, raytjr13@gmail.com), J. Daniel Park (Appl. Res. Lab. Penn State Univ., State College, PA), and John F. Doherty (Elec. Eng., Penn State Univ., University Park, PA)

Accurate object classification is achieved by minimizing false positives and false negatives across diverse environments. Real-world data often contains significant noise or confounding information, posing challenges to consistent classifier performance. To simulate varying levels of environmental complexity, synthetic aperture sonar (SAS) images were generated using different backgrounds with block letters “O” and “Q.” The objective of this work is to resolve this binary hypothesis test using traditional and modern machine learning (ML) approaches, highlighting their strengths and differences. Traditional methods, such as singular value decomposition (SVD) combined with lasso logistic regression, facilitate evaluating the importance of each feature in classification. Modern approaches like convolutional neural networks (CNNs), which rely on stacked layers of convolutional filters and nonlinear activation functions to learn intricate patterns in the data, capturing both local and global structures. Even CNNs with significantly fewer learnable parameters than more sophisticated models can achieve relatively robust performance, despite the challenges of interpreting the small filters. By exploring these ML approaches, this work provides insights into the trade-offs between interpretability and automation in classification tasks.

4:40

4pSPa10. Optimal evasive maneuver planning for submarines against multi-torpedo threats using reinforcement learning. Eonyak Kang (Sejong Univ., 209, Neungdong-ro, Gwangjin-gu, Seoul 05006, Korea, eonyak1635@sju.ac.kr), wooyoung hong (Sejong Univ., Seoul, Korea), Jongmoo Lee (LIG Nex1, Seoul, Korea), Hyukjae Baik (LIG Nex1, Seongnam-si, Korea), and Youngmin Choo (Sejong Univ., Seoul, Korea)

We propose a methodology for submarine evasion against multi-torpedo threats, extending conventional single-torpedo, geometry-based strategies through the use of reinforcement learning. Recent advancements in marine weaponry have shifted the threat paradigm from single-torpedo launches to simultaneous multi-torpedo attacks, exposing the limitations of existing geometry-based approaches. To address the lack of standardized multi-torpedo evasion tactics, we employ the proximal policy optimization (PPO) algorithm to determine optimal evasive maneuvers in three-dimensional combat scenarios. Our approach models submarine motion dynamics (e.g., turn radius, acceleration, velocity limits), decoy deployment, torpedo pursuit behavior, and sonar systems. By systematically exploring the continuous state space—including submarine positions, headings, and velocities—our PPO-based agent learns adaptive strategies that balance rapid course changes, effectively evading multi-torpedo engagements. We validate the proposed method via MATLAB simulations, demonstrating that the PPO-driven policy outperforms traditional geometry-based approaches in terms of both times to intercept and overall survival probability. This work was conducted with research funding support from LIG Nex1.

5:00

4pSPa11. Learning linear separable spectral features with self-supervised pretrained audio model for sound event detection and localization task. Adeoluwawale Adewusi (Dept. of Elec. Eng. and Comput. Sci., Technische Universität, Berlin, March straße 23, MAR 5.051, Berlin 10587, Germany, adewusi@ni.tu-berlin.de) and Klaus Obermayer (Dept. of Elec. Eng. and Comput. Sci., Technische Universität, Berlin, Berlin, Germany)

Owing to insufficient strongly labeled data for sound event detection and localization (SELD) task, the realization of a robust SELD classifier across varying acoustic conditions without model complexity becomes an uphill task. It has also been shown that the extraction of high-quality features is pivotal to model performance of deep networks. However, most of the current modeling approaches rely on residual networks (CNNs) for feature representations learning from raw SELD audio data. It is therefore pertinent to understand how learning additional linear separable spectral features from the raw SELD datasets improves SELD model performance and reduces the need for data augmentation. To learn these spectral features, we introduced an audio teacher–student transformer frame-wise(ATST-frame) model pretrained on sound event detection (SED) task for this purpose. These supplementary spectral features provide continuous information over the consecutive audio frames which shows the location of the audio event class within the audio data. These extra embeddings are thereafter combined with the extracted representations from the CNNs. The fusion of these complementary features yields enhanced discriminative input features for SELD task modeling. Results show that the extracted linear separable features tremendously improve the performance of the SELD classifier in strong reverberant spatial acoustic scenes.

Session 4pSPb

Signal Processing in Acoustics and Underwater Acoustics: Universal and Doubly Adaptive Methods for Signal Processing II

John R. Buck, Cochair
Electrical and Computer Engineering, UMass Dartmouth, 285 Old Westport Road, Dartmouth, MA 02747

Kathleen E. Wage, Cochair
George Mason University, 4400 University Drive, Fairfax, VA 22151

Andrew C. Singer, Cochair
Stony Brook University, Stony Brook, NY

Invited Papers

1:00

4pSPb1. Algorithms for universal acoustic mode filtering. Bhargabi Chakrabarti (ECE, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, bchakra@gmu.edu) and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

Underwater applications such as tomography often require the estimation of acoustic mode time series from vertical array data. Mode filters are designed to pass a desired mode while suppressing noise and interference. For transient signals or non-stationary environments, the mode filter must adapt its weights over time. We developed the Performance-Weighted Blended (PWB) mode filter that combines pseudo-inverse (PI) mode filters of different ranks to achieve results better than or equal to a single PI filter [IEEE Oceans 2022, Asilomar 2023]. The PWB mode filter is universal, similar to Buck and Singer's universal beamformer [IEEE SAM 2018]. Performance depends on the user's choice of filters to include in the set. This talk explores a different approach to blended mode filter design. Instead of blending a discrete filter set, we use a modified eigenspace beamformer operating in the signal subspace defined using Gram–Schmidt orthonormalization of the sampled modeshapes. This eliminates the *ad hoc* filter design required in the previous approach. Blending different diagonal loading levels in the signal subspace with the PWB algorithm protects against modeshape mismatch. This talk compares the original PWB mode filter with the new eigenspace design. Performance is evaluated using simulated and experimental data. [Work supported by ONR.]

1:20

4pSPb2. Performance-weighted blended FxLMS for changing secondary paths. Kanad Sarkar (Univ. of Illinois at Urbana Champaign, B10 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801-2447, kanads2@illinois.edu), Manan Mittal (Stony Brook Univ., Urbana, IL), Yongjie Zhuang (Stony Brook Univ., Stony Brook, NY), Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, Chicago, IL), and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

The Filtered-x LMS (FxLMS) algorithm is a linear adaptive filtering approach in active noise control that estimates the signal played from a secondary speaker to cancel the presence of a noise source in a microphone. One of the requirements for this algorithm is the filter coefficients of the secondary acoustic path between the secondary speaker and the error microphone. For scenarios where the secondary path is time-varying, we would require external information to determine the current path at each time step. If the secondary path at any time lies in a finite set of possible transfer functions, we present a performance-weighted blended FxLMS algorithm that is robust to rapid changes within the set. We compare the effects of blending paths with the online selection of a single path in the set. We also compare the blended approach with the noise injection method commonly used in online secondary-path estimation.

1:40

4pSPb3. An adaptive blend of truncated order statistics filters for power spectral density estimation. David C. Anchieta (Elec. and Comput. Eng., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, danchieta@umassd.edu) and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

When estimating the background noise power spectral density (PSD) from underwater acoustic recordings, order statistics filters (OSF) effectively mitigate the bias caused by outliers in data, such as broadband loud transients. The Schwock and Abadi [ICASSP, 2019] Welch Percentile (SAWP) is an example of a spectral estimator that scales a single-order statistic (OS) of consecutive overlapping periodograms of acoustic data to estimate the background noise PSD. However, in a dynamic environment, the rate at which loud transients occur is time-varying, requiring the OSF to adjust its rank accordingly to keep low bias and variance. Previously, we proposed applying a mixture of experts to blend SAWP estimators of different ranks according to their short-time performance, thus eliminating the need to explicitly set the OSF rank [Campos Anchieta & Buck, POMA, 2024]. The performance of each estimator is measured by

their sample variance over a fixed time window. In this talk, we apply the same performance-weighted blend (PWB) algorithm to the truncated linear order statistics filter (TLOSF), an OSF that is itself a weighted sum of OS up to a threshold rank [Campos Anchieta & Buck, IEEE JOE, 2024]. When compared to any of their fixed rank counterparts, the PWB versions of both SAWP and TLOSF accumulate less squared error estimating the PSD. When compared to each other, the performance-weighted TLOSF has 0.25–0.5 dB lower mean squared error than the PWB SAWP mainly due to a lower variance. [Work supported by ONR Code 321US.]

2:00

4pSPb4. A blended active sonar receiver that adaptively trades detection gain for range resolution. Radienxe Bautista (Univ. of Massachusetts Dartmouth, 1176 Howell St., Newport, RI 02841, rbautista@umassd.edu) and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

Active sonar systems interrogate their surroundings by transmitting pulses and listening for echoes. A classic approach to range estimation is pulse compression, where a high bandwidth pulse is transmitted and echoes are processed with matched filters. Matched filters optimize output SNR while range resolution is limited by the pulse bandwidth. Extensive research has focused on waveform design for sonar range resolution, often assuming matched filter receivers. However, relatively little research has explored alternative processing for active sonar receivers. Sharma and Buck (2011) proposed the variable resolution and detection receiver (VRDR), which smoothly adjusts range resolution between the matched filter and the inverse filter with one parameter, trading off detection gain for range resolution. In practice, the VRDR requires prior knowledge of the target and noise background to choose the best resolution and detection gain tradeoff. A new receiver is proposed to blend different VRDR receivers with different detection and resolution tradeoffs to adapt to the environment. The outputs of each VRDR receiver are weighted based on performance, using a mixture of experts' approach inspired by universal linear prediction [Singer and Feder, 1999] to implement the blending and minimize the regret of using a fixed individual VRDR filter. [Work supported by ONR.]

THURSDAY AFTERNOON, 22 MAY 2025 — STUDIOS 9/10, 1:00 P.M. TO 2:40 P.M.

Session 4pUWa

Underwater Acoustics, Acoustical Oceanography and Signal Processing in Acoustics: Directional Sensing: Applications and Methods II

Kaustubha Raghukumar, Cochair
Integral Consulting Inc., 200 Washington Street, Suite 201, Santa Cruz, CA 95060

Aaron M. Thode, Cochair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Drive, MC 0206, La Jolla, CA 92093

Kerri D. Seger, Cochair
Applied Ocean Sciences, 11006 Clara Barton Drive, Fairfax Station, VA 22039

Ludovic Tenorio, Cochair
CIMAS/NOAA, 75 Virginia Beach Drive, Key Biscayne, FL 33149

Contributed Papers

1:00

4pUWa1. Towards more accurate sound field verification using directional acoustic filtering. Emily B. Barosin (Thayer School of Eng., Dartmouth College, 26 Oak Point Dr. South, Hanover, NH, emily.b.barosin.25@dartmouth.edu) and Kaustubha Raghukumar (Integral Consulting Inc., Santa Cruz, CA)

Associating omni-directional sound levels with a specific source in the ocean can be challenging when there are other sources of sound such as boats, or biological activity. Here, we present a method to directionally filter acoustic measurements based on acoustic vector sensor measurements of acoustic pressure and particle velocity. The directional discrimination is applied to estimate sound energy from two marine energy sources: the decommissioning of an oil platform and a tidal energy converter. We find that the application of a directional mask leads to distinctly different spectra and some differences in energy, relative to the unmasked scenarios.

1:20

4pUWa2. Directional acoustic measurements around a small-scale tidal turbine. Emma D. Cotter (Coastal Sci. Div., Pacific Northwest National Lab., 1100 Dexter Ave. N, P.O. Box 329, Seattle, WA 98122, emma.cotter@pnnl.gov), Kaustubha Raghukumar (Integral Consulting Inc., Santa Cruz, CA), Joseph H. Haxel (Coastal Sci. Div., Pacific Northwest National Lab., Newport, OR), Polagye Brian (Dept. of Mech. Eng., Univ. of Washington, Seattle, VA), and Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, Seattle, NJ)

Characterizing the underwater sound produced by tidal turbines is essential to understand their environmental impact. Two stationary acoustic vector sensor systems and an array of drifting hydrophones were deployed in the vicinity of a small tidal turbine (i.e., <2 kW rated power) in Sequim Bay, WA. The turbine itself was also equipped with hydrophones on its foundation. Both directional sensing modalities effectively localized and characterized sounds from the turbine. Turbine sound consisted of tones between 100 Hz and 4 kHz and their harmonics, intermittent impulses, and frequency-modulated sounds that varied with turbine operation. Directional processing facilitated the identification of low-intensity sounds from the turbine system that would likely have been overlooked in measurements from a single hydrophone. The vector sensors effectively characterized changes in turbine sound throughout the turbine deployment, and the drifting hydrophones measured turbine sound during peak flow speeds with minimal flow noise contamination. While the deployment of directional acoustic sensors for acoustic characterization of all marine energy converters is neither feasible nor recommended, they can provide uniquely valuable information and simplify the attribution of sounds to sources in an acoustically complex environment.

1:40

4pUWa3. A scalable acoustic system for ropeless fishing applications. Oscar A. Viquez (Acbotics Res., LLC, 82 Technol. Park Dr., Falmouth, MA 02536, oviquezr@acbotics.com), Nicholas Rypkema, Sam Fladung (Acbotics Res., LLC, Falmouth, MA), and Erin M. Fischell (Acbotics Res., LLC, East Falmouth, MA)

An increasingly urgent need in underwater acoustics is a low-cost, easy to deploy solution to fishing without vertical lines in the ocean. We are working with the Lobster Foundation of Massachusetts as a part of a National Fish and Wildlife Foundation (NFWF) project "Grappling with Technology" to demonstrate a one-way travel ultra-short baseline (USBL) approach to low-cost pingers on lobster traps. We present the pinger technology developed as a part of this program, as well as the AcLocalize, a USBL with automatic multi-contact tracking for collaborative and non-collaborative targets. Due to the limited processing power and cost constraints of embedded systems, and the need for robust, predictable performance in consumer-facing applications, the selected approach uses a layered architecture consisting of well-understood methods at each individual stage. These layers can then be instantiated systematically to accommodate an increasing number of contacts, up to a performance limit driven by the computational power available on the embedded system, and by the choice of array geometry. Results from simulations and early field tests will be presented, with a focus on the use of multi-contact tracking in ropeless fishing operations, as an alternative to using surface buoys known to cause whale entanglement.

2:00

4pUWa4. Vector acoustic properties of acoustic modes as generated from explosive sources in a shallow-water waveguide. Robert W. Drinnan (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6785, drinnan@uw.edu), Peter H. Dahl, and David Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Vector acoustic measurements of explosive sources can show structure in the sound field that is not discernible from traditional pressure-only measurement. This structure, derived from coherent combinations of acoustic velocity and pressure, is known to be sensitive to sediment layering and composition. Measurements made during the 2022 Seabed Characterization Experiment along a range-independent section of the New England Mud Patch (NEMP) are presented. Recordings of 32 explosive SUS charges at ranges of 4–15 km show significant dispersion of 4 modes allowing for vector properties of individual modes to be inferred. These vector properties are compared with results from the acoustic normal mode model ORCA based on a nominal geoacoustic model of the NEMP. The results are analyzed for their usefulness in locating modes and characterizing seafloor acoustic properties.

2:20

4pUWa5. Laser-acoustic detection of objects buried underwater by measuring vibration of sediment. Vyacheslav Aranchuk (National Ctr. for Physical Acoust., Univ. of Mississippi, 145 Hill Dr., University, MS 38655, aranchuk@olemiss.edu), John D. Heffington, and Ina Aranchuk (National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

The laser-acoustic technique proved to be effective for the detection of landmines buried in dry land. It is based on exciting vibration of ground and measuring the vibration of ground surface with a laser Doppler vibrometer. The application of this technique for the detection of landmines buried in sediment underwater is impacted by the effect of water layer on the propagation of sound and a sensing laser beam. We experimentally investigated laser-acoustic detection of objects buried in the sediment underwater. Laboratory experiments were conducted for two measurement scenarios: a sound source and a laser vibrometer positioned in air above water surface and a sound source and a laser vibrometer positioned underwater. Effects of surface waves in water, a turbulent water flow and water turbidity on detection of buried objects were investigated. [Work supported by the Office of Naval Research under Award No. N00014-21-1-2247.]

Session 4pUWb

Underwater Acoustics: Underwater Acoustic Communications

Jennifer J. Johnson, Cochair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543

Adam D. Kingsley, Cochair
RDA, Inc., Warrenton, VA 20187

Contributed Papers

3:20

4pUWb1. Underwater acoustic channel simulation using conditional diffusion models. Ningyuan Yang (Stony Brook Univ., Stony Brook, NY, ningyuan.yang@stonybrook.edu), Yongjie Zhuang (Stony Brook Univ., Stony Brook, NY), Manan Mittal (Stony Brook Univ., Urbana, IL), and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

For underwater acoustic (UWA) applications that require a large set of UWA channel realizations such as outage-capacity analysis or network simulation, experimental data captures realistic effects such as multipath propagation and large spatiotemporal variability but is costly to collect. Physics-based tools such as BELLHOP or other acoustic propagation tools are easier to simulate different channel realizations but lack accuracy in capturing realistic channel characteristics. Previous work showed that using unconditional diffusion models can effectively generate a large set of channel realizations from a limited set of experimental data. However, each ocean environment requires the training of a separate diffusion model. Thus, this method lacks adaptability to different ocean environments. In this talk, we propose the use of a conditional diffusion model for generating UWA impulse responses, where the model is conditioned on environmental parameters such as signal depth, channel width, and varying ocean characteristics. Unlike unconditional models that generate data independently of these parameters, our approach allows the diffusion model to learn the underlying relationships between the environmental conditions and the corresponding acoustic signals. Experimental results show that the generated data aligns closely with real-world measurements, which serves as a flexible tool for simulating UWA channels.

3:40

4pUWb2. Reducing the noise by estimating the acoustic channel and removing active sonar reverberation. Adam D. Kingsley (RDA, Inc., 6593 Merchant Pl., Warrenton, VA 20187, adam.kingsley@rdainc.com) and Josh Kjar (RDA, Inc., Warrenton, VA)

Active sonar signals are received by sonobuoy platforms that must not only characterize the signal received but also confirm that no other signal was masked by the arrival of the large signal. If the original signal can be identified, it is possible to estimate the impulse response of the channel over the bandwidth of the original signal. With proper conditioning, it is also possible to then remove the identified signal from the received waveform. It is then easier to search the residual waveform for additional signals of interest. This paper outlines the constraints, methodology, and simulated results of detecting and identifying masked signals in a bottom-limited ocean environment during the simultaneous reception of multiple masked signals. [This paper is based on work supported by the US Navy through the Small Business Innovation Research program.]

4:00

4pUWb3. Reciprocal sensitivity kernels: Optimizing source/receiver positionning in acoustic systems. Alexis Bottero (DGA, Av. De la tour Royale, Toulon 83000, France, alexis.bottero@gmail.com) and Cédric Bellis (LMA, Marseille, France)

In their conventional formulation, sensitivity kernels are employed to identify the regions within a medium where an infinitesimal alteration in its properties would most affect a pressure (or displacement) measurement at a fixed observation point. These kernels typically exhibit elongated, cigar-like geometries, characterized by minimal sensitivity along the direct propagation path. In this work, we extend this framework to address the reciprocal problem, where the goal is to determine the regions within the medium that are most (or least) affected by a given perturbation when the sources (or receivers) locations are held constant. The resulting reciprocal kernels display flared geometries, emphasizing sensitivity in the areas surrounding the perturbation. Potential applications of these findings to optimize source/receiver positioning in underwater acoustic systems are explored.

4:20

4pUWb4. Simulating the feasibility of Ocean Acoustic Waveguide Remote Sensing (OAWRS) for Great Lakes fisheries monitoring. Evan Lucas (Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, eglucas@mtu.edu), Steven Senczyszyn (Mech. Engineering-Eng. Mech., Michigan Technolog. Univ., Houghton, MI), Jacob Krier, Yaman Aljnadi, and John Lothian (Michigan Technolog. Univ., Houghton, MI)

Ocean Acoustic Waveguide Remote Sensing (OAWRS) has been used successfully in continental shelf regions as a method of monitoring fish populations. In this work, we explore the application of OAWRS to a region of Lake Superior as a tool for observing fish behavior on an ecologically important spawning bed, Buffalo Reef. Buffalo Reef provides a challenging area to image with OAWRS due to the varying bathymetry. To demonstrate feasibility for future experiments, we perform beam tracing simulations showing the insonification region we can use to monitor Buffalo Reef's spawning beds from a practical source and receiver array location. We also simulate impulse responses anticipated for a solid fish target, which can be extended to more realistic fish school responses with additional simulation work.

4:40

4pUWb5. Doppler estimation technique using continuous wave signals in hyperbolic frequency modulated preamble based underwater acoustic communication. Dan-bi Ou (Korea Martime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Busan 49112, Korea, eksq1018@naver.com), Hyung-In Ra, and Ki-Man Kim (Korea Martime and Ocean Univ., Busan, Korea)

In this presentation, we propose a novel Doppler estimation technique that employs hyperbolic frequency modulated (HFM) signals in underwater

acoustic communication. The Doppler effects resulting from rapidly changing channels significantly degrade communication performance, especially in fast-moving platforms such as autonomous underwater vehicles (AUVs). Therefore, the Doppler shift must be estimated and compensated. The previous method relied on sequences of down- and up-chirps, calculating the correlation function for each chirp, and estimating Doppler shifts based on time or frequency changes from the present correlation peak to the next correlation peak. In contrast, the proposed method uses continuous-wave (CW) signals instead of chirp signals to correlate with the received signal and analyze its variations. It does not require the knowledge of precise modeling of the transmitted HFM signals. Furthermore, it calculates Doppler shifts across the entire bandwidth, thereby allowing the use of a larger dataset. The performance of the proposed method will be compared with that of previous techniques by the sea trial. [This work is supported by Korea Research Institute for defense Technology planning and advancement (KRIT) grant funded by the Korea government (DAPA) (KRIT-CT-23-035-01, Multi AUV operation Technology for Mine Detection ('23–'28))]

5:00

4pUWb6. Acoustic communication and modeling in shallow water constrained environments. Jennifer J. Johnson (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, jjjohnson@whoi.edu), Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Michael B. Porter (Heat, Light, and Sound Res., Inc., San Diego, CA), Peter Traykovski, and Sandipa Singh (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Sound behavior in shallow water constrained environments such as ports, harbors, and estuaries renders unique challenges and acoustic observations. Improving sound models in these underwater environments offers many advantages, for instance an increased effectiveness in model-based design of underwater acoustic systems enabling harbor security. Our ongoing efforts focus on acoustically characterizing these complicated channels, analyzing acoustic communication system performance, and refining a 3-D beam tracing propagation model for various shallow water sites. This talk will highlight field observations from a bistatic experiment conducted in an industrialized tidal estuary that provided transmission paths in very shallow (<3 m) to deep (approximately 15 m) water depths up to 1500 m in range. Signals transmitted, which included acoustic communication data packets, along various transmission paths and ranges at 8–34 kHz resulted in considerable data for assessing and explaining transmission performance variability. The observations from across the site are compared with simulations from the model to identify feature and boundary reflection properties that contribute or dominate echo structure. A verified comprehensive modeling ability for these dynamic, and acoustically distinct, constrained environments is ideal for executing successful underwater operations in the context of acoustic modem throughput and optimization of other systems. [Work supported by ONR.]

5:20

4pUWb7. Reduced cyclic prefix- orthogonal time frequency space modulation in underwater acoustic communication: Pulse shaping strategies for the practical implementation. Deeksha Varshney (IIT, Delhi, IIT Delhi Main Rd., IIT Campus, Hauz Khas, New Delhi 110016, India, bsz198698@iitd.ac.in), Monika Aggarwal (IIT, Delhi, New Delhi, India), and Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

Orthogonal Time Frequency Space (OTFS) modulation has recently been proposed for wireless communication and has demonstrated superior performance to OFDM in underwater channels in terms of error rates and complexity. However, existing research primarily focuses on ideal or rectangular pulse shaping, limiting its applicability to practical scenarios. In this work, we derive the input–output relationship using practical pulse-shaping waveforms, such as rectangular pulses, in a simplified matrix form. This derivation is then extended to accommodate arbitrary waveform designs. We further evaluate the performance of a reduced cyclic prefix (RCP) OTFS-based underwater acoustic communication (UWAC) system with various practical pulse-shaping waveforms. The system's performance is assessed using the Bellhop-based statistical channel model with a channel length of 500 m and depth of 100 m, where the transmitter and receiver are located at 98 and 85 m, respectively. The channel exhibits a delay spread of 22 ms and a Doppler scaling factor of 10^{-3}. The OTFS system employs an 8×8 grid, 4-QAM modulation, 4 kHz bandwidth, 2 ms symbol duration, and achieves delay and Doppler resolutions of 0.25 ms and 62.5 Hz, respectively. Using rectangular pulse shaping, we achieved a BER of 10^{-2} with robust detection via a message-passing algorithm.

Session 5aAA

Architectural Acoustics: Theatres, Auditoria, and Other Gathering Spaces

Molly Smallcomb, Cochair
Threshold Acoustics, 141 West Jackson Blvd., Suite 2080, Chicago, IL 60604

Heui Young Park, Cochair
Pennsylvania State Univ., State College, PA 16801

Contributed Papers

7:00

5aAA1. Abstract withdrawn.

7:20

5aAA2. Audiovisual virtual reality experiment on the distance effect on loudness perception in concert halls. Gülnihan Atay (Inst. of Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, guelnihan.atay@akustik.rwth-aachen.de), Josep Llorca-Bofí (Dept. of Architectural Representation, Univ. Politecnica de Catalunya, Barcelona, Spain), Nils Rummler, and Michael Vorlaender (Inst. of Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

In this study, perceptual assessment of loudness based on the visual distance information is examined. The effect of source-receiver distance on loudness perception in Munich Gasteig Concert Hall is aimed to be studied through a series of listening tests. The listening tests are designed to be conducted in a room acoustics simulation embedded in a virtual reality environment. The acoustic simulation uses the loudspeaker orchestra published by Pätynen *et al*. In the first step, the simulation is validated against the previously published measurement results. In the second step, the initial phase of the perceptual listening tests is conducted. The preliminary results are presented and discussed.

7:40

5aAA3. The use of electrocoustic enhancement systems in the design of orchestral rehearsal rooms. Cameron Hough (Marshall Day Acoust., 10/50 Gipps St., Collingwood, Victoria 3067, Australia, chough@marshall-day.com), Nick Boulter (Arup, Oxford, United Kingdom), and Simon Tait (AmberTech, Melbourne, Victoria, Australia)

Rehearsal rooms for orchestras pose design challenges in controlling room loudness as a fundamental health and safety issue. Given the practical constraints on volume, providing sound absorption is an way of reducing the overall sound level and the masking effect of strong reverberation. However, this can make the room too "dry" which can negatively affect tone production, and make ensemble blend more difficult. Also, the decay characteristics of a room are a prerequisite for the musicality of the rehearsal. Are smaller-scale rehearsal rooms, therefore, always acoustically compromised? Electroacoustic enhancement systems offer a means to allow loudness, ensemble, and musicality factors to be addressed in a physically smaller environment. With undesirable reflections managed via absorption and diffusion, new "useful" reflections can be added digitally without requiring physical surfaces. These can be flexibly mapped to instrumental sections that require them—something very difficult to achieve otherwise. Late sound can be created at a lower gain so it does not result in excessive loudness, whilst still allowing a longer decay rate to enhance the musical blend. A case study of the Australian Chamber Orchestra's rehearsal and performance studio is presented, including experiences from the process of tuning the room with the Orchestra.

8:00

5aAA4. Reconstruction of acoustics of the ancient Roman theatre in Aspendos. Piotr Wojdyllo (Światowida 14 m. 102, Warszawa 03-144, Poland, p@ancient-acoustics.com)

The reconstruction solution presented here is implemented in the form of Python software functions, which is advantageous for adding as many details as possible—if only they are regular. The model is based on the actual Aspendos theatre dimensions supplemented by the data from M. Vitruvius' text and with the logic of forward ray tracing. The second level is dispersion coming from the surface irregularities. The third level is an analysis of the composition of the surfaces. It is implemented to high detail and although the first few millimeters of depth are of most importance, we do not neglect deeper parts, too. Here, we use the "path integral" approximation of the wave equation solution. The wall composition is analyzed to reconstruct impulse response of the wall which is then used in the convolution powers dependent on the count of reflections in the walls of the specified composition. The T30 parameters derived from the 20 reconstructed impulse responses aggregated across the two lines of receivers were compared with the values measured *in situ* and MAPE (Mean Absolute Percentage Error) went down from 8% to 6.5%. The relative changes between different frequencies were also more accentuated compared to the state-of-the-art solution. Also the outcome is not dependent on the porosity level.

8:20

5aAA5. Determining the sound power of musical ensembles in realistic performance conditions. Ingo B. Witew (Audio Commun. Group, TU Berlin, Einsteinufer 17c, Sekr. EN-8, Berlin 10587, Germany, ingo@witew.de), Max Wiedemann, and Stefan Weinzierl (Audio Commun. Group, TU Berlin, Berlin, Germany)

The acoustic optimization of performance spaces for musical ensembles depends on achieving suitable sound levels to ensure auditory balance and performer comfort. Accurate sound power data for individual instruments and ensembles, particularly under realistic performance conditions, are essential for this purpose. While prior studies have largely relied on laboratory-based measurements, these results may not fully capture the dynamic, non-stationary nature of live performances or the influence of performance space acoustics. This study explores the extent to which reliable sound power measurements can be obtained in realistic conditions, employing both natural and artificial excitation signals. By addressing challenges, such as varying musical intensities and the acoustic variability of performance spaces, the research bridges the gap between controlled experiments and real-world scenarios. Using the ISO 3743-1 framework, sound power statistics were determined for various chamber music ensembles.

8:40

5aAA6. Characterizing large public spaces at an aquarium. Taylor A. Jackson (Phys. and Astron., Brigham Young Univ., N284 ESC, Provo, UT 84602, haycock1@byu.edu), Michael M. Hogg, Jason D. Bickmore, Peter K. Jensen, Joshua T. Mills, Dallin E. Jackson, Tracianne B. Neilsen (Phys. and Astron., Brigham Young Univ., Provo, UT), Michael Ashcroft, and Ari Fustukjian (Loveland Living Planet Aquarium, Draper, UT)

Aquariums often have a few large and busy public areas attached to aquatic habitats, with the sound levels in the rooms possibly affecting the sound levels in the tanks. In this project, we wanted to characterize these large public rooms at the Loveland Living Planet Aquarium. Specifically, we did room characterization measurements in the viewing area and tunnel by the shark tank. Sound level meters and microphones recorded white noise and chirps. The data have been analyzed, and sound pressure levels, impulse responses, and reverberation times are presented. These results are compared to prior characterization of similar public spaces. The characterization of these public spaces is tied to a larger study regarding sound transmission into the shark tank. Preliminary results from the hydrophone recordings are shown to complement the room characterization efforts. [Undergraduate research assistance provided by the College of Computing, Mathematical, and Physical Sciences at Brigham Young University.]

9:00–9:20 Break

9:20

5aAA7. A study on noise disturbance and sound exposure of large-scale indoor pop concerts. Wei-Hwa Chiang (National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd. Sec. 4, Taipei 10607, Taiwan, whch@mail.ntust.edu.tw), Hui-Ping Wu, and Chun Chai (National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan)

In recent years, large-scale indoor pop concerts have become popular in Taiwan, but they have caused a variety of acoustic problems inside and outside the buildings. Under the premise of having a reasonable indoor acoustic environment, this study investigates the disturbance to neighbors and other building users, as well as the safe sound exposure of audiences and performing groups. It includes case studies of existing and on-going projects in Taiwan and spectrum comparison of commercial sound tracks with live concerts. Special attention was paid to the control of low-frequency components and the awareness of noise and health issues by performing groups. Preliminary results will be presented for architectural design, speaker system configuration, and volume warning application.

9:40

5aAA8. Impact of diverse genres of sacred music on the Revitalising Soundscape of Mission Concepcion Church in San Antonio, Texas. Menino A. Tavares (Acoust. & Soundscaping of Revitalising Sacred Environment, Archdiocese of San Antonio, 510 Wernett St., Del Rio, TX 78840, minino.tavares@archsa.org), Wade R. Bray (HEAD Acoust., Inc., Brighton, MI), and Rebecca Simmons (Las Misiones, Archdiocese of San Antonio, San Antonio, TX)

This soundscape study (based on ISO 12913-1/2/3) uses rendition of sacred music by clarinet, violin, saxophone, human voice, and ensemble testing different locations in the church to comparatively analyze binaural perceptions of loudness, sharpness, and tonality (using Artemis SUITE version 16.5 and HEADspace-5600); subjective acoustical perceptions of loudness, clarity, tonal balance, and intimacy; and auditory metaphysical perceptions of heightened awareness and stillness (using ORIGIN 6.1 and EXCEL). Among significant results ($p<0.05$): loudness (N5) had higher values for ensemble rendition at Nave and Choir-loft and for saxophone rendition at President's window; Tonality was highest for saxophone rendition and Sharpness was highest for solo-voice rendition at all locations. Saxophone was perceived to be "loudest," "clearest," "most tonal," "most intimate," and evoked better "heightened awareness" and "stillness" at President's Window; "Solo-voice" was "clearest," "most tonal," and "most intimate" at Choir-loft. Solo-voice rendition evoked "stillness" at Nave at $p = 0.07$. This study is part of efforts to explore the functional diversity of the revitalising soundscape that initiates a tourist to experience the "Sacred" in this world heritage worship space.

10:00

5aAA9. Comparative acoustic study of atriums in Toronto. Bryce Lemert (Aercoustics Eng. Ltd, 1004 Middlegate Rd., Ste. 1100, Mississauga, ON L4Y 0G1, Canada, brycel@aercoustics.com)

Atriums have traditionally been overlooked in the acoustic design of buildings as they have primarily served as transitional spaces with minimal acoustic treatment. However, as atrium designs evolve to accommodate multipurpose functions where people may be gathered for longer periods of time, the need for effective acoustic treatment has become increasingly important. This study shares the results of measurements taken in atriums across Toronto. The analysis looks at how different room sizes, shapes, and materials affect the sound quality in each space, both overall and in specific areas. Insights from the findings offer ideas for improving the acoustic design of future atriums and suggest the best ways to adjust sound treatments for different needs.

Session 5aAB

Animal Bioacoustics: Animal Vocal Communication and Physiology

Kaitlin Frasier, Cochair
Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093-0205

Xavier Mouy, Cochair
Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543

Contributed Papers

8:00

5aAB1. Fin whale song stability and evolution in the North Atlantic. Regina A. Guazzo (NIWC Pacific, 53560 Hull St., San Diego, CA 92152, regina.a.guazzo.civ@us.navy.mil), Dorene L. Stevenson, Michael K. Edell, George J. Gagnon (Marine Acoust., Inc., Arlington, VA), and Tyler Helble (NIWC Pacific, San Diego, CA)

Low-frequency pulses of fin whale song have been recorded in every ocean basin with patterns that shift over time, both gradually and suddenly. Two primary note types are interwoven in song patterns with both singlet and doublet inter-note intervals. Fin whale population size, structure, distribution, and connectedness are not well understood, but passive acoustic monitoring is a tool that could be applied to improve management decisions for this species. We analyzed 119 fin whale tracks from 2013–2023 generated with US Navy arrays in the North Atlantic to define song patterns, identify changes over time, and calculate cue rates. We observed four distinct song patterns with inter-note intervals for one pattern increasing over inter-annual timescales and inter-note intervals for another pattern increasing over intra-annual timescales. We report that a higher frequency upsweep note has been decreasing in frequency for at least 30 years. Since passive acoustic monitoring has been suggested as a tool to estimate abundance, we examined two different options for "cues" (individual notes and gaps between song bouts) and compared their stability over time. We contrast these results with our previous findings from fin whales in the North Pacific. Many of these nuances in fin whale singing behavior have not been described in previous research and can be applied to assessing responses to disturbance and estimating the abundance and distribution of a poorly understood species. [Work supported by US Navy's LMR Program, US PACFLT, OPNAV N974.]

8:20

5aAB2. Acoustic behavior of endangered false killer whales (*Pseudorca crassidens*) using biologging devices in Hawai'i. Brijonnay Madrigal (Marine Mammal Res. Prog., Univ. of Hawai'i at Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, bcm2@hawaii.edu), William Gough, Lars Bejder, Jens Currie (Pacific Whale Foundation, Maalaea, HI), Aran Mooney (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Aude Pacini (Marine Mammal Res. Prog., Univ. of Hawai'i at Manoa, Kaneohe, HI)

The main Hawaiian Islands insular false killer whale (*Pseudorca crassidens*) population is endangered due to anthropogenic stressors; therefore, understanding the acoustic behavior of this social species is imperative to conservation. Previous studies have used satellite telemetry tags to track movement, distribution, and diving behavior but provide little information on behavioral context related to acoustic communication and foraging behavior. Suction cup multi-sensor Customized Animal Tracking Solutions (CATS) tags and a digital acoustic recording tag (DTAG) containing hydrophones (SR 96 and 240 kHz, respectively) and a camera (CATS) were deployed off the main Hawaiian Islands in 2011 and 2023–2024 ($n = 4$). Cumulatively, 21 h of acoustic data were recorded. Calls were manually detected/classified using Raven Pro and features extracted using PAMGuard ROCCA. Individuals predominantly produced rarely documented pulsed calls and repertoires consisted of >10 previously undescribed stereotyped calls. Call types were associated with dive phase and call rates varied across individuals. Nonlinear phenomena were common including biphonation, frequency jumps, and deterministic chaos. This study marks the first-time concurrent audio/video tags have been deployed on odontocetes in Hawai'i and provides high-resolution insight into call function and behavioral states to supplement passive acoustic monitoring efforts and inform species management and conservation.

8:40

5aAB3. Identifying dolphin whistle producers with machine learning: Moving beyond signature whistles. Brittany L. Jones (Marine Mammal Program, NIWC Pacific, 52560 Hull St., San Diego, CA 92152, jonesey.brittany@gmail.com) and Maximilian Du (Stanford Univ., Stanford, CA)

Bottlenose dolphins produce a range of vocalizations, including signature whistles, shared whistles, copies of others' signature whistles, and non-signature whistles. While signature whistles are individually unique and often easily recognizable by human observers, shared whistles and signature whistle copies have proven challenging to classify accurately. Non-signature whistles, meanwhile, remain largely unexamined for individual identification through visual or aural analysis. This study evaluated the use of machine learning (ML) to classify dolphin whistle producers across multiple whistle types. Using the extensive whistle catalogue from the U.S. Navy's Marine Mammal Program's bottlenose dolphins, we assessed ML based classifiers for identifying individual producers. Preliminary findings suggest that the classifier performs better than chance for all whistle types, with the highest accuracy observed for signature whistles. Historically, even with a well calibrated hydrophone array, the localization error, speed of movement, and close proximity of social groups has made identifying the caller a difficult problem to overcome when studying odontocete acoustic behavior. This is a first step toward overcoming this issue for well-studied groups.

9:00

5aAB4. Assessing acoustic occurrence of the endangered Franciscana dolphin near artisanal fishing nets in San BorombomBay, Argentina. Gisela V. Giardino (Ciencias Marinas, Instituto de Investigaciones Marinas y Costeras (IIMyC), FCEyN, UNMdP-CONICET., Funes 3360, Mar del Plata, Buenos Aires 7600, Argentina, gvgiardi@mdp.edu.ar), Fernanda Zapata (Centro de Estudios en Ciencias Marinas, Aquamarina, San Clemente del Tuyu, Argentina), Mel Cosentino (Section for Marine Mammal Res., Aarhus Univ., Aarhaus, Denmark), Ricardo Bastida, and Diego Rodriguez (Ciencias Marinas, Instituto de Investigaciones Marinas y Costeras (IIMyC), FCEyN, UNMdP-CONICET, Mar del Plata, Argentina)

Bycatch is a major threat to small cetaceans all over the world. In Argentina, the Franciscana dolphin (*Pontoporia blainvillei*) is especially impacted, with over 500 dolphins entangled annually in artisanal gillnets in Buenos

Aires Province alone. Classified as vulnerable, this coastal species is known for its small size, cryptic coloration, and minimal aerial displays. This study aims to detect Franciscana dolphins' acoustic presence around artisanal gillnets with active bycatch. High-frequency acoustic detectors (C-PODS and F-PODS, Chelonia Ltd.) were deployed in two artisanal fishing nets in the Bahia San Borombon estuary (36°10′676″ S, 57°08′863″ W), Argentina, from October 28, 2023, to January 15, 2024, totaling four deployments. Detectors averaged two days per deployment, with a maximum of six continuous days. Narrowband high-frequency clicks (NBHF), associated with Franciscana dolphins, were recorded 2% of the time in the first deployment, 31.3% in the second, 17.4% in the third, and 17.7% in the last. A peak in detections per minute was observed around midday, coinciding with high tide. The preliminary results from this ongoing investigation provide valuable insight into the interactions between Franciscana dolphins and artisanal fishing activities.

9:20–9:40 Break

9:40

5aAB5. Information theory of leopard seal (*Hydrurga leptonyx*) acoustic communication sequences. Lucinda E. Chambers (Ctr. for Marine Sci. and Innovation (CMSI), UNSW Sydney, UNSW Sydney High St., Kensington, New South Wales 2052, Australia, lucinda.chambers@student.unsw.edu.au), Tracey Rogers, and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

Leopard seals are solitary animals, which communicate via acoustic sequences during their breeding season, producing a sequence of stereotyped vocalisations underwater. These vocalisation bouts consist of ordered sequences of discrete elements from a shared vocal repertoire, and the patterns that underlie the choice of sounds in these acoustic displays could be used for encoding individual identity. To investigate the properties of these sequences, we used information theory to estimate the entropy of the calling bouts of a population of 26 leopard seals from Davis Sea, Eastern Antarctica. We used three estimators for comparison – an independent identically distributed (i.i.d.) model, a first order Markov model, and a sliding window match length (SWML) estimator. The majority of the SWML estimates are lower than the i.i.d. and first order Markov estimates, suggesting that the leopard seal vocal sequences have more complex temporal structures than can be modelled by first order Markov chains. This finding suggests that the temporal structure of male leopard seal song conveys information to receivers of male mating displays. Our results highlight the potential of the temporal structure of these stereotyped sequences to convey cues of individual identity in leopard seal calling bouts.

10:00

5aAB6. Vocalizations in a male-only houlout of the South American Sea Lion (*Otaria flavescens*) in Argentina. Ingrid Holzmann (Ecoson, CONICET, Vaqueros, Salta, Argentina) and Gisela V. Giardino (Ciencias Marinas, Instituto de Investigaciones Marinas y Costeras (IIMyC), FCEyN, UNMdP-CONICET, Funes 3360, Mar del Plata, Buenos Aires 7600, Argentina, gvgiardi@mdp.edu.ar)

In South American Sea Lions (*Otaria flavescens*) when the austral breeding season ends and harems dissolve, mothers and pups usually remain near the breeding area and males disperse to congregate in male-only houlouts. Our objectives were to identify, for the first time, the male vocalizations in a houlout in Mar del Plata (Argentina) when immediate reproductive pressure was relaxed. Between the 2 and 10 July 2023 we recorded males using a Marantz PMD 661 and a directional microphone Sennheiser ME67. Recordings were analyzed with spectrum-based automatic measurements in Avisoft-SASLab Pro 5, characterizing: 1-frequency values from maximum peaks in amplitude (Hz), frequency values at 25% and 75% (Hz) energy of the call and call duration (s). We obtained 249 recordings (mean duration: 1m39s, range: 3s-490s) with 14720 vocalizations, all classified within four types: 1- *Barks* (the most frequent call, n = 8544; 58.04%), 2- *Growls* (n = 4324; 2.37%), 3- *Exhalations* (n = 1069; 7.26%), and 3- *High Pitch Calls* (n = 783; 5.31%). Despite our short study, we found the same call types described previously in a breeding colony in southern Argentina. Probably, male vocal repertoire in this species is restricted to these four call types and changes in rate/order could occur when competition for harem formation happen.

10:20

5aAB7. Establishing acoustic baselines for Lahille's bottlenose dolphin (*Tursiops truncatus gephyreus*) in Argentina. Agustina Macchi (Marine and Coastal Res. Inst. (IIMyC), Dept. of Marine Sci., Faculty of Exact and Natural Sci., National Univ. of Mar del Plata, CONICET. Mar del Plata, Argentina., Mar del Plata, Argentina), Juan Pablo Loureiro, Florencia Speciale, Sergio Moron (Fundación Mundo Marino, San Clemente del Tuyú 7105, Argentina;, San Clemente del Tuyú, Argentina), Diego Rodriguez, Ricardo Bastida, and Gisela V. Giardino (Marine and Coastal Res. Inst. (IIMyC), Dept. of Marine Sci., Faculty of Exact and Natural Sci., National Univ. of Mar del Plata, CONICET, Mar del Plata, Argentina., Funes 3360, Mar del Plata, Buenos Aires 7600, Argentina, gvgiardi@mdp.edu.ar)

The Lahille's bottlenose dolphin (*Tursiops truncatus gephyreus*), distributed from southern Brazil to southern Argentina, is classified as Vulnerable by the Argentine Society of Mammalogy (SAREM).The population of Buenos Aires has drastically declined since the 1980s, leaving small groups in Buenos Aires and Río Negro. This study provides the first acoustic characterization of this subspecies in Argentina, focusing on vocalizations recorded in a controlled environment (Mundo Marino facility). Recordings were made using HydroMoth hydrophones at 96 kHz. Acoustic analyses included whistle emission rates and robust parameters: peak frequency, duration, and bandwidth 90%. A total of 371 min were analyzed. The whistles exhibited a peak frequency of 8.67 ± 1.93 kHz, a duration of 0.7 ± 0.49 s, and a bandwidth of 6.55 ± 2.74 kHz. Signature whistles were observed, with a convex shape predominant (77.7%). Similarities were found in the whistle duration on a previously studied in Uruguay, but differences in maximum frequency were higher in Uruguay dolphins (22.4 kHz versus 14.15 kHz in Mundo Marino). The results validate controlled environments as essential for establishing acoustic reference data and support future comparisons with recordings from natural environments. This provides a reference for studying their acoustic behavior, with implications for conservation.

Session 5aBA

Biomedical Acoustics: General Topics in Biomedical Acoustics: Quantitative Ultrasound

Marvin Doyley, Cochair
Univ. of Rochester, Rochester, NY 14623

Scott J. Schoen, Cochair
Radiology, Harvard Medical School and Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114

Contributed Papers

7:40

5aBA1. Quantitative ultrasound assessment of transarterial radioembolization treatment of LI-RADS 5 hepatocellular carcinoma. Michael P. Andre (Radiology, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, mandre@ucsd.edu), Aiguo Han, Jingyi Zuo (Biomed. Eng. and Mech., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), Haotian chen (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), Zachary Berman, Mansur Ghani (Radiology, Univ. of California, San Diego, La Jolla, CA), Adam Burgoyne (Medicine, Medical Oncol., Univ. of California, San Diego, La Jolla, CA), Claude Sirlin (Radiology, Univ. of California, San Diego, La Jolla, CA), Matthew A. Wallig, William D. O'Brien (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), and Yuko Kono (Medicine, Hepatology, Univ. of California, San Diego, La Jolla, CA)

This prospective pilot study examines feasibility of ultrasound (US) imaging methods to evaluate patients with known HCC prior to and after treatment with transarterial radioembolization (TARE). Quantitative backscatter US (QUS) and contrast-enhanced US (CEUS) were successfully performed in the tumor and background liver, using contrast-enhanced (CE) MRI as the reference. Scans were performed pre-TARE and at 6- and 12- or 24-weeks post treatment. Enrollment of 25 patients is planned, but in 15 patients to date (age 47–94, mean 71), all were MRI LI-RADS 5 with tumor diameters 2.1–13.8 cm, had uniformly cirrhotic background liver with low steatosis. Attenuation (AC), backscatter (BC) coefficients, and envelope statistics were computed, with results in the background liver very consistent between acquisition timepoints and patients. Tumor QUS was more heterogeneous but QUS tumor-background contrast was significant. Changes in tumor AC and/or BC post-TARE were significant for all but two patients but variable. Based on MRI data, some tumors may exhibit more delayed response to TARE, so examinations at additional timepoints as well as new patient recruitment are in process. All tumor boundaries corresponded well to CEUS and CE MRI to identify tumors, demarcate tumor size, and boundaries, as well as identify response to therapy.

8:00

5aBA2. A novel quantitative ultrasound metric for osteoporosis diagnosis using instantaneous phase. Hugh E. Ferguson (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, ferhe-27@rhodes.edu), Brent K. Hoffmeister, and Ann M. Viano (Phys., Rhodes College, Memphis, TN)

Quantitative ultrasound (QUS) has emerged as a promising and cost-effective method for assessing bone health. Traditional QUS parameters such as broadband ultrasound attenuation and speed of sound are valuable; however, they are difficult to measure at central skeletal sites such as the hip and spine where most osteoporotic fractures occur. This study introduces a novel QUS method derived from the instantaneous phase of the backscattered ultrasound signal. Backscatter techniques use a single ultrasonic probe which allows access to more skeletal locations. By leveraging the unwrapped phase information extracted from the Hilbert transform, the slope of the phase accumulation was measured as a function of time. The slope correlated strongly ($R > 0.9$) with the density of specimens of human cancellous bone measured *in vitro* with a 3.5 MHz transducer. These results suggest that phase accumulation measurements obtained from the Hilbert transform of backscatter signals may detect osteoporotic changes in cancellous bone.

8:20

5aBA3. Quantitative ultrasound-based characterization of placental microstructure during preeclampsia. Andrew Markel (Biomedical Eng., Tulane Univ., 6823 St. Charles Ave., New Orleans, LA 70118, amarkel@tulane.edu), Cameron Hoerig (Radiology, Weill Cornell Med., New York, NY), Allan Alencar, Kenneth Swan (Tulane Univ., New Orleans, LA), Alexander Gleed (Radiology, Weill Cornell Med., New York, NY), Lili Shi (Biomedical Eng., Tulane Univ., New Orleans, LA), Jonathan Mamou (Radiology, Weill Cornell Med., New York, NY), and Carolyn Bayer (Biomedical Eng., Tulane Univ., New Orleans, LA)

Preeclampsia is a life-threatening pregnancy disorder resulting from improper development of the placenta. B-mode ultrasound imaging is the ubiquitous tool for pregnancy imaging but does not provide enough contrast to detect changes in placenta microstructure during preeclampsia. Quantitative ultrasound (QUS) methods have been used to assess tissue microstructure in various organs, including the placenta. Our goal was to demonstrate how QUS parameter maps of the placenta correlate with biological features in the placenta's microstructure. Ultrasound image data of 9 pregnant Sprague-Dawley rats were collected on gestational day 18 using the Vevo 2100 equipped with the LZ250 transducer (21 MHz center frequency). QUS parameter maps of 4–6 placentas per rat were generated using radiofrequency (RF) data from ultrasound image acquisitions. For each placenta, we generated 9 parameter maps: 5 using backscatter coefficient methods and 4 using envelope statistics. After imaging, placenta tissue samples were embedded in paraffin, sectioned, stained with hematoxylin and eosin (H&E), and scanned at 20x magnification using the Zeiss Slide Scanner. Backscatter coefficient parameters exhibited a linear correlation with cell nucleus diameter across the different regions of the placenta ($R2 = 0.58$ and $p = 3.8e\text{-}6$). Future studies will focus on studying preeclampsia with QUS in human placentas.

8:40

5aBA4. Preliminary results from use of multifrequency ultrasound imaging for quantitative tissue characterization. Joshua Hanson (Texas A&M Univ., 800 Raymond Stotzer Pkwy, TIPS, College Station, TX 77843-4478, jrhanson2301@tamu.edu) and Kenneth Hoyt (Texas A&M Univ., College Station, TX)

The attenuation coefficient (AC) measures the loss of ultrasound (US) signal amplitude as it propagates at depth. In this research, we detail a pixel-based AC estimation method using a new multifrequency US imaging (MFUI) approach. A research scanner equipped with an L11-5v transducer

was used in a plane wave imaging mode. Radiofrequency data at 24 different transmit frequencies that span the range of the transducer bandwidth were sequentially collected. Each acquisition involved 21 angled planes waves in the $\pm 18°$ range. B-mode US images were generated by applying a Hilbert transformation applied to the angled compounded data followed by logarithmic scaling. Local ACs were estimated from the series of multifrequency US images based on knowledge of the relationship between US image intensity, transmit frequency, and depth at each spatial location. Preliminary MFUI tests were conducted using an US phantom having two different regions with ACs of 0.70 or 0.95 dB/cm/MHz. AC images were summarized as mean $\pm$ standard deviation. Average MFUI-derived ACs from the phantom material with known values of 0.70 and 0.95 dB/cm/MHz were estimated to be 0.70 ± 0.007 and 0.95 ± 0.002 dB/cm/MHz, respectively. AC estimates were within 0.04 dB/cm/MHz of the true value. Preliminary results indicate MFUI is a promising approach for local estimation of AC values.

9:00–9:20 Break

9:20

5aBA5. Motion correction for improved ultrasound measurement of hyoid bone movements during swallowing. Nicholas S. Schoenleb (Biomedical Eng., Univ. of Cincinnati, 3159 Eden Ave., Cincinnati, OH 45219, schoenna@mail.uc.edu), Maxim V. Lushpin (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Brittany N. Krekeler, Anna K. Hopkins (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Rebecca J. Howell (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Ultrasound imaging is able to track oropharyngeal structures used in swallowing, e.g., the hyoid bone. Studies using simultaneous B-mode and x-ray videofluoroscopic imaging have shown that accuracy of ultrasound-measured hyoid movements is highly dependent on motion of the ultrasound array in relation to vocal tract anatomy. Here, we address this challenge by correcting for array motion, measured either by videofluoroscopy or by an inertial measurement unit (IMU) integrated into a handheld, wireless scanner (Clarius C7 HD3). For videofluoroscopy, a Harris corner tracker is used in CASM (Computational Analysis of Swallowing Mechanics) MATLAB-based software for semi-automatic tracking of reference points on the array. For IMU measurements, time-dependent 3-axis acceleration and angular velocity measurements are smoothed and integrated to obtain time-dependent array displacement and angle. The measured hyoid trajectory is then corrected using coordinate transforms accounting for the array's time-dependent translation and in-plane rotation. Hyoid trajectories and velocities, measured using CASM from synchronized B-mode and standard-of-care fluoroscopy recordings, are compared using intraclass correlation coefficients and root-mean-square error between the two modalities, with and without correction for array motion. Results indicate significant improvement in measurement accuracy when correcting for array motion, while also characterizing the magnitude of measurement errors for uncorrected handheld ultrasound imaging.

9:40

5aBA6. A combined U-Net and transformer approach for simultaneous segmentation and beamforming in plane wave ultrasound imaging. Bo Wen (Elec. and Comput. Eng., Univ. of Rochester, 220 East Squire Dr., Apt 8, Rochester, NY 14623, bwen3@ur.rochester.edu) and Marvin Doyley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Machine learning has revolutionized ultrasound imaging by addressing key limitations in plane-wave imaging. This paper presents a novel self-supervised hybrid model combining U-Net and transformer architectures to perform beamforming and segmentation using single-angle plane-wave channel data simultaneously. The model outputs images comparable to those produced by 33-angle compounded plane-wave imaging while retaining the high frame rates of single-angle acquisition. Extensive evaluations on simulated and experimental phantom datasets with hypoechoic inclusions (5–11 mm diameter) demonstrated the model's ability to produce high-quality, clutter-free images with sharp boundaries and accurate segmentations. The average DICE similarity coefficient of the segmented inclusions was 0.97, reflecting substantial overlap with ground truth masks. Performance metrics, including Laplacian variance, mean squared error (MSE), signal-to-noise ratio (SNR), and generalized contrast-to-noise ratio (gCNR), showed that the hybrid model effectively suppressed side-lobes and preserved inclusion details. This study demonstrates that integrating beamforming and segmentation into a unified framework can address trade-offs in current techniques, paving the way for more efficient, real-time ultrasound imaging systems with superior diagnostic capabilities.

10:00

5aBA7. Reconstruction of volumes from imperfectly known transducer location for low-cost ultrasonic imaging. Jeremy Dahl (Radiology, Stanford Univ., 3155 Porter Dr., MC 5483, Palo Alto, CA 94304, jjdahl@stanford.edu), Brian Boitnott (Biomedical Phys., Stanford Univ., Stanford, CA), and Raghu Raghavan (Sonovance, Inc., Baltimore, MD)

Reconstructing a three-dimensional ultrasound volume from freehand planar scans for low-cost applications requires the acquired planar data be placed in three-dimensional space. Placement is based on the reported position and orientation of the transducer from 6- or 9-degree of freedom sensors. While the literature reports adequate resolution and accuracy for reconstruction from expensive electromagnetic sensors, low-cost inertial sensors typically have poor position estimates due to drift in accelerometer readings. We formulate this problem as one of estimation of the parameters of position and orientation for ultrasound signals in the complex signal domain, wherein the "new" pose of the transducer is a Euclidean transform of the "prior" pose. It can be shown that this transform can be estimated from calibrated correlation functions from one cartesian translation and two rotations in combination with the low-cost sensor. Ideal cartesian translation of a Verasonics L12-3v transducer were used to validate the proposed method and demonstrated mean errors of -4.7 and 6.0 microns for translations of λ and 2λ with respective standard deviations of ± 10.3 and ± 20.2 microns and error ranges (min,max) of 33.7 (-22.8,10.9) and 76.9 (-31.1,45.8) microns. We further demonstrate the validity of the approach in simulations and experiments in tissue mimicking phantoms.

10:20

5aBA8. Ultrasonic characterization of the transmural structure of human scalp tissue. Catherine N. Prabish (Phys., Rhodes College, 1800 North Parkway, Memphis, TN 38112, pracn-27@rhodes.edu), Blake C. Lawler, Thomas Conroy (Phys., Rhodes College, Memphis, TN), Cecille Labuda (Phys. and Astronomy, Univ. of Mississippi, University, MS), and Brent K. Hoffmeister (Phys., Rhodes College, Memphis, TN)

Interest in transcranial ultrasound has motivated numerous ultrasonic studies of the skull and brain; however, the ultrasonic properties of the scalp are relatively unknown. The goal of this study was to ultrasonically characterize the transmural structure of scalp tissue. Sixty-four formalin fixed specimens were prepared from four human donors and scanned in a water tank with a 25 MHz transducer to create parametric images of the speed of sound (SOS), frequency slope of attenuation (FSA), and integrated attenuation coefficient (IAC). Images revealed three distinct layers: a dermis/epidermis layer, a subcutaneous layer, and a connective tissue layer. A statistically significant difference was observed between all layers for all parameters in most cases. Exceptions were between the dermis/epidermis layer and the subcutaneous layer for FSA and between the subcutaneous layer and the connective tissue layer for SOS.

10:40

5aBA9. Combined speed of sound correction and coherence masking improves ultrasound imaging of renal cysts. Scott J. Schoen (Radiology, Harvard Med. School and Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, sschoenjr@mgh.harvard.edu), Theodore Pierce, Marko Jakovljevic, Sai Dhanush Reddy Jeggari, David Hunt, Kathleen Pope (Radiology, Massachusetts General Hospital, Boston, MA), Brian Lause, Rimon Tadross, Michael Wang, Mike Washburn (GE HealthCare, Waukesha, WI), and Anthony E. Samir (Radiology, Harvard Med. School and Massachusetts General Hospital, Boston, MA)

Renal cystic lesions are common, with approximately 25% of individuals over 40 and 50% over 50 having at least one cyst, the majority of

which are benign. However, some lesions can become malignant, necessitating regular imaging to distinguish between benign and malignant cysts. Ultrasound (US) is a preferred imaging modality due to its safety and cost-effectiveness, but its efficacy is often limited in obese patients due to poor image quality resulting from fat attenuation and phase aberration. This can lead to misdiagnoses, as smaller features indicative of pathology can be obscured or made ambiguous. In this study, we employ a two-stage approach to enhance *in vivo* US images of renal cysts. First, a bulk speed of sound correction is applied to optimize resolution in the delay-and-sum images. Next, we implement a spatial correction based on short-lag spatial coherence of the channel data, computed for the identified optimal speed of sound. Among all subjects ($N = 10$) with body mass indices ranging from 18 to 38, cystic boundaries were sharpened, and the generalized contrast-to-noise ratio (gCNR) improved by 7.4% [CI (5.5, 9.2)%, $p < 0.0001$] compared to uncorrected delay-and-sum images. Furthermore, the weighting of the coherence mask can be varied in real time during evaluation to enhance the diagnostic value of the composite image. This approach demonstrates significant potential for rapid clinical translation and for augmenting the critical role of US in monitoring kidney cysts.

FRIDAY MORNING, 23 MAY 2025 BISSONET/CARONDELET, 11:10 A.M. TO 12:00 NOON

Session 5aID

Interdisciplinary: Plenary Lecture: Acoustic Ecology of Marine Mammals in the Era of Blue Economy: Navigating Development and Ocean Noise Challenges

Jennifer Miksis-Olds, Chair
Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824

Chair's Introduction—11:10

Invited Paper

11:15

5aID1. Acoustic ecology of marine mammals in the era of blue economy: Navigating development and ocean noise challenges. Christine Erbe (Ctr. for Marine Sci. and Technol., Curtin Univ., B301, Kent St., Perth, Western Australia WA 6102, Australia, C.Erbe@curtin.edu.au)

Industrialisation and ocean economy development have led to a steady increase in underwater noise across the world's oceans. Concerns about marine noise, research on the effects of noise on ocean ecosystems, engineering solutions to minimise noise and mitigate effects, and policies to manage noise have been developed as corollaries. Given how well sound travels under water, compared to other cues, such as light and chemicals, we assume most animals in the ocean have evolved to rely on sound to support key life functions, including environmental sensing and communication. However, the actual information we have on marine fauna hearing, sound usage, and noise impacts is limited to perhaps a hundred species, with marine mammals being amongst the best studied. Bioacoustic research outputs do not seem to be able to keep up with the pace of offshore development (e.g., subsea mining, subsea processing)—leaving us in a situation where we must manage underwater noise in a great absence of data. Let us review what we do know, what we do not know, and what we need to know. Is the Blue Economy as "green" as we want it to be?

Session 5aMU

Musical Acoustics: General Topics in Musical Acoustics III

Nicholas Giordano, Cochair
Phys., College of Sci. and Math., Auburn Univ., Auburn, AL 36849

Pablo L. Rendón, Cochair
Instituto de Ciencias Aplicadas y Tecnología, Universidad Nacional Autónoma de México, Ciudad Universitaria, México City, 04510, Mexico

Contributed Papers

8:40

5aMU1. End corrections at both ends of a flue instrument. Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu) and Katherine L. Saenger (Phys., Auburn Univ., Ossining, NY)

We describe a study of end corrections for a soprano recorder sounding its lowest note (C5). Using both experiments and simulations, we have determined how the oscillation amplitude for the density standing wave in the interior of the instrument varies with position along the flue pipe. At the so-called open end, i.e., the end farthest from the mouthpiece, this amplitude extrapolates to zero (i.e., a node), a short distance beyond the physical end of the pipe, by a distance that agrees well with the theory of Levine and Schwinger [1] and subsequent work. This end correction is found to be independent of blowing pressure. At the blown end, i.e., the end of the instrument where an air jet emerges from the channel, the amplitude of the pressure standing wave is much larger than at the open end. For the lowest blowing pressures studied, for which the note is flat by about a semitone, this pressure amplitude is about 40% of the peak amplitude found near the center of the pipe. Some implications of this relatively large pressure amplitude at the blown end and how it varies with blowing pressure are discussed. H. Levine and J. Schwinger, Phys. Rev. 73 (4), 383–406 (1948). [Work supported by U.S. National Science Foundation (Grant No. PHY-2306035)]

9:00

5aMU2. The influence of side hole undercutting on vorticity and frequency content in wind instruments. Titas Lasickas (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria, lasickas@mdw.ac.at), Giuseppe C. Caridi, Alfredo Soldati (The Vienna Univ. of Technol., Vienna, Austria), and Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Vienna, Austria)

In this work, we conducted an experimental campaign to observe the oscillating flow velocity spectra and the vorticity inside a wind instrument. Time-resolved particle image velocimetry (PIV) measures the internal flow velocity within the transparent bore of a square bass recorder. Although vortex formation at the side holes of wind instruments has been identified in numerical simulations, limited experimental investigations have been carried out. While frequency content analysis is an unexplored approach to investigate the interior flow of a wind instrument with side holes, it has the potential to provide new insights. Flow velocity measurements at a 3.5 kHz sampling rate resolve at least five harmonics of the acoustic flow and reveal vortex formation. Measurements across the full length of the bore reveal vertical standing waves towards the open end. Concurrent air pressure measurements confirm the locations of flow velocity frequency peaks. The experiments were conducted under realistic input conditions, where a commercial recorder mouthpiece and a constant pressure source were used for excitation. We compare flow velocity oscillation spectra for varying side hole geometries. Undercutting was observed to concentrate more energy at the fundamental frequency and to reduce vortex formation at the side hole.

9:20

5aMU3. Modeling the sounding mechanism of flute-like instruments based on the measured jet response. Seiji Adachi (Tianjin Univ., No. 92 Weijin Rd., Nankai District, Tianjin 300072, China, seiji_adachi@yahoo.co.jp) and Zhiwen Qian (Tianjin Univ., Tianjin, China)

Various sound production models for flute-like instruments have been proposed in the past. Most of them are based on the jet oscillation model proposed by Fletcher *et al*. Based on this model, Verge *et al*. constructed a model that can reproduce the correct sound pressure level, which is now considered to be the standard model. Price *et al*. measured the reflection function of a recorder's head, which describes the response of the jet to sound. To reproduce the reflection function using Verge's model, it was necessary to introduce an unnatural gain adjustment factor. Recently, a model allowing the jet to deflect overall has been proposed and shown to reproduce the reflection function correctly. It was, however, found that the sound pressure level was underestimated in time-domain simulation. In this study, instead of starting from Fletcher's model, a jet oscillation model was directly constructed from Price's measured reflection function using spline curve approximation. When a time-domain simulation was performed using this model, it was found that the sound pressure level and mode transition diagram of a recorder were reproduced very similarly to those observed in an artificial blowing experiment.

9:40–10:00 Break

10:00

5aMU4. Exploring the acoustic color signature paterns of Bansuri, the traditional Indian bamboo flute using principles of the Helmholtz generator and geometric signal processing techniques. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Trevor Smith (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Panchajanya Dey (Kolkata, West Bengal, India)

The traditional Indian bamboo flute, commonly known as the Bansuri, has rich harmonic features in the acoustic color domain and has been part of South Asian classical, indigeneous, and folks traditions for as long as recorded history. In this talk, I will explore the Bansuri as a composite Helmhotz resonator and present my initial investigations on how some of the physics of the Helmholz resonator create the rich melodic sound of the Indian Bansuri. In the signal processing and data science domain, we will also present rich acoustic color features generated by traditional melodic movements in North indian classical music. In particular, the talk will cover computational techniques designed to separate the modes of acoustic propagation and sound production in the Bansuri, e.g., filtering out leakage due to

mixing of modes. If time permits, I will also present how the geometric aspects of the acoustic color features may be exploited to create a fluid feature dictionary that may be harnessed with large language models (LLM)-like architecture to enable machine interpretation of Indian. classical music traditions.

10:20

5aMU5. Fluid dynamic harmonica bending model. Kuiliang Li (Tufts Univ., 640 Boston Ave., Unit 509, Medford, MA 02155-1330, likuiliang123456@gmail.com) and Chris Rogers (Tufts Univ., Medford, MA)

Bending is a phenomenon commonly used in the playing of diatonic harmonicas traditionally by changing the player's mouth shape to lower the pitch of a specific note. It is always regarded as a technique specific to the harmonica family instead of being applicable to the entire free-reed instrument family. The cause has been studied and widely recognized as being the effect of acoustic feedback as well as the resonance of the double reed system inside a single hole of a harmonica. In recent years, the manufacturing technique of chromatic harmonica has improved enough that single reed bending has become more and more consistent. Without the second reed in resonance, the cause of single-reed bending has become complicated. We have proposed a work-in-progress model of reed activation and pitch bending based on our experimental data of pressure and velocity measurements on a single-reed 1:1 model. The sources of control in the experiments consist of both human breath and a fixed volumetric flow rate air inlet. We will also present particle image velocimetry measurements around a reed.

10:40

5aMU6. Audio performance comparison and analysis across double reed materials, performance spaces, articulation and rhythmic variability, and melodic scales. Lizette M. Wong (Elec. Eng. & Music, Univ. of Texas at Austin, 2501 Speedway, Austin, TX 78712, lizettewong@utexas.edu) and Zoelle F. Wong (Aerosp. Eng., Georgia Inst. of Technol., Atlanta, GA)

Bassoon reeds, commonly constructed with *Arundo donax*, frequently contain inconsistent audio and structural performance due to their natural degradation from saliva and humidity exposure. Though a necessary tool for sound production, reeds are reported to be an expensive recurring cost which creates a financial obstacle for musicians. While nonorganic solutions such as polypropylene reeds are commercially available, they have demonstrated physical and acoustic inconsistencies. In a previous study, the audio performance of a bio-composite reed design, Fermata, was investigated as an effort to introduce alternative reed options. In this study, spectrograms and FFT plots were constructed after recording pure tone pitches and melodic lines. Comparisons and audio analyses were made across *Arundo donax*, synthetic, and Fermata reeds relative to performance spaces, articulation and rhythmic variability, and melodic scales. Current work indicates that reed selection and performance spaces impact the timbral coefficients despite sounding similar to the human ear.

FRIDAY MORNING, 23 MAY 2025 GALERIE 5, 9:00 A.M. TO 10:20 A.M.

Session 5aNS

Noise: General Topics in Noise: Measurement and Processing I

S. Hales Swift, Chair
Sandia National Lab., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082

Invited Paper

9:00

5aNS1. Enhanced noise measurement and analysis in distributed acoustic sensing systems. Gizem Karslioglu (SAGE, Earthscope Consortium, 505 Bullock Ave. Socorro, NM 87801, gizem.karslioglu@earthscope.org)

A Distributed Acoustic Sensing (DAS) unit is widely used for seismic, volcanic, and health monitoring activities. Most DAS systems are based on Phase-Optical Time Domain Reflectometry (φ-OTDR) which offers high sensitivity by detecting phase changes in Rayleigh backscattered light along the optical fiber cable. The performance and accuracy of DAS systems can be affected by various noise types, which depend on system components and environmental conditions. Therefore, noise measurement and analysis are essential both prior to and during data acquisition to ensure optimal system functionality. One technique to evaluate the performance of a DAS system is the Spatial Noise Resolution test, which provides critical insights into system precision and reliability. The Spatial Noise Resolution test in DAS systems is essential for evaluating the system's ability to differentiate between spatially distinct signals in the presence of noise. The system's precision in detecting events along the fiber is directly correlated with the Signal-to-Noise Ratio (SNR). Following this, filtering techniques are applied to reduce noise, enhancing the precision and reliability of the detected signals and improving overall performance of the DAS system.

9:20

5aNS2. Localization and sound power estimation of noise sources in industrial plants using coherence scanning holography. Thomas Rittenschober (Seven Bel GmbH, Hafenstrasse 47-51, Linz 4020, Austria, thomas.rittenschober@seven-bel.com) and Antoine Decloux (Seven Bel GmbH, Linz, Austria)

Reducing noise emissions from industrial plants requires an initial step of localizing key contributing sources. Acoustic cameras are pivotal for quickly identifying these sources, yet challenges persist in low-frequency source localization and presenting results for effective impact assessment. This contribution introduces Coherence Scanning Holography—a method based on scanning the sound field by means of a rotating linear microphone array covering a measurement surface with diameter of 2.5 m for low-frequency source localization from 125 Hz. The underlying sound imaging method compensates for Doppler distortions in the moving microphone signals and evaluates coherence spectra with a non-moving reference microphone to compute an acoustic image. For analysis, a method quantifying sound power contributions from specific regions in the optical image is presented. Comparing sound power metrics of the full optical field of view with metrics of regions of interest enables users to quickly rank source contributions and assess the impact of reducing these contributions on total sound power. In complex scenarios, preliminary investigations can be streamlined with a few measurements from diverse angles, expediting the problem-solving process with confidence. The method is exemplified through the analysis of sound emissions from a production plant.

9:40

5aNS3. FPGA implementation of a GHKSS filter for real-time noise reduction in nonlinear time series data. Marcin Konczyk (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Sydney, New South Wales, Australia) and Sebastian Oberst (School of Mech. and Mechatronic Eng., Ctr. for Audio, Acoust. and Vib., Faculty of Eng. and IT, Univ. of Technol. Sydney, Lord St 32-34, Sydney, New South Wales 2019, Australia, sebastian.oberst@uts.edu.au)

The GHKSS nonlinear projective filtering method as a representative of nonlinear time series analysis, has been used as a "posterior filtering" noise reduction technique in a range of applications. Compared to Fourier and model-based techniques it has the advantage of reducing the noise with broadband spectra, without knowledge of the system model. The contamination of time series data with measurement noise may result in an apparent loss of intrinsic dynamical information of the nonlinear systems. A real-time implementation and working principle of the GHKSS algorithm is widely ignored due to its complexity and high computational cost. Here, we present a real-time implementation of this algorithm using a Field Programmable Gate Array (FPGA) and High-level synthesis (HLS) tools. The computational burden is reduced by running the false nearest neighbour (FNN) search and delay estimation part of the GHKSS, on a separate DSP slice using HLS tools. The effect of noise reduction on dynamic preservation using GHKSS was illustrated by estimating the correlation dimension and the maximal Lyapunov exponent of the 3-D Chay model in the chaotic regime. This research is important since the real-time implementation of the GHKSS algorithm will widen its application in nonlinear time series analysis.

10:00

5aNS4. Sound directionality characterization in a digital MEMS microphone-based acoustic monitoring prototype. William D. Fonseca (Acoust. Eng., Federal Univ. of Santa Maria, Av. Nossa Sra das Dores, 305, ap503B, Santa Maria, Rio Grande do Sul 97050-531, Brazil, will.fonseca@eac.ufsm.br) and Felipe R. Mello (Acoust. Eng., Federal Univ. of Santa Maria, Santa Maria, Rio Grande do Sul, Brazil)

This paper presents the characterization of the sound directionality of an acoustic monitoring prototype developed with digital MEMS (Micro-Electro-Mechanical Systems) microphones. The objective is to evaluate the device's ability to record sounds originating from different angles and frequencies, as well as to verify its efficiency in relation to a free-field reference microphone. To this end, the instrumentation and measurement chain adopted, from component selection to calibration and data acquisition procedures are described. In the experimental stage, various field tests were conducted, encompassing frequency and incidence angle variations, which enabled the generation of polar plots. These plots highlight the prototype's directional response and the diffraction effects by frequency, thereby allowing a detailed analysis of its acoustic performance. Comparative results with the reference microphone demonstrate the suitability of the proposed prototype for noise monitoring and acoustic mapping applications. In summary, the solution described in this work provides a cost-effective alternative for sound signal capture, contributing to the advancement of monitoring systems based on MEMS microphones.

Session 5aPAa

Physical Acoustics, Underwater Acoustics and Acoustical Oceanography: Meteorological Acoustics

Roger M. Waxler, Cochair
Univ. of Mississippi, P.O. Box 1848, University, MS 38677

Natalia Sidorovskaia, Cochair
Physics, Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504-4210

Invited Papers

8:00

5aPAa1. Wind, rain, and breaking waves in tropical cycles observed by underwater ambient sound. Zhongxiang Zhao (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, zzhao@uw.edu) and Eric D'Asaro (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The air–sea interface beneath tropical cyclones is studied using underwater ambient sound measured by hydrophones onboard air-deployed Lagrangian floats. We obtained ~200 h of broadband sound data at 1–50-m depth with wind speed 15–45 m/s. We detected acoustic signatures of wind, rain, and breaking waves by decomposing the sound data into three components according to their timescales. There is a complex dependence of sound level on wind speed due to the competing effects of sound generation by breaking waves and attenuation by quiescent bubbles. Sound level increases monotonically with increasing wind speed for low frequencies <200 Hz. At frequencies >200 Hz, sound level first increases and then decreases with increasing wind speed. Rain-generated sound fluctuations have spectra around 10 kHz. At high speeds, rain- and wind-generated sound levels are comparable. Our rain detection algorithm is based on 5–30-kHz sound with periods >20 s and <30 min, and the detected rain events usually last for a few minutes. Our fast sampling rate (two samples per second) revealed sound bursts caused by breaking waves, which usually last for 6–8 s and do not change much with wind speed. At low winds, the sound bursts cover frequencies from 50 kHz down to 200 Hz. At high winds, the sound bursts are disrupted around 1 kHz, with asynchronous high- and low-frequency sound bursts.

8:20

5aPAa2. Weather observations from deepwater soundscapes in the Northern Gulf of Mexico. Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504-4210, natalia.sidorovskaia@louisiana.edu), Roger M. Waxler (Univ. of Mississippi, University, MS), Claus Hetzer (National Ctr. for Physical Acoust., Tempe, AZ), and Bin Liang (National Ctr. for Physical Acoust., Oxford, MS)

Wind Observations Through Ambient Noise (WOTAN) technique was first developed in the 1980s to address the limitations of low temporal-resolution satellite data and sparse ship- or surface-buoy-based measurements that can be severely impacted by bad weather. The recent integration of passive acoustic monitoring systems on autonomous platforms (e.g., gliders) has reinvigorated the application of WOTAN for high-resolution, near-surface wind and wave forecasts. This presentation will highlight the opportunities, challenges, and validation efforts associated with using WOTAN in the Northern Gulf of Mexico, a region significantly affected by industrial activities and hurricanes. Raw or processed acoustic data can be assimilated into operational weather-forecasting models (those used by NOAA and the U.S. Navy) to refine cyclone intensification and overall marine weather predictions. These advancements provide the potential of passive acoustic methods to enhance meteorological modeling and advance our understanding of the physics of air-sea interactions in dynamic coastal ocean environments. [Work supported by the Office of the Under Secretary of Defense for Research and Engineering (Award No. FA9550-21-1-0215)]

8:40

5aPAa3. Bayesian inversion for hurricane parameters based on underwater sound levels. Bin Liang (National Ctr. for Phys. Acoust., Oxford, MS), Roger M. Waxler (Univ. of Mississippi, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu), and Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

Hurricanes classification has been a critical task for meteorologists working to mitigate loss of life and property. *In situ* observations using aircrafts and remote observations based on radar and satellite are commonly used. We report on investigations of an acoustic method for hurricane classification using measured underwater sound levels. The relation between the sound level and the surface wind speed is approximately linear, within certain ranges. The slope of the linear relation is believed to be universal. The Holland model for hurricane surface winds is a commonly used axisymmetric approximation. The method investigated inverts for the parameters of the Holland model using sparse measurements of the sound levels under a hurricane. The method is validated first using the synthetic data and then applied to field measurements. It is shown that the maximum wind speed and radius of maximum wind speed in axisymmetric hurricane models are estimated accurately using Bayesian inversion.

9:00

5aPAa4. Comparison of infrasound array data to NEXRAD weather radar for tornadic and non-tornadic storms in the southeastern United States. William E. Audette (Benchtop Eng. LLC, 281 Sweet Pond Rd., Guilford, VT 05301, chipaudette@benchtopeng.com), Roger M. Waxler (NCPA, Univ. of Mississippi, University, MS), Garth Frazier, Carrick Talmadge, Bin Liang (NCPA, Univ. of Mississippi, Oxford, MS), Claus Hetzer (NCPA, Univ. of Mississippi, Tempe, AZ), and Hank Buchanan (NCPA, Univ. of Mississippi, Oxford, MS)

Long-duration infrasound recordings were performed using several arrays deployed in the southeastern United States during the Spring of 2018. This data has been analyzed via several beamforming methods to detect and locate sources related to severe weather, including tornadic storms. In parallel with this analysis, we also collected weather-related data products from National Weather Service, including their database of severe weather events as well as NEXRAD radar maps covering the entire time period. In this paper, we compare the weather-related data to infrasound activity through time. As previously reported, we found infrasound tracks that are clearly aligned with NWS-identified tornados. Through this new, extended comparison spanning two months of recordings at each site, we have also found that some non-tornadic storms also generate infrasound tracks. The data show that some strong (but non-tornadic) storms generate clear infrasound tracks while others do not. This finding continues to motivate additional research into the relationship between storm structure and infrasound emissions.

9:20–9:40 Break

Contributed Papers

9:40

5aPAa5. Predictions for acoustic sensing in Saturn's atmosphere. Andrew Powell (Dept. of Phys., Univ. of Louisiana at Lafayette, 240 Hebrard Blvd., Lafayette, LA 70503, andrew.powell1@louisiana.edu), Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), Rishabh Chaudhary, Robert D. White (Mech. Eng., Tufts Univ., Medford, MA), Don Banfield (NASA Ames Res. Ctr., Mountain View, CA), and Ian Neeson (VN Instruments, LTD, Elizabethtown, ON, Canada)

We investigate using ultrasonic sensing to measure the helium and ortho/para-hydrogen ratios in Saturn's atmosphere, in the context of NASA's New Frontiers Saturn Probe mission concept. We use a theoretical wavenumber model incorporating hydrogen rotational relaxation into a linearized fluid dynamics framework. We assume a windless and cloudless hydrogen-helium environment. Quantitative gas analysis can be achieved using both the frequency dependent sound speed and attenuation coefficient. We present the sensitivity of the two acoustic properties to ortho-H_2/para-H_2/He ratios at temperatures and pressures typical of Saturn. H_2-H_2 and He-H_2 collisions both affect the hydrogen relaxation times. Ortho/para-H_2 ratio changes can be detected from the overall sound speed and attenuation changes and from the corresponding shifts in the characteristic H_2 relaxation frequencies (e.g., $\sim$1 MHz at 100 mbar, $\sim$10 MHz at 1 bar). Helium abundance should be detectable primarily through the molecular weight difference. The results are encouraging, showing that ultrasonic sensors operating during descent in Saturn's atmosphere from 100 mbar to 10 bar could be able to detect the He content and discriminate between ortho- and para-H_2. [Funding was provided by NASA-CIF (Grant No. 80NSSC22M0106)]

10:00

5aPAa6. Determining the potential temperature for real gases: Application to Venus's lower atmosphere. Gil Averbuch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA) and Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, 240 Hebrard Blvd., Broussard Hall, Rm. 103, Lafayette, LA 70503-2067, andi.petculescu@louisiana.edu)

The potential temperature is an important meteorological parameter quantifying the response of an atmosphere to dynamic perturbations. It is defined as the temperature that a gas parcel would have if it were transported adiabatically from the local pressure to a reference pressure. It is obtained by invoking the First Law of thermodynamics for enthalpy in adiabatic conditions. For ideal gases, the infinitesimal fractional changes of pressure and temperature are coupled simply through the constant isobaric specific heat, making the integration from the reference to the local point trivial. For real gases, however, the coupling between pressure and temperature fluctuations is no longer constant, depending instead on the local pressure, density, and temperature. Therefore, the potential temperature can be obtained either by approximation or numerically. We present the framework for incorporating the Peng-Robinson real-gas equation of state in setting up the adiabatic pressure-temperature differential equation. Then, we discuss its solutions in terms of the compressibility factor and their implications to the stability and buoyancy frequency of the Venusian atmosphere using ambient data acquired during the descent of the Vega-2 probe.

10:20

5aPAa7. Characterizing the impact of turbulence on an acoustic camera. Michele L. Eggleston (Phys. and Astronomy, Brigham Young Univ., Brigham Young University, Provo, UT 84604, megglest@byu.edu), Michael B. Muhlestein, Matthew J. Kamrath (US Army Engineer Res. and Development Ctr., Hanover, NH), D. Keith Wilson, Vladimir E. Ostashev, Sergey N. Vecherin (US Army Engineer Res. and Development Ctr., Hanover, NH), and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT)

This presentation investigates the impact of atmospheric turbulence on sound propagation using a beamforming array. Experimental measurements were conducted at the North Hartland Dam in Hartland, VT with a Nor848B acoustic camera and a dual-diaphragm compression driver serving as the sound source playing simultaneous frequencies at 225, 450, 900, 1800, and 3600 Hz. The source and acoustic camera were positioned approximately 100 m apart, with the source approximately elevated 55 m and the camera 1.5 m above ground level. A weather station, situated 2 m above the ground, was placed along the propagation path to monitor atmospheric conditions. Measurements were carried out at various times of the day to capture variability in different weather conditions. Observations using the acoustic camera revealed significant acoustic image wander and scintillation. This paper presents a statistical analysis of the acoustic image wander over time, and correlations between these variations and weather conditions.

10:40

5aPAa8. Analysis of urban microclimate acoustic tomographic heatmaps. Claudia Bahamon Lopez (Eng., Univ. of the District of Columbia, 4200 Connecticut Ave NW, Washington, DC 20008, claudia.bahamonlopez@udc.edu), Lirane Mandjoupa, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, acoustic tomography is employed as a technique for monitoring urban microclimate variables and mapping temperature distributions in urban street canyons, focusing on identifying and analyzing Urban Heat Island (UHI) hot spots. The microclimate temperature, wind profiles, and urban morphology are exciting, as they influence intra-urban characteristics and contribute to UHI effects. The acoustic tomographic heatmap provides street-level resolution and detailed insights into spatial temperature deviations, including minor acoustic time delays associated with faster sound speeds and higher temperatures, characteristic of UHI hot spots within urban street canyons. Preliminary results of UHI hot spots in Washington, D.C., during a heat wave will be discussed.

Session 5aPAb

Physical Acoustics: Physical Acoustics Best Student Paper Award Poster Session

Ian C. Bacon, Cochair
Brigham Young University, Provo, UT 84602

Christopher M. Kube, Cochair
Engineering Science and Mechanics, The Pennsylvania State University, 212 Earth and Engineering Sciences Bldg., University Park, PA 16802

All posters will be on display and all authors will be at their posters from 8:00 a.m. to 9:30 a.m.

The ASA Technical Committee on Physical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Physical Acoustics. Additionally, each student will give an oral presentation in a regular/special session.

Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers.

1pPAa10. Particle deflection in a macroscale ultrasonic angled wave device
Student author: Nicholas Rivet

1pPAa11. Modular platform for reconfigurable bulk acoustic wave-based acoustofluidics
Student author: Mohammad Kamali

2aPAb2. Local assessment of viscoelastic properties with acoustical tweezers
Student author: Antoine Penneron

2aPAb3. Radiation force on inhomogeneous subwavelength scatterers due to progressive waves
Student author: Chirag A. Gokani

2aPAb6. Nearfield forces generated using tunable liquid lenses
Student author: Sina Rostami

2pPAb3. Stiffness model and object manipulation in near-field acoustic levitation
Student author: Yaoke Wang

2pPAb7. Metamaterial-assisted acoustic manipulation of large objects in water
Student author: Dajun Zhang

2pPAb8. Non-contact actuation of elastic lattices using Bjerknes forces
Student author: Laurin Sartori

4pPAa2. A survey of signals and noise at high-frequency infrasound arrays in Nevada and South Korea
Student author: Jonathan Yeh Reiter

4pPAa8. Capturing atmospheric dynamics with cross-correlation functions: Insights from infrasound propagation during the April 8, 2024 solar eclipse
Student author: Ketan Singha Roy

4pPAa9. Development of synthetic infrasound signature models from window of opportunity observations
Student author: Aprameya Satish

4pPAb1. Mapping thermal phase transformations in NiTi using immersion ultrasonic testing
Student author: Olivia J. Cook

4pPAb5. Characterization of graphite nodules in nodular cast iron via multimodal ultrasonic scattering measurements using a rosette configuration
Student author: Olivia J. Cook

4pPAb8. Unconsolidated granular materials sound absorption model using sound propagation theories in porous media
Student author: Yousif Badri

5aPAa5. Predictions for acoustic sensing in Saturn's atmosphere
Student author: Andrew Powell

5aPAa7. Characterizing the impact of turbulence on an acoustic camera
Student author: Michele L. Eggleston

5aPAc5. Impact of bended neck on the impedance of Helmholtz resonator
Student author: Jhalu Gorain

5aPAc9. Acoustic levitation for additive manufacturing
Student author: Quintin David Dumouchelle

5pPAa4. Comparison of finite-amplitude acoustic models: Single equation of motion for potential vs system-level approaches
Student author: Mohammad Javad Mohaghegh Kojidi

5pPAb12. Vibration control in rotor chain structures with dual topological states
Student author: Soroush Soltani

FRIDAY MORNING, 23 MAY 2025 STUDIO FOYER, 9:30 A.M. TO 11:00 A.M.

Session 5aPAc

Physical Acoustics: General Topics in Physical Acoustics II (Poster Session)

Teresa J. Ryan, Cochair
Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353

Matthew Stengrim, Cochair
Dept. of Eng., East Carolina Univ., 1000 East Fifth St., Greenville, NC 27858

Joel B. Lonzaga, Cochair
Struct. Acoust. Branch, National Aeronautics and Space Administration, 2 N. Dryden St. (MS 463), Hampton, VA 23681

All posters will be on display and all authors will be at their posters from 9:30 a.m. to 11:00 a.m.

Contributed Papers

5aPAc1. Anisotropy of elastic and photoelastic properties of lithium niobate crystals. Farkhad Akhmedzhanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, 33 Durmon yuli St., Tashkent 100125, Uzbekistan, akhmedzhanov.f@gmail.com)

The elastic and photoelastic constants of lithium niobate crystals and the attenuation coefficients of acoustic waves at room temperature have been determined by the Bragg diffraction of light by acoustic waves. The components of the photoelasticity tensor were determined by the modified Dixon method. The obtained values of the constants were used to determine the anisotropy of the effective elastic and photoelastic constants with a change in the direction of propagation of acoustic waves in the (100) symmetry plane, for different directions of the wave vector and polarization of light relative to the acoustic wave vector. It is shown that, for certain geometries of the Bragg light diffraction, there is a correlation between the orientation dependences of the effective elastic and photoelastic constants in the investigated plane. The results of the study can be used to analyze the anisotropy of the photoelastic properties of lithium niobate crystals and determine the diffraction geometry with the highest efficiency of Bragg light diffraction.

5aPAc2. Experimental study of nanomechanical properties of 2-D transition metal dichalcogenides. Alem Teklu (Phys. and Astron., College of Charleston, 58 Coming St., Charleston, SC 29424, teklua@cofc.edu)

In this study, nanoindentation was used to measure the nanomechanical properties of four two-dimensional transition metal dichalcogenides (TMDCs), namely, Molybdenum Disulfide (MoS_2), Rhenium Disulfide (ReS_2), Rhenium Diselenide ($ReSe_2$), and Tungsten Diselenide (WSe_2). An atomic force microscope (AFM) capable of measuring the nanomechanical properties of these two-dimensional nanomaterials through nanoindentation was used to generate force-distance curves for analysis. From the force-distance curves, reduced Young's modulus, stiffness, and nanohardness of each of these nanomaterials were determined and compared between each other and existing data. The values obtained for reduced Young's modulus of MoS_2, ReS_2, $ReSe_2$, and WSe_2 were 143.93, 78.57, 37.06, and 37.94 GPa, respectively. Among the samples, MoS_2 has the highest values for its reduced Young's modulus, stiffness, and nanohardness followed by, in order, ReS_2, WSe_2, and $ReSe_2$. The values of reduced Young's modulus obtained for these four samples were in agreement with theoretical calculation.

5aPAc3. Characterization of aluminum nitride film properties for high signal-to-noise ratio piezoelectric micro-electromechanical system microphones. Fabio Saba (INRiM, National Inst. of Metrological Res., Strada delle Cacce 91, Torino 10135, Italy, f.saba@inrim.it), Alessandro Schiavi, Diego Pugliese (INRiM, National Inst. of Metrological Res., Torino, Italy), Stefano Stassi (Dept. of Appl. Sci. and Technol., Politecnico di Torino, Torino, Italy), Miriam Cadenas, Sirona Valdueza-Felip, Fernando B. Naranjo (Photonics Eng. Group, Sensors and Photonic Technol. Assoc. Unit, Polytechnic School, Alcalá Univ., Madrid, Spain), Alberto Roncaglia (Inst. for Microelectronics and Microsystems, CNR-IMM, National Res. Council of Italy, Bologna, Italy), Luigi Ribotta, and Giovanni Durando (INRiM, National Inst. of Metrological Res., Torino, Torino, Italy)

The Signal-to-Noise Ratio (SNR) of piezoelectric microphones is significantly dependent on the material properties of the piezoelectric film, namely, the piezoelectric coefficient, the dielectric constant, and the dielectric loss. This work presents a metrological approach for the characterization of Aluminum Nitride (AlN) films used as sensing elements in high SNR piezoelectric Micro-Electromechanical System (MEMS) microphones. A dynamic measurement method based on micro-Laser Doppler Vibrometer (μ-LDV) was adopted to evaluate the piezoelectric d_{33} coefficient of 500 nm thick AlN films deposited by reactive magnetron sputtering over conductive cantilever test samples. The μ-LDV measurement results were also compared against the d_{33} value determined by a Piezo Evaluation System coupled to a single point laser vibrometer, exploiting the converse piezoelectric effect. The accurate and precise evaluation of the piezoelectric properties, together with the proper mechanical design of the MEMS microphone, is fundamental to provide reliable estimations of the electroacoustic performances, in terms of SNR, dynamic range, and frequency response. Furthermore, a metrological approach for the evaluation of the measurement uncertainty of the piezoelectric coefficient allows predicting its contribution to the uncertainty associated with the electroacoustic characteristics of the MEMS microphone.

5aPAc4. Ultrasonic backscatter measurements using phased array ultrasonic transducers. Geoffrey R. Soneson (Mech. and Mater. Eng., Univ. of Nebraska - Lincoln, 2320 Y St., Apt. 1, Lincoln, NE 68503, gsoneson2@unl.edu), Nathanial Matz, and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska - Lincoln, Lincoln, NE)

Diffuse ultrasonic backscatter techniques are used to characterize microstructures and work well when grain scattering is confined within the single-scattering regime. When effects from higher-order scattering are present within the measurements, grain sizes are often overestimated. These effects have been primarily examined with single-element transducers, but phased array ultrasonic transducers (PAUTs) offer several measurement advantages, particularly for samples with complex geometry. In this presentation, the influence of higher-order scattering on PAUT experiments is discussed for samples of weakly scattering aluminum and strongly scattering steel. Spatial variance measurements from PAUT signals show differences in scattering content that are dependent on focal depth and material scattering strength when compared with experiments using single-element transducers. We hypothesize that the multiple transducer elements within PAUTs increase the probability of double or multiple scattering effects within backscatter measurements. These effects are dependent on the material single-crystal anisotropy, the grain size, frequency, experimental configuration, and specifics of the PAUT. Models to describe the PAUT experimental results are also discussed. Accurate theoretical models are critical for validation in order to provide insight into the limits of such measurements for microstructure quantification.

5aPAc5. Impact of bended neck on the impedance of Helmholtz resonator. Jhalu Gorain (ERS, HCLTech, Chennai SEZ SDB-5, Chennai, Tamil Nadu 600119, India, jhalu.gorain@hcltech.com)

Helmholtz resonator-based acoustic blankets are widely used in the industry. The blanket's effectiveness depends upon the bandwidth and peak amplitude of the absorption curve, which, in turn, is dictated by the resonator's impedance. The existing impedance model of a Helmholtz resonator estimates reactance and resistance quite well in the case of a straight neck. However, in the case of the neck with one or multiple bends, the same model shows a deviation in estimating the resistance part. Several test cases have been investigated using the FEA approach to estimate the loss due to bends. Considering a base model of a Helmholtz resonator, bends are introduced and the additional resistance in the impedance has been estimated from the absorption curve. With the increase in bends, the amplitude of the absorption curve reduces and becomes wider. The FEA results have been validated with the experimental results. The investigation shows that a bend in the neck may impact the impedance and hence the absorption coefficient.

5aPAc6. Singular surface analysis and properties of solutions for a theoretical thermoporoacoustic model. Vittorio Zampoli (Univ. of Salerno, Via Giovanni Paolo II, Fisciano, SA 84084, Italy, vzampoli@unisa.it)

Within the framework of theoretical thermoporoacoustics, some recent results are proposed that were obtained by investigating the linearized version of the so-called Eringen-Cattaneo-Christov-Straughan model; it arises from a well-established theory of swelling porous elasticity developed by A.C. Eringen (1994) involving, however, the Cattaneo-Christov approach for the heat flux (B. Straughan, 2020). Wave propagation in a rigid, stationary porous matrix saturated by a perfect gas is studied, allowing for temperature variations; in this regard, the classical Fourier law is abandoned and second sound—i.e., the phenomenon whereby heat travels as a thermal wave—is contemplated. A singular surface-based analysis is performed to examine the evolution of coupled thermal and acoustic pressure wavefronts arising in a signaling problem, that is, considering a Heaviside temperature input source. The Laplace transform and its properties are applied for this purpose, and numerical results are presented to support the analytical findings. In detail, the amplitudes and speeds of the emerging shock waves are derived and analyzed, several special/limiting cases are identified, and the effects of variations in the Eringen coefficient are determined. Finally, for a suitably constructed initial-boundary value problem, some results on the qualitative properties of the solution are provided.

5aPAc7. Metamaterial-enhanced acoustic beamformer for reduced-element arrays. Theodore P. Martin (Precise Systems, 22290 Exploration Dr., Lexington Park, MD 20653, tmartin@goprecise.com), Rachel J. Suitor (Code 5510, U.S. Naval Res. Lab., Washington, DC), Marius D. Pruessner (Code 6930, U.S. Naval Res. Lab., Washington, DC), Joseph F. Lingevitch, Caleb F. Sieck (Code 7160, U.S. Naval Res. Lab., Washington, DC), Alisha J. Sharma (Code 6040, U.S. Naval Res. Lab., Washington, DC), Donald A. Sofge (Code 5510, U.S. Naval Res. Lab., Washington, DC), and Jason Geder (Code 6040, U.S. Naval Res. Lab., Washington, DC)

Modern autonomous vehicle navigation relies on a fusion of passive and active sensors that are often limited due to computational power, space, and weight constraints. We present a technique to enhance directional beamforming for acoustic fields when the number of available sensors or array size is significantly limited. We have developed a diffraction-based metamaterial approach that maps a broad but finite spectrum of acoustic frequencies

into a directional discriminator. Our approach does not physically constrain the acoustic beamforming array so that the constituent sensors can be used in tandem with a distance estimator that relies on an unobstructed interaction with incident fields. We apply our technique to an unmanned aerial vehicle (UAV) and present a real time hardware demonstration where the vehicle determines the direction to obstacles such as a wall. The experiments use the UAV's broadband self-noise as the interrogating acoustic field, but out technique is general and can be used with passive or active signals. This solution reduced the requisite number of sensors required for acoustic beamforming, which improves the headroom on unmanned platforms to include a wider range of computational tasks and sensor versatility. [Work sponsored by the Office of Naval Research.]

5aPAc8. Acoustics based sensor positining system. Hanieh Agharazi (Univ. of Mississippi, 145 Hill Dr., University, MS 38677, agharazi@olemiss.edu), William G. Frazier, and Wayne Prather (Univ. of Mississippi, Oxford, MS)

The proposed acoustic positioning system addresses the challenge of accurately determining the locations of acoustic sensors in GPS-denied environments, achieving localization within 1% error. In this setup, sensors are deployed by an unmanned ground vehicle (UGV) without prior knowledge of their positions. The system employs an iterative acoustic technique where each sensor broadcasts signals detected by neighboring sensors. The receiving sensors calculate the signal's time-of-flight (ToF) and relay these data to a central computational node. The process repeats until all sensors have transmitted, allowing the central node to compute precise locations by aggregating ToF data. Relative distances between sensor pairs are derived from ToF measurements, and Multidimensional Scaling (MDS) techniques transform these distances into the exact spatial configuration of the sensors. To further enhance accuracy, the wind vector affecting sound propagation is estimated via least squares optimization. Finally, a nonlinear optimization procedure, solved using the Levenberg-Marquardt algorithm, refines both the sensor positions and the wind vector by minimizing discrepancies between measured and calculated acoustic delays.

5aPAc9. Acoustic levitation for additive manufacturing. Quintin D. Dumouchelle (School of Mech. Eng., Purdue Univ., 43936 Tavern Dr., Ashburn, VA 20147, qdumouch@purdue.edu), Luz D. Sotelo, and Michael P. Sealy (School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Laser Powder Bed Fusion (LPBF) is an Additive Manufacturing (AM) process where a laser is used to sinter and fuse powder particles together on a baseplate which acts as a heatsink. This heat transfer results in undesirable residual stresses and potential part failure. A proposed method to study the process of sintering powder without a baseplate is acoustic levitation. A type of acoustic levitation uses a standing pressure wave generated between a set of transducers and a reflector to trap particles at the nodes of the wave. The purpose of this work is to leverage and optimize current standing wave levitation techniques to enable the levitation of metal powder for sintering, i.e., to achieve "bed-less" LPBF through acoustic levitation. Common acoustic levitators are limited to very small or low-density particles. Thus, the first step of this work was to create a simulation tool for the pressure field of an acoustic levitator. The program architecture prioritized input flexibility, including phase, voltage, frequency, number, and position of transducers. The program also outputs the lift capacity to facilitate the following stage of this project which will involve finding optimization solutions for the transducer inputs and acoustic levitator setup suitable for bed-less LPBF.

5aPAc10. Abstract withdrawn.

5aPAc11. Infrapy—An infrasound signal analysis toolkit. Jeremy Webster (Los Alamos National Labs, LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544, jwebster@lanl.gov), Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., Los Alamos, NM), and Jordan W. Bishop (Los Alamos National Lab., Los Alamos, NM)

Infrapy (https://github.com/LANL-Seismoacoustics/infrapy) is a python-based infrasound analysis toolkit developed at Los Alamos National Labs. Its capabilities include waveform analysis, acoustic array beamforming, detection, association, and location. Additionally, it has a spectrogram based single sensor detector. Infrapy can ingest data from local files, from FDSN servers, and from SQL databases that utilize KBCore and CSS schemas. The package includes a fully documented library for script development, a Command Line Interface for analysis from terminal interfaces and shell scripts, and a Graphical User Interface that allows for analysis by those with no programming experience. An overview of the algorithms, workflow, and interfaces will be presented along with example analysis of a recent event of interest.

5aPAc12. Atmospheric sound transmission loss at the coast: A summary of observations. Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Heath Faircloth, Brielle Wagner, Matthew Stengrim, Jeff Foeller (Dept. of Eng., East Carolina Univ., Greenville, NC), John Judge, Diego Turo, and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents advances in an ongoing effort to amass a catalog of atmospheric acoustic transmission loss measurements that are made with coordinating high resolution meteorological observations in a near-shore environment. The measurement layout places the acoustic source at ranges from 250 to 2000 m from shore with a seven-channel vertical linear acoustic receiver array at the shoreline. Pure tone bursts are used with frequencies up to 2 kHz. The concurrent meteorological observations include high resolution temperature profiling, scanning Doppler LIDAR wind profiling measurements in the source-to-receiver direction, and water surface profiling. Observation generalizations are presented for cases of upward and downward refraction at different ranges, normalized to the prevailing wind direction. [Work supported by Office of Naval Research Awards N00014-22-1-2492 and N00014-24-1-2437.]

5aPAc13. Advances in sound ranging from WW1 to WW2. Richard D. Costley (Geotechnical and Structures Lab., U.S. Army ERDC, 3909 Halls Ferry Rd., Vicksburg, MS 39180, richard.d.costley@usace.army.mil)

At the end of the World War 1 (WW1), the Bull-Tucker was the standard sound ranging system used in the British and American Expeditionary Forces for locating artillery. The system consisted of five to six hot wire microphones, spread over 3500 to 7500 yards, and a string galvanometer. After the war, Sound Ranging sections were organized onto observation battalions, alongside Flash Ranging, and transferred from the U.S. Army Corps of Engineers to Corps Artillery. Modifications and improvements in the years following WW1 included switching to condenser microphones with vacuum tube amplifiers. Other modifications included re-packaging the instrumentation for robustness and transportation, however, even then the system weighed 2600 lbs. and occupied 60 ft^3, not counting the 3 tons of twin conductor wire needed for communications and microphones. After the beginning of World War 2 (WW2), other configurations were investigated with the goal of making the system more portable and quickly deployed. The Dodar system was developed with these goals in mind. The Marine Corps deployed the Dodar system in combat areas in the Pacific theater, most successfully in the Battle of Iwo Jima. [Permission to publish was granted by the Director of the Geotechnical & Structures Laboratory.]

5aPAc14. Sound Propagation on Mars: Physics based acoustic predictions. Antonella Bevilacqua (Foster + Partners, 22 Hester Rd., London SW114AN, United Kingdom, abevilacqua@fosterandpartners.com) and George Cann (Foster + Partners, London, United Kingdom)

Studies on sound propagation have always complemented the description of physics related to planetary atmospheres. Since the first space missions, the satellites, entry probes, and landers were provided with microphones to record the propagation through media that differ from Earth. This paper deals with the acoustic predictions on how sound propagation on Mars varies with distance from a sound source. The research is carried out at different octave frequencies, including also the attenuation factor from the Stokes' law in isotropic fluids. The results show that the attenuation at low frequencies is negligible, also at far distance. At medium octaves around 500 Hz, the attenuation starts to be consistent, while the high pitches result inaudible already at 1m distance from the sound source.

Session 5aSP

Signal Processing in Acoustics: Signal Processing Potpourri III

William F. Jenkins, Cochair
Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093

Yongsung Park, Cochair
Woods Hole Oceanographic Inst., Woods Hole, MA

Contributed Papers

8:00

5aSP1. Low-cost underwater acoustic array testbed. Adiba Asad (George Mason Univ., 4400 University Dr., Fairfax, VA 22030, aasad6@gmu.edu), Lukas G. Baines, Jason Coker, Yaoxia Guan, Timothy Higgins, Nadra M. Shaboot, Jeff Tucker, and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

Sensor arrays are important signal processing tools for improving Signal-to-Noise Ratio (SNR) and enabling direction finding. Underwater arrays are used in a variety of applications, such as communications, environmental monitoring, and SONAR. Unfortunately, student access to underwater arrays is often limited by the need for expensive specialized hardware. This talk presents a flexible, low-cost underwater acoustic array testbed built with consumer off the shelf components. The system costs approximately $2200. The transmitter consists of an amplifier driven by a standard laptop audio output and a Lubell Labs UW30 transducer. The receiver consists of an array of four Aquarian Audio A5 hydrophones connected to a USB audio interface for data acquisition by a second laptop. The talk describes the system design and characterizes the transmitter and receiver frequency responses. An experiment illustrates how the testbed can facilitate the study of arrays in underwater communications. The transducer and array are deployed in shallow water environments, i.e., a competition pool and a lake. The experiment measures output SNR and bit error rate using several communication protocols, including on-off keying and frequency-shift keying.

8:20

5aSP2. Delay lines, traveling waves, and filterbanks. Timothy A. Wilson (Slacker Systems LLC, P.O. Box 454, Guymon, OK 73942-0454, wilsonti7@outlook.com)

Delay lines turn time into space: For a tapped delay line, delay increasing with tap index, the instantaneous output considered across the taps is a sampled, reversed, possibly compressed or dilated, version of the input over an earlier time interval. Timing of signal events is estimated using a ruler (cheap) instead of a clock (expensive). Since a delay line solves the transport equation, its output over time is a traveling wave in the direction of increasing tap index. Incremental delay, the difference in delay between adjacent taps, corresponds both to sampling interval and temporal estimation resolution. Graded incremental delay results in tonotopicity: different periodicities map to different places. Incremental delay increasing with increasing tap index yields wavelengths that decrease with tap index such that a constant frequency sinusoidal input presents along the delay line as a decreasing wavelength chirp. Time-domain lowpass filtering at each tap ensures no spatial aliasing. Cutoff frequencies decrease with increasing tap index, restricting higher frequency signals to lower index taps while lower frequency signals extend further along the taps. Preemphasizing high frequencies gives each frequency a best place, with the resulting structure's output resembling that of a crude auditory filterbank.

8:40

5aSP3. Trans-dimensional reflection coefficient inversion of seabed sediments in two spatial dimensions. Tim Sonnemann (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave. FAB 160-01, Portland, OR 97201, tim@pdx.edu), Jan Dettmer (Earth, Energy, and Environment, Univ. of Calgary, Calgary, AB, Canada), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR)

We introduce an adaptive spatially two-dimensional (2-D) inference method for seabed sediment structure and geoacoustic parameters from wide-angle reflection coefficient spectra which does not require piece-wise one-dimensional (1-D) inversions or fixed assumptions about the 2-D parameter space. This is an advance in reflection coefficient inversion, more accurately capturing the information content of the data while retaining a parsimonious representation of the seabed. The approach employs (a) Bayesian inference with the reversible jump Markov chain Monte Carlo algorithm to allow the number of model parameters to change (i.e., trans-dimensional) and (b) 2-D Voronoi tessellations to enable a spatially irregular model grid with a variable number of cells. Synthetic tests indicate that the method estimates the 2-D geoacoustic model and its uncertainties in a more objective and straightforward way than approaches with fixed dimensions or multi-step 1-D inversions, while the computational cost remains similar to previous approaches. The resulting structure and uncertainties are more directly interpretable than those of fixed dimensional modeling methods.

9:00

5aSP4. Vehicle detection and classification using acoustic and seismic data. Abdoulaye Barry (SEAS, Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, abdoulaye.barry@udc.edu) and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, co-operative detection and classification algorithms from measured sound and vibration produced by ground vehicles are presented. Vehicle classification utilizing acoustic and seismic data enables the identification of vehicle types through analysis of sound and vibration characteristics. Machine learning models employed include CNN, LSTM, SVM, and federated learning. The SVM, KNN, and CNN models performed exceptionally well on the test set, with accuracies ranging from 98.78% to 99.76%. The KNN model achieved the highest accuracy of 99.76%, while the CNN model demonstrated a strong performance with a slightly lower accuracy of 99.51%.

9:20–9:40 Break

9:40

5aSP5. Gunshot caliber classification using convolutional neural networks. John Irungu (Comput. Sci., Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, john.irungu@udc.edu) and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, the acoustic classification of raw high and low caliber gunshot sounds using deep learning is investigated. Of particular interest is employing a Convolutional Neural Networks (CNN) prediction model. The raw gunshot sounds are preprocessed to separate the gunshot muzzle blast from other ambient sounds using Blind Source Separation(BSS) implemented using Short-time Fourier Transform (STFT) and Non-Negative Matrix Decomposition (NMF) to the magnitude's spectrogram. The proposed model has an accuracy of 97%, precision, recall, and F1-score of 97%,99%, and 98%, respectively.

10:00

5aSP6. Acoustic sensing for overbuild-recoater blade collision detection in laser powder bed fusion additive manufacturing. Emmeline Evans (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30308, eevans70@gatech.edu), James Mavo (NASA Marshall Space Flight Ctr., Huntsville, AL), and Aaron Stebner (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Laser powder bed fusion (LPBF) additive manufacturing is widely used to fabricate geometrically complex metal components. In LPBF, a layer of metal powder is swept onto the build surface by a recoater blade and then melted with a laser; this process is repeated to build a part layer-by-layer. A common flaw in LPBF processes is overbuilding, where some region of a layer is built higher than the desired layer height. Collisions between the overbuild and the recoater blade on subsequent sweeps can lead to a persistent defect in the part or damage to the recoater blade. In this research, acoustic emission sensors mounted to the recoater blade subassembly of an EOS M290 machine are used to detect collisions with the recoater blade. This work presents the sensor configurations for and results of preliminary experiments conducted during powder sweeps only and during fabrication of small test coupons. [Work supported by NASA Space Technology Graduate Research Opportunities (Grant No. 80NSSC23K1213)]

10:20

5aSP7. Unveiling persistent structures in long-range acoustic m-sequence data with regularized machine learning. Andrew J. Christensen (College of Elec. and Comput. Eng., Univ. of Iowa, 3100 Seamans Ctr., Iowa City, IA 52242, andrew-christensen@uiowa.edu), Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Nicholas C. Durofchalk (Dept. of Phys., Naval Postgrad. School, Monterey, CA), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI), and Kay L. Gemba (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

Monitoring ocean dynamics via long-range acoustic transmissions is a topic of much interest. Characterizing complex physical systems, such as the ocean, is challenging because their dynamics often cannot be fully described by simple analytical models. Recently, machine learning methods have gained traction in the sonar signal processing community, offering alternatives to classical statistical and signal processing methods. These traditional methods often struggle in real ocean environments, where nonlinear effects, such as internal waves, distort transmitted acoustic pulses. This work investigates using modern machine learning techniques to unravel these complexities by identifying persistent, slowly evolving features. We assess the capabilities of the proposed machine learning methods using transmitted maximum length sequence signal pulses from the Kauai Beacon source collected at the International Monitoring Station near Wake Island. Additionally, we demonstrate that incorporating regularization constraints enhances interpretability, providing deeper insight into the learned models. [Work funded by DoD Navy (NEEC) Grant No. N00174201001, and the ONR grant numbers N000142112420 and N000142312503.]

10:40

5aSP8. HybrA: Learnable, interpretable, and stable filterbanks for feature extraction. Peter Balazs (Acoust. Res. Inst., Austrian Acad. of Sci., Dominikanerbastei 16, Vienna 1010, Austria, peter.balazs@oeaw.ac.at), Daniel Haider, Felix Perfler (Acoust. Res. Inst., Austrian Acad. of Sci., Vienna, Austria), and Vincent Lostanlen (Nantes Université, École Centrale Nantes, Nantes, France)

When designing machine learning models, especially for acoustics applications, there is an ever-repeating question: how should feature extraction be done? The two most common approaches are to use a hand-crafted fixed filterbank or to process the audio signal directly in an end-to-end manner. Each has its own pros and cons, e.g., the first is not adaptive, and the second is poorly interpretable. We present a hybrid approach that combines the strengths of both. This approach involves convolving each filter of a fixed filterbank with a short learned kernel. In this way, we obtain a new *hybrid filterbank* that keeps (some) interpretability while being able to adapt to the problem at hand. To ensure that no information is lost during this initial processing stage, we additionally incorporate a regularizer that maintains stability. We present a concrete example of constructing a hybrid filterbank using an audlet transform—an invertible, perception-related time-frequency transform—and demonstrate its effectiveness in a denoising task.

Session 5aUW

Underwater Acoustics and Acoustical Oceanography: Bill Kuperman (1943–2024): Contributions to the Field of Underwater Acoustics I

Kevin D. Heaney, Cochair
Applied Ocean Sciences, Fairfax Station, VA 22039

Christopher Verlinden, Cochair
Applied Ocean Physics LLC, Fairfax Station, VA

David R. Barclay, Cochair
Dept. of Oceanogr., Dalhousie Univ., P.O. Box 15000, Halifax B3H 4R2, Canada

Chair's Introduction—7:00

Invited Papers

7:05

5aUW1. Bill Kuperman and the coupled mode parabolic equation. Ahmad T. Abawi (HLS Res., 12625 High Bluff Dr., Ste. 211, San Diego, CA 92130, abawi@hlsresearch.com)

I worked with Bill Kuperman as his first postdoc in the fall of 1993, after he joined the Scripps Institution of Oceanography as director of the Marine Physical Laboratory earlier that summer. I had just completed my Ph.D. at the physics department at UCSD. Bill's first task for me was to develop a 3D propagation model using normal modes. The resulting model was a modification of the adiabatic parabolic equation, which Bill aptly named the Coupled Mode Parabolic Equation (CMPE). The solution consists of normal modes and modal coefficients that satisfy the coupled horizontal wave equation. The CMPE is an energy conserving model and can be solved efficiently using the operator splitting method. This work led to the publication of [Abawi, Kuperman and Collins, "The coupled mode parabolic equation," JASA, 102 (1), July 1997]. In this presentation, the CMPE solution is outlined and compared to solutions obtained using more modern techniques. While this work has inspired numerous other studies, it represents only a small fraction of Bill Kuperman's vast contributions to underwater acoustics and his enduring legacy. Bill was a mentor and a close friend, who is largely responsible for my chosen career path.

7:25

5aUW2. Passively localizing a quiet moving source with less prior information. Franklin H. Akins (Scripps, UCSD, 562 Arenas St., La Jolla, CA 92037, fakins@ucsd.edu)

Bill Kuperman had a restless energy and insatiable appetite for new research topics, which led him to contribute in many areas of acoustics including rough elastic surface scattering, sea surface noise, matched field processing (MFP), time reversal, and even the acoustic properties of neurons. When I first met Bill, he summarized his research interest as finding ways of "doing something with less." In the context of passive acoustic localization, this amounted to finding a source with lower signal-to-noise ratio (SNR) or with a smaller array or with less knowledge of the acoustic environment. This led us to develop my thesis work on "range-coherent" matched field processing which coherently combines data collected from a moving source traversing multiple range cells. The method successfully localized low SNR tonal sources in the SWellEx-96 experiment. Another topic of interest to Bill was the use of ambient noise as a signal. In this spirit, we developed a normal mode estimation algorithm using a partially spanning vertical line array (as opposed to a full water column spanning array) which we called modal-MUSIC. We were then able to use the normal modes estimated from noise for low-SNR localization with range-coherent MFP. In addition to his many technical contributions to the field of underwater acoustics, Bill had a gift for mentorship and was a joy to work for and with.

7:45

5aUW3. Bill Kuperman: Navy Scholar, author of seminal articles and mentor. Arthur B. Baggeroer (Mech. and Elec. Engineers, Massachusetts Inst. of Technol., Rm. 5-205, MIT, Cambridge, Cambridge, MA 02139, abb@mit.edu)

Bill Kuperman: Navy Scholar, author of seminal articles, mentor to many and ambassador for acoustics in the United States (US). We are all familiar with Bill's extraordinary contributions in the first three categories, but few would appreciate his efforts for promulagation of research at the forefromt of ocean acoustics. Most in the ASA community is aware of his many conferences, but few are probably aware of the extensive dialog he maintained with those beyond the US. I was fortunate enough to join him in our visits. We had many trips to the FSU during the period of Perestroika, including several Academician Berovovskikh, many visits to the PRC where we

observed the rapid rise of PRC ocean acoustics, a series of ongoing seminars in Corsica, lectures at many, many European universities and companies. He brought the excellence of US acoustics overseas which few could have done.

8:05

5aUW4. Bill Kuperman: The middle years. David L. Bradley (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Portsmouth, NH, david.bradley@unh.edu)

Bill returned to The Naval Research Laboratory in the mid 80's. The multi-decadal cold war was coming to an end. The Office of Naval Research (ONR) had established the Office of Naval Technology (ONT) in 1980 which had an emphasis of transitioning fundamental research results to applications for the operational Navy. Bill, a master of establishing partnerships and an excellent judge of research potential in young candidates, and reflecting his early work at NRL, the SACLANT Centre in La Spezia, Italy and NORDA (The Naval Oceanographic Research and Development Activity) in Bay St Louis, MS, quickly became a major contributor to the ONT undersea program and Chief Scientist of the NRL Acoustics Division. In retrospect, though clearly not the case, it looks like he was preparing for his time at Scripps and The Marine Physical Laboratory

8:25

5aUW5. Memories of Bill Kuperman at the Naval Ocean Research and Development Activity, circa 1981. Stanley A. Chin-Bing (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, chin-bing@att.net)

Bill Kuperman arrived at the Naval Ocean Research and Development Activity (NORDA), Stennis Space Center, MS, circa 1981. He had just completed a sabbatical tour at the NATO SACLANT Undersea Research Center in La Spezia, Italy, and had accepted the position of Division Head of the NORDA Numerical Modeling Division. As a junior member of the Acoustic Branch in this division I was assigned to work with Bill. Specifically, Bill was going to mentor me in underwater acoustic research. During this mentoring period I assumed several different roles: student, junior colleague, and Acoustical Society of America (ASA) abstract reviewer and arranger of underwater acoustics session for ASA meetings. On occasion, I also acted as Bill's pseudo-administrative assistant. My various interactions with Bill allowed me the opportunity to know and observe him on several different professional levels. In this talk, I will present some interesting examples that revealed the leadership qualities and subtle humor that a young Bill Kuperman possessed, both as a manager and as a researcher.

8:45

5aUW6. The ray angle diagram in ocean acoustic propagation. Henry Cox (RMS, Lockheed Martin, 9500 Godwin Dr., Manassas, VA 20110, harry.cox@lmco.com)

The seldom mentioned ray angle diagram (RAD) allows visualization of important properties of acoustic propagation. Based on the sound speed profile and Snell's law, the RAD presents depth on the ordinate and tan $\theta(z; c_n)$ on the abscissa for the full ray cycle of selected rays, parameterized by the ray parameter, c_n. The angle $\theta(z; c_n)$ is the angle the ray makes with the horizontal at depth z. Equivalently, it is the angle of tilt of the wavefront at depth z. Because many useful characteristics of acoustic propagation depend on tan $\theta(z; c_n)$, it is used as the abscissa of the RAD. The tilt integral is defined as the integral of tan $\theta(z; c_n)$ with respect to z between the upper and lower limits of the ray. The tilt integral can be visualized by an "area" on the RAD. The WKB phase integral, which relates rays and normal modes, is an integral of tan $\theta(z; c_n)$ with respect to depth. Individual mode characteristics and number of propagating normal modes are represented on the RAD. Relationships to the adiabatic invariant, time delay for matched field processing, depth dependence of ambient noise, range-averaged transmission loss, and low frequency mode cut-off are discussed.

9:05–9:25 Break

9:25

5aUW7. If could turn back time reversal communications. Geoffrey Edelmann (Code 7160, The U.S. Naval Res. Lab., 521 E Luray Ave. Alexandria, VA 22301, geoffrey.f.edelmann.civ@us.navy.mil)

This presentation will provide an overview of Bill Kuperman's groundbreaking research in acoustic communications, specifically his work on Time Reversal (TR), a notable chapter in his distinguished career of achievements. TR backpropagates a signal to its origin, despite the complexity of the propagation channel. A pioneering at-sea experiment was conducted to measure the focus of a 3.5-kHz centered time-reversal mirror (TRM), which not only demonstrated the feasibility of the concept but also introduced a novel methodology for generating binary-phase shift keying communication sequences using a TRM. A comparative analysis of the results reveals that time reversal is a highly effective approach for mitigating inter-symbol interference caused by channel multipath. Furthermore, the presentation will explore the advantages of cascading the TR process with adaptive channel equalization, a combination that eliminates the need for spatial diversity in the receiver structure. Bill's remarkable body of research will be celebrated through personal anecdotes and stories of his mentorship, friendship, and signature humor, offering a unique glimpse into the man behind the science. [Work supported by ONR.]

9:45

5aUW8. Global acoustic propagation: Perth-Bermuda revisited, again, and again. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

Bill Kuperman (1943-2024) made a foray into global acoustics to address a long-standing puzzle facing the ocean acoustics community. In 1960, a set of detonations set off in the ocean near Perth Australia were recorded on underwater hydrophones off the island of Bermuda. Walter Munk (1917-2019) had looked into this problem and concluded the source and receiver were in the shadow zones and, therefore, should not have been heard. By adding the vertical component of the sound speed, rather than using the surficial global sound

speed, Heaney and Kuperman showed that with refraction around islands a hybrid mode-ray code could explain the phenomenon. Bill launched me into the field of ocean acoustics and global acoustic propagation, in particular. This work has led to the recognition that most propagation in the ocean below 30 Hz is 3-dimensional in nature. This work has led to the inclusion of 3D propagation effects in the Comprehensive Test Ban Treaty Organization real time detection and localization system in Vienna. Bill's impact on my life, as recruiter, mentor, and sage, is hard to overstate. His example of generosity and creative curiosity is unmatched in my experience.

10:05

5aUW9. Time reversal experiments in the ocean. William Hodgkiss (SIO, UCSD8820 Shellback Way, San Diego, CA, whodgkiss@ucsd.edu) and Heechun Song (SIO, UCSD, La Jolla, CA)

A time-reversal mirror (TRM) (or phase conjugate array) spatially and temporally refocuses an incident acoustic field back to its origin. Bill Kuperman and colleagues at the Marine Physical Laboratory of the Scripps Institution of Oceanography carried out a series of time reversal experiments 1996–2006 in collaboration with Tuncay Akal and colleagues at the NATO SACLANT Undersea Research Centre (now Center for Maritime Research and Experimentation) in La Spezia, Italy. The initial experiments were carried out in shallow water off the west coast of Italy. These utilized a 77 m aperture source-receiver array (SRA) along with a probe source deployed by the R/V Alliance and a 90 m aperture receive array for exploring the focal region from the SRA transmissions. These experiments demonstrated TRM focusing at ranges 5–30 km, explored temporal stability of the focus (hours to days), demonstrated moving the focus $+/-10\%$ range through shifting the center frequency of the received pulse, and impact of implementing an iterative time reversal process. In subsequent experiments, the SRA was redesigned to operate at high frequency (3.5 kHz) enabling exploring time reversal applications in acoustic communications and reverberation nulling. The presentation will show highlights from this series of time reversal experiments.

10:25

5aUW10. Applications of adjoint modeling. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

An adjoint model backpropagates a special field from the receiver back through the medium to the source. The field is initialized by the forward model mismatch with measurements—i.e., we run our forward model through a medium whose properties are only known approximately, then compare the forward-propagated field with the measured field at all receiver points. Because we do not know the medium properties perfectly, there is mismatch. The special field back-propagated by the adjoint model is initialized by that mismatch. The special field yields the gradient of the medium properties with respsect to the mismatch, enabling steepest descent methods to be used to invert the mismatch for the unknown refinements to the medium properties that would reproduce the measurements. An adjoint modeling approach to obtaining the gradients needed for the inversion requires two modeling runs, one run of the forward model, and one run of the adjoint model. We will review methods for deriving an adjoint model for some underwater acoustic applications, and demonstrate how an adjoint model can be used in inversion problems. We will also discuss connections to similar methods used in similar fields.

Contributed Paper

10:45

5aUW11. Experimental validation of passive time-reversal and ray-tracing methods for underwater sound source localization. Kuan-Wen Liu (Dept. of Hydraulic and Ocean Eng., National Cheng Kung Univ., No. 1, Daxue Rd., East Dist., Tainan 701, Taiwan, kwliu@gs.ncku.edu.tw) and Ching-Jer Huang (Dept. of Hydraulic and Ocean Eng., National Cheng Kung Univ., Tainan, Taiwan)

This study presents method for localization of an underwater sound source using a passive time-reversal method (TRM) in combination with a ray-tracing approach. For 2-D localization, sound signals from an underwater source are collected via a vertical array of four hydrophones. These signals are processed using the ray-tracing code BELLHOP to determine the acoustic pressure field of the time-reversed waves. The location of maximum pressure, based on TRM's retrofocus characteristic, identifies the source's range and depth. Validation experiments in a towing tank, along with field tests near Yanpu Harbor and Small Liuqiu Island, Taiwan, demonstrate that estimated positions closely match actual source locations. Furthermore, a cross-shaped hydrophone array is used to capture source signals. The source's 2-D position is first determined from signals collected by the vertical hydrophones in the r-z plane, followed by calculating the pressure field from the horizontal hydrophones to find the 2-D position in the x-y plane, thus determining the source location at (x, y, z).

Session 5pAA

Architectural Acoustics: Materials for Sound Absorption, Diffusion, and Transmission Loss

Molly Smallcomb, Cochair
Threshold Acoustics, 141 West Jackson Blvd., Suite 2080, Chicago, IL 60604

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 County Rd. 102, Oxford, MS 38655-8604

Contributed Papers

1:00

5pAA1. Exploration for a new pathway on the absorption coefficient calculation of glass reinforced gypsum material in temple hall's acoustic improvement. Ziqing Tang (Taiyuan Univ. of Technol., No. 79 Yingze West St., Taiyuan 030024, China, 525831368@qq.com)

Absorption coefficient is a long-standing point of discussion in acoustic research field, and research methods of its calculation rely on experimental test (classic $RT_{reverberation}$ chamber method) or software application simulation (for example, $_{Zorba}$). We put forward with a different pipeline for absorption coefficient in the process of sound absorption quantitative evaluation for one porous material, a new thinking for. In this study, we chose a chanting hall in Longquan Temple located in Haidian District in Beijing of China, discovering the deficiency of its existing acoustic Rt (1.9 s, higher than that from the China National Standard GB/T 20247). We designed the decoration of the three positions' layers in the roof, walls, and seats on the floor, in order to upgrade its interior reverberation performance, by the Glass Reinforced Gypsum material, which is very suitable for ancient timber structure architecture. Alternatively, it proves to be a relatively simpler pathway for calculation of in this new attempt than the traditional RT method test, concerning with the grand space timber architecture, especially for the intangible cultural heritage conservation as for sound environment of the temple's hall, in further acoustic exploration.

1:20

5pAA2. Examining the impact of surface material design treatments on traditional and collaborative instructional approaches in hybrid classrooms. Abdul Wafi Razali (Dept. of Quantity Surveying, Universiti Malaysia Sarawak, Kuala Lumpur, Wilayah Persekutuan, Malaysia), Nazli Bin Che Din (Dept. of Architecture, Universiti Malaya, Faculty of Built Environment, Universiti Malaya, Lembah Pantai, Kuala Lumpur, Wilayah Persekutuan 50603, Malaysia, nazlichedin@um.edu.my), Musli Nizam Yahya (Dept. of Transportation Eng. Technol., Tun Hussein Onn Univ. of Malaysia, Pagoh, Johor, Malaysia), Raha Sulaiman (Dept. of Bldg., Surveying, Universiti Malaya, Kuala Lumpur, Wilayah Persekutuan, Malaysia), and Asrul Sani Razak (Dept. of Architecture, Universiti Malaya, Kuala Lumpur, Wilayah Persekutuan, Malaysia)

Hybrid teaching has gained substantial interest in higher education and is anticipated to influence future pedagogical reforms worldwide in the aftermath of the pandemic. Despite the adaptability of hybrid teaching as a flexible instructional design, various limits have emerged. Poor sound quality was a significant challenge for distant students, impairing their ability to effectively comprehend lectures and engage in interactions with on-site peers and instructors. Therefore, this study aims to investigate the impact of surface material design treatment on two distinct educational approaches in hybrid classroom environments. This study incorporates field measurements and acoustic modeling methods to develop effective surface design treatments that enhance the listening experience for remote students. Three significant acoustical parameters were assessed: reverberation time (RT), speech transmission index (STI), and speech clarity (C50). The results of an in-depth investigation are beneficial for designers and educational institutions in ensuring appropriate acoustic quality for hybrid learning settings. This setting concurrently enhances students' learning experiences and performance.

1:40

5pAA3. A field experiment on the effect of sound absorption on acclimation of new children to kindergarten—Part 1: Analysis of crying children and indoor noise levels. Ikuri Matsuoka (Kumamoto Univ., 2-39-1 Kurokami Chuo-ku Kumamoto, Kumamoto, Kumamoto 860-8555, Japan, musicalonmyown@yahoo.co.jp) and Keiji Kawai (Kumamoto Univ., Kumamoto-shi, Japan)

A field experiment was conducted where sound absorbing materials were temporarily installed in actual kindergarten classrooms to examine the effect of sound absorption on acclimation of new children to kindergarten. There were two classrooms for two 2-year-old groups and sound absorbing material was temporarily installed in one of the two rooms in April, beginning of the school year in Japan, and removed and re-installed in November. By the installation, the reverberation times in the 1 kHz band of the two rooms, which have different room dimensions, were reduced from 0.6/0.8 s to 0.4/0.5 s. The indoor activities were recorded by sound and video recorders. As a result, the time ratio of children's crying during morning gatherings in April was observed for six days, and the time ratio in the room without sound absorption was between 3%-93% in 5 days, whereas 10 % in one day with sound absorption. SPL in the two rooms were compared in the same condition without sound absorption in the both rooms in November, and the SPL in the room with sound absorption until November was lower by around 2 dB, indicating that the voice level was lower than the other room without absorption throughout.

2:00

5pAA4. A field experiment on the effect of sound absorption on acclimation of new children to kindergarten—Part 2: Estimating children's crying by acoustic event detection. Keiji Kawai (Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto-shi 860-8555, Japan, kkawai@kumamoto-u.ac.jp), Ikuri Matsuoka, Reo Hori, and Naoya Maruyama (Kumamoto Univ., Kumamoto, Japan)

A field experiment was conducted where sound absorbing materials were temporarily installed in actual kindergarten classrooms to examine the effect of sound absorption on acclimation of new children to kindergarten. There were two classrooms for two 2-year-old groups and sound absorbing material was temporarily installed in one of the two rooms. Since children who have just entered kindergarten often cry because they do not want to be there, the time ratio at which children were crying was analyzed in Part 1 judging based on recorded video and sound, which required much time and labor. Therefore, this study attempted to employ an acoustic event detection method using Mel-Frequency Cepstral Coefficients (MFCC) features in the

analysis. The training data consisted of 1180 2-s sound segments recorded in the classroom on five days, and then 400 2-s segments consisted of 200 sounds each judged by human to be crying or non-crying, recorded on different 14 days than the training data, were analyzed to compare human and computer judgments. The result showed that 153/200 segments were judged to be crying and 183/200 were judged not to be crying by computer, indicating a good correspondence.

2:20

5pAA5. The application of small coupled reverberation rooms for testing sound-insulating metamaterial structures. Agata Szelag (Cracow Univ. of Technol., Ul. Warszawska 24, Kraków 31-155, Poland, agata.szelag@pk.edu.pl), Katarzyna Baruch-Mazur, and Dorota Mlynarczyk (Cracow Univ. of Technol., Cracow, Poland)

The paper presents possible applications of small coupled reverberation rooms for testing sound-insulating metamaterial structures. First, the authors describe in detail the construction and validation of this measurement stand proving that both the stand and the testing methodology conform to the ISO 10140 standards and that the obtained measurement uncertainty is comparable with the typical uncertainty given in ISO 12999-1 for full-size rooms. Next, the modifications of the stand are described in the context of the possibility of mounting samples of different sizes and thicknesses. Finally, several examples of the use of the stand for measuring the sound insulation of metamaterial structures were presented. The following types of metamaterials were tested at the stand: nested Helmholtz resonators, locally resonant metamaterials, vibroacoustic metamaterials. The measurements concerned both scaled structures to smaller dimensions and full-size samples. [Work was funded by the National Centre for Research and Development, Poland, Grant No. LIDER13/0030/2022.]

2:40

5pAA6. Characterization of the propagation coefficient and characteristic impedance in an extended frequency range. ZIQI CHEN (Graduate Program in Architectural Acoust., RPI, 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., RPI, Troy, NY)

Characteristic impedance and propagation coefficient are two critical parameters in acoustical material research. This work uses Bayesian inference to explore the performance of characteristic impedance models at high frequencies. The characteristic impedance and propagation coefficient are measured using the three-mic method [Salissou & Panneton, J. Acoust. Soc. Am. 128, pp. 2868–2876 (2010)]. To extend the frequency range, multiple microphone measurements are averaged. The empty impedance tube measurement with a hypothetical air layer enables the validation of the extended-range characterization. The critical parameters in characteristic impedance models are estimated using Bayesian inference. Porous materials are used to compare the main existing characteristic impedance models.

3:00–3:20 Break

3:20

5pAA7. Investigation of errors and uncertainty in low-frequency impedance tube measurements. Matthew Ripley (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 1129 Sage Ave. Troy, NY 02180, riplem@rpi.edu), ZIQI CHEN (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

The accuracy of impedance tube measurements can be improved by determining the acoustic conditions within the impedance tube prior to introducing a sample for test. By measuring the empty impedance tube, the influence of environmental changes on the critical parameters can be minimized. The low-frequency dissipation is estimated using Bayesian inference to incorporate into a transfer function model, reducing error when measuring samples. Previous work [Chen & Xiang, J. Acoust. Soc. Am., 155, 2646–2658 (2024)] has focused on a frequency range of 1.5–5 kHz; the subject of this study is a lower frequency range (<1.5 kHz) using larger sized impedance tubes. This talk details the methodology and results of the study, its inclusion within an updated transfer function model, the accuracy of the model, recommendations for improved impedance tube measurement procedures, and next steps.

3:40

5pAA8. Engineering tools for reliable sound insulation predictions. Klas Hagberg (Acouwood, Dockgatan 43, Malmö 21173, Sweden, klas.hagberg@acouwood.com)

Measurements are an important part in the field of acoustics and something acousticians rely on. A significant part of a building acoustician's job is to design and predict sound insulation. Accurate measurements are essential for developing reliable tools for sound insulation predictions. However, apart from ISO 12354, there is no common view of how the sound insulation should be predicted, and even this excellent standard series is not widely used, apart from some small regions globally. Different approaches can be considered when developing engineering tools to predict sound insulation. Apart from the in-situ calculations described in ISO 12354, other prediction tools are needed for different combinations of floor and wall configurations. The possible combinations of materials are nearly infinite, and new materials are continually entering the market. For some building parts, pure physical models yield the highest prediction accuracy while in other cases, empirical models based on measured data are preferable. Artificial neural networks can also serve as a prediction model. However, these models require extensive training on many measurements to yield accurate results. To achieve high accuracy, engineering tools may need to incorporate different models for specific combinations. Then, the key question remains: will acousticians trust such prediction tools?

4:00

5pAA9. Comparative study: Various ceramic acoustic underlayments for ultra high-end penthouse renovation in Canada. Felix Baronet (Mech. Eng., Université Laval, 150 Bd Léon Vachon, Saint-Lambert-de-Lauzon, QC G0S2W0, Canada, fbaronet@acousti-tech.com)

In multi-residential construction, achieving effective great sound insulation in flooring systems is a critical yet challenging objective, particularly with ceramic flooring. This study explores an in-depth comparison of acoustic performance of several floor solutions, including the usual engineered wood floating floors assembly, tested in a high-end penthouse unit. Key tests and measurements were conducted by *SiBE Acoustique* and *AcoustiTECH*, assessed impact transmission noise isolation, using ASTM-compliant methodologies. The findings highlight how innovative underlayment solutions, can enhance the acoustic performance of ceramic assemblies, meeting or surpassing the acoustic benchmarks of traditional floating floors. The study also addresses frequency-specific behaviors, emphasizing the importance of low-to-mid frequency performance in occupant comfort. This presentation will detail the comparative analysis of acoustic ratings (ISR, AIIC), provide insight into the technical particularities of each tested systems' design and discuss the implications for construction professionals and acoustic consultants in developing practical, high-performance flooring solutions. By bridging the gap between aesthetics, durability, and high acoustic constrains, this study contributes to advancing complex impact attenuation performance system in multi-residential environments.

4:20

5pAA10. Numerical validation of the low-frequency procedure for façade sound insulation measurement: Verification of applicable room dimensions. Jinyu Liu (Kanagawa Univ., 3-27-1 Rokukakubashi, Kanagawa-ku, Yokohama-shi, Kanagawa 221-8686, Japan, rika0626@kanagawa-u.ac.jp), Naohisa Inoue (Kyushu Univ., Fukuoka prefecture, Japan), Tetsuya Sakuma (Architecture, The Univ. of Tokyo, Tokyo, Japan), and Yosuke Yasuda (Kanagawa Univ., Yokohama, Kanagawa, Japan)

In the ISO 16283 series for field measurement of sound insulation, the low-frequency procedure is specified for determining indoor average sound pressure level in rooms below 25 m^3, where the sound field is dominated by only a few modes. In our previous study, the validity of the procedure was examined through field measurements of façade sound insulation in a wooden house with a room volume of about 20 m^3, and some possible modifications were proposed in the procedure. However, it remains unclear

whether those modifications are valid for other rooms with different room dimensions, different building structures, and indoor sound absorption conditions. Specifically, the upper limitation of room volume of 25 m^3 specified for the low-frequency procedure has not been fully verified. Therefore, this paper investigates the applicability of the modified procedure through numerical simulations with various room configurations. In the simulations, a vibro-acoustic FEM is employed for a simplified room model on the rigid ground surface, where building components are modeled with plate elements. The results demonstrate the validity of the modified procedure in a wider range of room conditions and clarify the applicable range of room dimensions.

4:40

5pAA11. Design and optimization of meta-material ventilated sound barriers for building facades: An experimental and numerical investigation. Mohammad Tabatabaei Manesh (Univ. of Washington, 3950 University Wy NE, Seattle, WA 98105, mhtaba@uw.edu) and Tomás I. Méndez Echenagucia (Univ. of Washington, Seattle, WA)

Conventional building facade materials often struggle to balance sound attenuation and ventilation. As urbanization accelerates and the demand for sustainable, occupant-friendly structures grows, innovative solutions are needed to address these dual challenges. This research introduces a meta-material ventilated sound barriers using helical cavities to enhance sound attenuation while ensuring natural ventilation. The methodology involves two stages: first, designing and optimizing arrays of helix resonators using numerical models in COMSOL Multiphysics. The optimization process evaluates critical parameters such as helix dimensions and arrangement within an array to maximize transmission loss and airflow. Second, the results are validated experimentally using an impedance tube. The proposed design is expected to deliver superior acoustic transmission loss while maintaining airflow. This combined numerical and experimental approach aims to advance the acoustic and ventilation performance of facade materials, offering a foundation for more sustainable and efficient building designs.

FRIDAY AFTERNOON, 23 MAY 2025 BALCONY K, 1:00 P.M. TO 4:40 P.M.

Session 5pBA

Biomedical Acoustics: General Topics in Biomedical Acoustics: Tissue Characterization

Yunbo Liu, Chair
FDA, 10903 New Hampshire Ave., Silver Spring, MD 20993

Contributed Papers

1:00

5pBA1. Ultrasonic reflection and attenuation by the bone cortex—Implications for backscatter measurements of cancellous bone. Keith T. Hoffmeister (Dept. of Phys., Rhodes College, 2000 North Parkway, Rhodes College, Memphis, TN 38112, hofkt-25@rhodes.edu), Layla A. Lammers, Kate E. Hazelwood, Hugh E. Ferguson, and Brent K. Hoffmeister (Dept. of Phys., Rhodes College, Memphis, TN)

Ultrasonic backscatter techniques are being developed to detect changes in the porosity of cancellous bone caused by osteoporosis. Cancellous bone is surrounded by a thin layer of cortical bone (the cortex) that can produce errors in backscatter measurements, especially for non-normal angles of incidence. The goal of this study was to develop a theoretical model to predict power loss in the backscatter signal caused by reflection and attenuation by the cortex. Predictions of the model were tested using a polymer foam to simulate cancellous bone with one surface embedded in a $\sim$3 mm layer of epoxy to simulate cortical bone. Measurements were made with a 3.5 MHz transducer for angles of incidence ranging from 0 to 30 deg relative to the normal. Good agreement was observed between the theoretical predictions and experimental results.

1:20

5pBA2. Enhancing ultrasound tissue characterization with the double Nakagami distribution model. Ladan Yazdani (Radiology, Weill Cornell Medicine, 416 E 55th St., MR-007, New York, NY 10022, lya4002@med.cornell.edu), Cameron Hoerig (Radiology, Weill Cornell Medicine, New York, NY), Tadashi Yamaguchi (Ctr. for Frontier Med. Eng., Chiba Univ., Chiba, Japan), Kazuki Tamura (Inst. of Photonics Medicine, Hamamatsu Univ., Hamamatsu, Shizuoka, Japan), Jonathan Mamou, and Jeffrey A. Ketterling (Radiology, Weill Cornell Med., New York, NY)

The single Nakagami distribution (SND) models backscatter envelope statistics for tissues with a single type of scatterer but is less effective for tissues containing multiple scatterer types. This study introduces a new Double Nakagami distribution (DND) estimator to model tissues with two types of scatterers. The DND model directly estimates three parameters corresponding to number density and relative volume ratio of the first scatterer type, from which two additional parameters corresponding to the second scatterer type are derived through a method-of-moments approach. An algorithm was developed for efficient model initialization. To validate the method, simulations were conducted by generating envelope statistics data with various

combinations of the five DND parameters. Experimental data were also collected from agar phantoms containing varying concentrations of nylon and acrylic spheres. The simulations showed the DND model parameter estimation was more accurate than the SND model with errors below 5%. For the phantom tests, the DND model accurately estimated scatterer proportions with errors below 6%. The DND fit model demonstrated significant advantages in terms of computational accuracy for tissues containing more than one type of scatterer. The DND model has the potential to improve quantitative tissue characterization when there are multiple scatterer types.

1:40

5pBA3. Effect of angle of incidence on ultrasonic backscatter measurements of bone. Brent K. Hoffmeister (Dept. of Phys., Rhodes College, 2000 N Parkway, Memphis, TN 38112-1624, hoffmeister@rhodes.edu), Kate E. Hazelwood, Hugh E. Ferguson, Layla A. Lammers, and Keith T. Hoffmeister (Dept. of Phys., Rhodes College, Memphis, TN)

Osteoporotic changes in the porous microstructure of cancellous bone can be detected using ultrasonic backscatter. Ideally, the incident pulse should be perpendicular to the bone surface, but this is difficult to achieve clinically with a handheld probe. The goal of this study was to assess the sensitivity of multiple backscatter parameters to errors caused by non-normal incidence. Measurements were performed on a polymer foam that simulates the ultrasonic properties of cancellous bone with one surface embedded in $\sim$3 mm of epoxy to simulate the outer bone cortex. Ultrasonic pulses were transmitted at angles of incidence ranging from 0 to 30 deg using a 3.5 MHz transducer. Six backscatter parameters were tested. Three were based on spectral analyses of the power in the backscatter signal. The other three were based on the rate the amplitude of the signal decreased with time, or equivalently depth, in the tissue. The latter three parameters performed well with errors less than 30% in most cases. Large errors ($>$100%) were observed for other parameters. These results indicate that errors in backscatter measurements of bone caused by non-normal incidence depend strongly on the parameter chosen.

2:00

5pBA4. Passive cavitation detection analysis of color Doppler twinkling from polymethyl methacrylate. Benjamin Wood (Grad. School of Biomed. Sci., Mayo Clinic, 200 1st St. SW, Rochester, MN 55902, wood.benjamin@mayo.edu), Md Aktharuzzaman, Christine Lee, and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

The color Doppler twinkling artifact has been attributed to existing microbubbles or cavitation has been reported for objects with rough surfaces like kidney stones, sandpaper, and polymethyl methacrylate (PMMA) bone cement. Passive cavitation detection (PCD) is a method for determining the presence of bubble oscillation from subharmonic, ultraharmonic, and inharmonic components in the Fourier domain. We evaluated multiple PMMA samples with varying twinkling amounts as well as a non-twinkling metal rod. Ultrasound scans were conducted using a Verasonics L11-4v probe on a V1 system from 3–9 MHz Doppler transmit frequency. A Sonic Concepts Y-107 PCD probe connected to a JSR DPR300 pulser/receiver collected data during Doppler ultrasound transmission. The Fourier spectra showed increased magnitude trends in the midband frequency range between the fundamental, f, and second harmonic, 2f, on the twinkling samples compared to non-twinkling metal rod. The midband increase was highest at frequencies that showed the most twinkling. This work indicates that the presence of color Doppler twinkling signal on PMMA gives rise to additional peaks in the Fourier domain, particularly in the midband range. Additionally, as there were no distinct subharmonic peaks, the increased midband magnitudes could be due to cavitation on the rough surfaces of the PMMA.

2:20

5pBA5. Left ventricular blood flow quantification of dilated cardiomyopathy murine knockout model. Geraldi Wahyulaksana (Radiology, Weill Cornell Medicine, 416 E 55th St., New York, NY 10022, gew4002@med.cornell.edu), Colin K. L. Phoon (Div. of Pediatric Cardiology, NYU Langone health, New York, NY), Glenn I. Fishman (Leon H. Charney Div. of Cardiology, NYU Grossman School of Medicine, New York, NY), and Jeffrey A. Ketterling (Radiology, Weill Cornell Med., New York, NY)

Mouse models are widely used to study cardiovascular diseases due to their genetic modifiability and rapid maturation. In this study, we utilized a cardiomyocyte-specific Tafazzin knockout mouse model, which develops progressive dilated cardiomyopathy (DCM). DCM is characterized by remodeling and dilation of the left ventricle (LV) or both ventricles, leading to systolic dysfunction. Echocardiography is commonly used to assess DCM, but functional parameters such as LV size and ejection fraction only detect the condition after adverse remodeling. Strain imaging also struggles to reliably identify early DCM. Preliminary human studies suggest that intracardiac flow patterns may serve as a biomarker for early detection and progression prediction of DCM. We seek to adapt this approach to murine cardiac imaging by employing vector Doppler imaging (VDI) to quantify blood flow patterns using a high-frequency ultrasound probe (28 MHz) and high-frame-rate plane-wave imaging. The results revealed noticeable differences in velocity profiles and kinetic energy between DCM and healthy mice, particularly in the LV mid and apex during systole. This study demonstrates the potential of using VDI with mouse models to assess the relationship between intracardiac flow and DCM progression, as well as its viability as a biomarker.

2:40

5pBA6. Signal processing analysis of twinkling artifact in ultrasound Doppler imaging. Abdullah Al Masud (Dept. of Radiology, Mayo Clinic, Box 41021, Lubbock, TX 79409-1021, Abdullah-Al.Masud@ttu.edu), Benjamin Wood (Graduate School of Biomedical Sci., Mayo Clinic, Rochester, MN), Christine Lee, and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

The Doppler twinkling artifact (TA) plays a crucial role in detecting some biopsy markers including those made of polymethyl methacrylate (PMMA), facilitating precise lymph node localization in diagnostic and therapeutic procedures. This study presents comprehensive signal processing analysis methods to quantify and characterize the TA phenomenon using raw, per-channel radiofrequency (RF) data acquired from PMMA markers scanned with an L7-4 linear array transducer. We evaluated signals at different stages along the image reconstruction pipeline including (1) beamforming, (2) beamformed RF signal conversion into in-phase/quadrature (IQ) signals, and (3) Doppler velocity calculations and image reconstruction. We analyzed changes of pulses within an ensemble and between ensembles (i.e., frames). We also evaluated the impact of wall filters with various cut-off frequencies on color and power Doppler image reconstruction. Comparative analysis between PMMA markers and a non-twinkling control sample demonstrated distinct TA signatures or lack thereof in color and power Doppler images. The IQ data from PMMA marker locations exhibiting TA signatures showed substantially higher variability compared to non-twinkling samples, with standard deviations approximately 20 times higher between ensembles and 5 times higher within ensembles. This quantitative analysis framework provides valuable insight into the TA signal, potentially leading to more reliable and accurate diagnostic interpretations.

3:00–3:20 Break

3:20

5pBA7. Numerical observation of piezoelectric signals generated by ultrasound irradiation in cancellous bone covered by cortical bone layer from different directions. Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, 679-3 Nishioka, Uozumi, Akashi 674-8501, Japan, hosokawa@akashi.ac.jp)

It is known that bone fracture healing can be accelerated by irradiating low-intensity pulsed ultrasound (LIPUS). It is considered that bone formation is associated with the piezoelectricity of the bone. Therefore, in order to improve the efficiency of bone formation, it is important to sufficiently

understand the piezoelectric properties in bone. As joint bones are mostly occupied by cancellous bone formed by a porous trabecular frame, a thorough understandings of the piezoelectric properties in cancellous bone are necessary to develop the treatment for joint bones. However, the piezoelectric signal in cancellous bone is too weak to analyze. In such cases, numerical simulations are useful. Then, the piezoelectric signals generated in cancellous bone by ultrasound irradiation have been numerically simulated using a piezoelectric finite-difference time domain (PE-FDTD) method. *In situ* cancellous bone is covered by thin layer of cortical bone, and therefore, the ultrasound irradiation to cancellous bone has to be performed through the cortical bone layer. In this study, the effects of the cortical bone layer covered from the directions parallel and perpendicular to the major trabecular orientation in cancellous bone.

3:40

5pBA8. An acoustic wearable for assessment of tendon health and loading condition. Amirhossein Yazdkhasti (Orthopaedics, Cedars-Sinai Medical Ctr., 8700 Beverly Blvd., West Hollywood, CA 90069, a.h.yazdkhasti@gmail.com), Hendrik De Klerk, Andreea R. Lucaciu (Orthopaedic Surgery, Massachusetts General Hospital, Boston, MA), Joseph H. Schwab, and Hamid Ghaednia (Orthopaedics, Cedars-Sinai Medical Ctr., Los Angeles, CA)

Current methods for assessing tendon health, such as clinical examinations, imaging techniques, and implanted pressure sensors, are often subjective, expensive, and primarily effective for detecting significant damage. Advanced wearables offer a precise, cost-effective, and non-invasive alternative. In this study, we propose an acoustic-based wearable device designed to estimate tendon load and predict damage severity in both deep and superficial tendons. The wearable comprises an array of acoustic transducers arranged in a cuff around the targeted body area. One transducer generates acoustic waves that propagate through various tissues, where their behavior is influenced by the mechanical and geometric properties of each tissue. By analyzing these interactions, we measure tendon strains and detect defects. The load estimation capability of the proposed wearable was tested using 18 porcine flexor digitorum tendons from cadaveric pig legs, while its diagnostic capability was evaluated by inducing three types of defects in the tendons. Results indicate that the acoustic wearable demonstrates robust performance in estimating the force applied to tendons and effectively compares tendon health conditions, accurately predicting the type of damage.

4:00

5pBA9. Egg white-based blood mimicking fluid for therapeutic ultrasound device evaluation. Yunbo Liu (FDA, 10903 New Hampshire Ave., Silver Spring, MD 20993, yunbo.liu@fda.hhs.gov) and Subha Maruvada (U.S. Food and Drug Administration, Silver Spring, MD)

An egg white-based blood mimicking fluid (BMF) was developed and characterized as a blood coagulation surrogate for the acoustical and thermal evaluation of therapeutic ultrasound, especially high intensity thermal ablation and histotripsy devices. Physical properties, including coagulation temperature, frequency-dependent attenuation, sound speed, thermal conductivity, and thermal diffusivity, were measured as a function of temperature (20–95 °C). The fluid viscosity was quantified at room and body temperature. With the addition of Nylon particles in the solution, the backscattering coefficient of the BMF was quantified before and after a complete thermal coagulation. For a 30 s thermal exposure, the egg white-based BMF (3 mm thickness) started to denature at 65 °C and coagulate into an elastic gel at 85 °C. The coagulation temperature can be lowered by adding a small amount of acid solution to the BMF. The temperature-dependent ultrasound attenuation and other physical parameters were found to be similar to the reported values of human blood. These properties make this egg white-based blood mimicking fluid a useful tool for pre-clinical bench testing of therapeutic ultrasound devices.

4:20

5pBA10. Acoustic tweezing coagulometry: Noncontact coagulation analysis of a drop of blood. Huy Q. Pham (Dept. of Biomedical Eng., Tulane Univ., New Orleans, LA), Elizabeth M. Cummins (Dept. of Biomedical Eng., Tulane Univ., New Orleans, LA), Nithya Kasireddy (Levisonics, Inc., Fishers, IN), Karen F. Bruzdoski, Vadim V. Kostousov, Jun X. Teruya (Div. of Transfusion Medicine & Coagulation, Texas Children's Hospital, Houston, TX), and Damir B. Khismatullin (Dept. of Biomed. Eng., Tulane Univ., 6823 St. Charles Ave., New Orleans, LA 70118, damir@tulane.edu)

Acoustic tweezing coagulometry provides containerless coagulation analysis using a small, 4–6 microliter drop of blood. It operates in two regimes: integrated quasi-static acoustic tweezing thromboelastometry (i-QATT), which measures clot firmness and turbidity, and acoustic tweezing spectroscopy (ATS) assessing clot viscoelasticity. In this work, these two techniques were clinically assessed on venous whole blood (WB) and platelet-poor plasma (PPP) samples from non-smoking healthy adult volunteers, smoking adult volunteers, pediatric patients with hemophilia A, and pediatric patients on ECMO/hemodialysis. The results of this study indicate that the values of i-QATT parameters MCF (maximum clot firmness), CIT (clot initiation time), and RT (reaction time) were strongly correlated with fibrinogen concentration and lab values obtained by gold-standard plasma tests (aPTT, PT) and viscoelastic hemostatic assays (ROTEM). Reference ranges for these parameters ruled out abnormal coagulation caused by factor deficiency, heparin, antiplatelet agents (Cytochalasin-D), or fibrinogen spiking. In the ATS technique, RT measured from the WB viscosity data demonstrated a strong correlation with plasma PT/aPTT. Linear correlation was obtained between the WB MCF output and the fibrinogen concentration. Acoustic tweezing coagulometry enables comprehensive drop-of-blood coagulation analysis that rapidly detects hypo- and hyper-coagulable states and measures PT, aPTT, and fibrinogen in WB or hemolyzed plasma samples.

Session 5pMU

Musical Acoustics, Architectural Acoustics, Noise, and Psychological and Physiological Acoustics: Discrimination Tests: Methodologies and Applications

Andrew A. Piacsek, Cochair
Physics, Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422

Claudia Fritz, Cochair
Central Washington Univ., Ellensburg, WA 98926-7422

Chair's Introduction—12:55

Invited Papers

1:00

5pMU1. Perception of timbral changes in a violin due to humidity exposure. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu) and Claudia Fritz (Institut Jean le Rond d'Alembert, Sorbonne Univ., Paris, France)

Understanding the sensitivity with which experienced or casual listeners can discern the timbral qualities of violin family instruments is important for the construction and marketing of these instruments. An important tool for investigating such sensitivity is the listening discrimination test, which requires subjects to listen to multiple sounds and identify those that are the same or different. The present study uses listening discrimination tests to investigate the ability of people to reliably distinguish the effect of water absorption from humidity exposure on the sound of a violin. Audio files used in the test were synthesised by convolving the bridge force signal from a played violin with the measured bridge admittance of different violins that were placed in an environmental chamber for 24 h at each humidity level. Absorbed water in the wood of the violin body produces measurable shifts in the frequencies of signature modes. The goal of the study is to quantify the amount of shift that is perceptible in a controlled listening environment. Results of the study will be discussed, along with implications for discerning changes in violin response due to other causes.

1:20

5pMU2. Musical practice in audio augmented reality: Testing virtual acoustics using reverb convolution via bone conduction headphones. Andrea Gozzi (École de musique, Université de Sherbrooke, 6982, Ave. de Chateaubriand, Montreal, QC H2S 2P1, Canada, gozzi.andrea@umontreal.ca), Gianluca Grazioli (Schulich School of Music, McGill Univ., Montreal, QC, Canada), Dominic Thibault (Faculté de musique, Université de Montréal, Montréal, QC, Canada), Martha de Francisco (Schulich School of Music, McGill Univ., Montréal, QC, Canada), and Alessandro Braga (École de technologie supérieure, Montréal, QC, Canada)

Bone conduction headphones primarily transmit the audio signal directly to the inner ear. By not obstructing the ear canal, the system facilitates the perception of two layers of sound: a seamless integration of unmediated and virtual sound, enhancing the realism of the latter in an audio augmented reality experience. Our study aims to evaluate the impact of virtual acoustics on the performance of professional musicians. This paper mainly investigates the effect of a real-time convolution-based system that convolves sounds produced by musicians and delivers it to them via bone conduction headphones during musical performance. We investigate the impact of this system on musical practices and compare it to four other practice conditions: (1) an acoustically treated studio, (2) reverb convolution via traditional air conduction headphones, (3) reverb convolution via a loudspeaker-based virtual acoustic system, and (4) a real concert hall, where impulse responses were used for auralization in the virtual acoustics simulation. Data collected from musicians during both ensemble and solo performances—including EEG readings, performance analysis from audiovisual recordings, and surveys—provide, in our knowledge, insights never been conducted before on this research topic.

1:40–2:00 Break

2:00

5pMU3. Loudness-based noise metrics for annoyance to rotorcraft noise using Thurstone analysis. Matthew Boucher (NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, matthew.a.boucher@nasa.gov)

Annoyance to rotorcraft noise continues to be an area of concern to communities, and the question remains for researchers and regulators regarding which noise metrics best capture the perception to these types of vehicles. Traditional metrics based on A-weighting have not been supplanted, but metrics based on loudness (ISO 532-1) may offer improvements since they include signal processing closer to the human auditory system. In addition to loudness, sound quality metrics (sharpness, tonality, etc.) have also been found to

correlate with annoyance. Using data collected in recent psychoacoustic tests on helicopter and urban air mobility vehicle noise, annoyance scales using Thurstone's Law of Comparative Judgment account for the relationship between perception, which is an internal judgment, and the external perceptual response. This work also investigates correlations between noise metrics and annoyance for traditional noise metrics as well as for loudness-based metrics, including psychoacoustic annoyance models (e.g., Zwicker). This leads to a psychoacoustic annoyance model for urban air mobility vehicle noise that correlates better with annoyance than Zwicker's model. Results also show that loudness-based metrics, integrated over time similarly to sound exposure level, are highly correlated with annoyance. Finally, Thurstone's scale yields just-noticeable-differences in noise metrics for annoyance to rotorcraft noise.

2:20–2:50 Panel Discussion

FRIDAY AFTERNOON, 23 MAY 2025 GALERIE 5, 1:00 P.M. TO 1:40 P.M.

Session 5pNS

Noise: General Topics in Noise: Measurement and Processing II

S. Hales Swift, Chair
Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082

Contributed Papers

1:00

5pNS1. Application of UAVs for low-frequency outdoor sound measurements. Ramesh Raja Subramanyam, Criostoir Gerasch, Rafael Castro Mota, Volker Wittstock, and Stefan Jacob (National Metrology Inst., Germany, Bundesalle, 100, Braunschweig, Germany, stefan.jacob@ptb.de)

A comprehensive multi-point sensor network is required to measure low frequency noise in atmosphere with respect to emissions from renewable energy sources and seismic activity monitoring. Uncrewed Aerial Vehicles (UAVs) offer a solution as sensor carrier for measuring outdoor noise. The feasibility of employing UAVs in low frequency acoustic measurement is examined here. Noise signature of different drones is measured in an anechoic chamber and compared with the atmospheric background noise. Frequencies below the first rotor tone had minimal disturbance from the inherent noise of UAVs. Directivity study with an automized sensor array revealed that sensors mounted in the rotor plane are least affected by the rotor noise. An infrasound reference source working on the principle of a quarter wave resonator is then used for *in situ* calibration of a microphone mounted on a UAV. Results of sound pressure sampled at multiple points with the UAV were in level with the ground-based measurements. Furthermore, a wind shield array is designed for filtering the noise caused by atmospheric turbulence, which is critical in the low frequencies. Several configurations of wind shields are tested to increase the frequency range of the measurement and to minimize internal resonance effects of the wind shield.

1:20

5pNS2. Complex surface impedances and spherical wave approximations in image source models for outdoor auralization. Jonas Heck (Insitute for Hearing Technol. and Acoust., RWTH Aachen Univ., Kopernikusstrasse 5, Aachen 52074, Germany, jonas.heck@akustik.rwth-aachen.de), Ronny Roos (Insitute for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany), Josep Llorca-Bofi (Fraunhofer Inst. for Bldg., Phys. IBP, Aachen, Germany), and Michael Vorlaender (Insitute for Hearing Technol. and Acoust., RWTH Aachen Univ., Aachen, Germany)

Auralization of different traffic scenarios enables urban planners to predict the effects of soundscape interventions. For efficient calculation of the required impulse responses, geometrical acoustic methods are widely used, particularly in room acoustics simulation. For outdoor applications, however, it has been shown that the precise modeling of reflections is more critical than in room acoustics since the reflection density usually is significantly lower, so that the individual reflections have a more specific impact. This study incorporates complex surface impedances and the Weyl-Van der Pol equation to approximate spherical-wave reflections into the filter generation process of the image source model. The transfer functions generated by direct sound and reflected sound are compared to those which use real-valued, random-incidence absorption coefficients as typically used in geometrical acoustic applications. The improvement of the filter accuracy is discussed on the basis of reference measurements and in the context of traffic noise auralization.

1:40

5pNS3. Abstract withdrawn.

Session 5pPAa

Physical Acoustics: General Topics in Physical Acoustics III

Robert Lirette, Cochair
Commun. Technol. Lab., National Institute of Standards and Technology, 325 Broadway, MS67201, Boulder, CO 80305

Contributed Papers

1:00

5pPAa1. Does backscattering from a solid prolate spheroid in water exhibit the axial focusing of glory scattering? Heather A. Moon (Phys. & Astronomy, Washington State Univ., Webster Hall, 100 Dairy Rd. Rm. 1245, Pullman, WA 99164-2814, heatherannettemoon@gmail.com) and Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA)

Many solid spheres in water illuminated by ultrasound display enhanced backscattering partially analogous to the optical glory of cloud droplets. The enhancement associated with glory scattering is axially focused. In the acoustical case, one of the mechanisms for producing the needed outgoing toroidal wavefront is the launching and guiding of a leaky Rayleigh-like wave around the surface of the sphere [K. L. Williams and P. L. Marston, J. Acoust. Soc. Am. 78, 722–728 (1985)]. A related backscattering enhancement was also demonstrated for spherical shells [S. G. Kargl and P. L. Marston, J. Acoust. Soc. Am. 85, 1014–1028 (1989)]. In the present research, we demonstrate that a similar backscattering enhancement is present for axisymmetric ultrasonic illumination of a solid prolate brass spheroid. By using tone burst illumination, the axially focused contribution to the scattering is distinguishable from the specular reflection contribution to the scattering. The evidence for the Rayleigh wave mechanism is seen in the timing of this contribution: experimental and model predictions are consistent and through bistatic measurements of the variation in its strength as scattering angle changes. This research is relevant to surface-guided ray theory extended to variable curvature objects in water. [Work supported by the Office of Naval Research.]

1:20

5pPAa2. Blind-label acoustic subwavelength imaging with static random scatterers. Jinuan Lin (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Rm. 3533, Madison, WI 53706, jlin328@wisc.edu) and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

The resolution of conventional acoustic sensing and imaging systems suffers from the diffraction limit. Most of existing technologies addressing this limit require controlled “labels,” i.e., metamaterials or contrast agents, to be deposited close to the objects and to either remain static or be tracked precisely during imaging, restricting their practical applications. We proposed a “blind-label” approach for acoustic subwavelength imaging. In our previous work, the blind labels are randomly distributed microspheres that move freely around the objects. The originally evanescent components in the scattered waves from the object are first converted to propagating components and then extracted by advanced computational algorithms. In this work, we extend the applicable scenarios of our previous work by exploiting static random scatterers as the blind labels instead of moving ones. We conduct experiments to detect wires with subwavelength separations buried in silicone polymer, with a thin layer of microspheres deposited around the wires as the blind labels. Measurements for reconstruction are taken by exciting plane waves of different incident angles. Our subwavelength imaging system can now be applied to various scenarios where intrinsic random scattering media is accessible, such as detecting cracks buried in inhomogeneous structures or small lesions embedded in tissue.

1:40

5pPAa3. Analytical solutions for acoustic vortex beam radiation from planar and spherically focused circular pistons. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Appl. Res. Lab., P.O. Box 9767, Austin, TX 78766-9767, chiragokani@gmail.com), Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Acoustic vortex beams are quasiplanar waves with helical wavefronts characterized by the orbital number ℓ. Although Gaussian amplitude distributions result in closed-form analytical solutions for the entire paraxial field [Gokani *et al.*, J. Acoust. Soc. Am. 155, 2707–2723 (2024)], acoustic vortex beams are usually radiated by sources with a uniform circular amplitude distribution. In this talk, analytical solutions for the field radiated by unfocused and focused uniform circular vortex sources of radius a are derived. Evaluation of the Fresnel diffraction integral in the far field of an unfocused source and in the focal plane of a focused source leads to a solution in terms of an infinite series of Bessel functions for $\ell > -2$. By calculating the first local maximum of this solution, it is found that the vortex ring radius is $r_\ell = \xi_\ell z/ka$ in the far field of an unfocused source and $r_\ell = \xi_\ell d/ka$ in the focal plane of a focused source with focal distance d, where $\xi_\ell = 1.23\ell + 1.49$ and k is the wavenumber. The solution given by the infinite series is reduced to closed forms for $0 \le \ell \le 4$, corresponding to orbital numbers commonly used in experiments. [CAG supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:00

5pPAa4. Comparison of finite-amplitude acoustic models: Single equation of motion for potential versus system-level approaches. Mohammad Javad Mohaghegh Kojidi (Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148-3520, mmohaghe@uno.edu) and Ashok Puri (Univ. of New Orleans, New Orleans, LA)

This study evaluates two finite-amplitude acoustic models, by solving the associated equations of motion (in potential form), and then compares the results with recent studies using hyperbolic system-level approaches. An implicit numerical method is employed to solve the second-order equations in one dimension. The performances of the Diaz *et al.* (2018) and Blackstock (1963) models in approximating the acoustic special case of the Euler system, via velocity profile plots and related metrics, are evaluated and compared. Working in the setting of the classical signaling problem with sinusoidal input, present results (using the single equation of motion approach) provide independent confirmation of earlier findings (based on the hyperbolic system-level approach) that the Diaz *et al.* model outperforms Blackstock’s.

2:20

5pPAa5. Abstract withdrawn.

2:40–3:00 Break

3:00

5pPAa6. Ultrasonically modified nonlinear dielectric spectra for liquids. Robert Lirette (Communications Technol. Lab., National Inst. of Standards and Technol., 325 Broadway, MS67201, Boulder, CO 80305, robert.lirette@nist.gov), Tomasz Karpisz (Communications Technol. Lab., National Inst. of Standards and Technol., Boulder, CO), Malgorzata Musial (Mater. Measurement Lab., National Inst. of Standards and Technol., Boulder, CO), Aaron Hagerstrom (Communications Technol. Lab., National Inst. of Standards and Technol., Boulder, CO), Jason A. Widegren (Mater. Measurement Lab., National Inst. of Standards and Technol., Boulder, CO), Nathan Orloff, and Angela C. Stelson (Communications Technol. Lab., National Inst. of Standards and Technol., Boulder, CO)

Acoustic waves can modify the dielectric spectra of a material by modulating its complex permittivity. This effect has been widely used in acousto-optic modulation devices to modulate light. Previously, we have shown the possibility for ultrasound to modify microwave electric signals inside a material using nonlinear vector network analyzer measurements. Our experiment consists of a liquid channel above an electric co-planar waveguide. An ultrasonic transducer acoustically excites the liquid, modulating its permittivity. We then measured the nonlinear dielectric spectra of the liquid at the sum and difference of the electric and acoustic frequencies. Using finite element models of the electrical properties of our system, we can de-embed our data and obtain the nonlinear contributions to the permittivity. Here, we present preliminary results for several different liquids, each showing a distinct nonlinear spectrum. Through these measurements, we aim to quantify the electrostrictive properties of fluids and detect intermolecular species and interactions through their nonlinear spectra.

3:20

5pPAa7. Baseline correction of transfer matrix-based standing wave tube measurements. Carson Willey (AFRL, 35 Hawthorne Glen Trl, Beavercreek, OH 45440, willeycl@gmail.com), Vincent Chen, and Abigail T. Juhl (AFRL, Wright Patterson Air Force Base, OH)

Material property characterization by audible normal incidence acoustic waves has been under development since the early 1980s. Initially, reflection-based measurements were investigated by the General Motors Corporation, later termed impedance tube testing. In the late 1990s, standing wave tube testing was developed using the transfer-matrix method and a commercially available impedance tube setup. Since then, this method has been standardized in ASTM E2611-17. The standing wave tube evaluation can give measurements of effective material or acoustic properties of samples or acoustic elements placed at the sample position. This method has been applied widely since its creation to great success; however, the experimental apparatus assumes that all segments are ideal (i.e. no broken seals, walls are rigid, etc.). In this talk, a method for baseline correction of a standing wave tube test, containing non-ideal segments, is presented. The procedure is based on the transfer matrix method and is, thus, compatible with standard standing wave tube theory. Specifically, effective transfer matrices are experimentally determined by additional measurements of the empty tube (i.e., the baseline system) and used to compensate for the errors that appear in the test derived measurements (e.g., transmission loss, density, and transmission/reflection coefficients).

3:40

5pPAa8. Localizing sound sources in air within built environments using ground-coupled airwaves. Samba Gaye (SEAS, Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, samba.gaye@udc.edu) and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, the feasibility of using ground-coupled airwaves for localizing and near-surface sound sources in built-up environments is determined. Of particular interest are the localization of gunshots and explosive sound sources in the air. The ground-coupled airwaves from an array of vertical geophones are used to estimate the sound source positions. Time-difference of arrival (TDOA) and array beamforming localization methods are investigated to overcome the built environment's multipath effects on the acoustic-seismic coupling, which makes it difficult to detect the ground-coupled airwaves.

4:00

5pPAa9. Comparing the Blackstock (1963) and Diaz–Solovchuk–Sheu (2018) acoustic models: Which best approximates the Euler and Navier–Stokes systems? Pedro M. Jordan (Acoust. Div., U.S. Naval Res. Lab., U.S. Naval Res. Lab., Acoust. Div., Stennis Space Ctr., MS 39529, pedro.m.jordan.civ@us.navy.mil) and Nicolas Valdivia (Acoust. Div., U.S. Naval Res. Lab., Washington, DC)

The propagation of weakly nonlinear sound waves in both lossless and viscous gases is considered under the 1-D versions of the Blackstock and Diaz–Solovchuk–Sheu (DSS) models. Employing both analytical and numerical methods, we seek to determine which of these two, competing, finite-amplitude models best approximates the irrotational special cases of the Euler and Navier–Stokes systems. In the Euler case, we work in the context of a "start-up" signaling-type problem involving sinusoidal boundary input. Employing finite-differencing methods and results from simple wave theory, we perform comparisons using metrics based on the velocity field. We show that, while the differences are small, the simpler DSS model outperforms Blackstock's. In the Navier–Stokes case, we work in the context of the piston problem on the real line, and thus, can invoke the assumption of (right-running) traveling waves. It is shown that the long-established Blackstock model, the finite-amplitude model with the fewest approximations, and the newer, but more heavily approximated, DSS model yield the same "tanh-type" traveling wave profile. Finally, charts illustrating the connections that exist between the current collection of finite-amplitude equations, and the degree of approximation required to derive each, are presented. [Work supported by ONR funding.]

4:20

5pPAa10. Surface impedance measurements of rough seas. Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, Washington, DC 20064, turo@cua.edu), Andrea Vecchiotti, Matthew Stengrim, Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), and Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents measurements of the effective surface impedance of rough water performed in the maneuvering and seakeeping basin (MASK) at the Naval Surface Warfare Center (Carderock Division, Bethesda MD). The ANSI/ASA S1 18 was implemented to measure the effective surface impedance of sea states 1, 2 and 3. An omnidirectional source and two-phase matched microphones are used to implement this method. The source generates spherical waves in the range of [250 5000] Hz. The MASK make use of 216 actuators distributed on two sides of a 109.7x73.2 m pool to generate sea states with a specified Pierson-Moskowitz spectrum. Wave time history and spectrum are recorded by an array of ultrasonic sensors placed above the water away from the experimental setup. This helps to verify that the designed sea state is indeed achieved. Cosine-windowed tones logarithmically distributed in the frequency range of interest were used as acoustic signals. Recorded signals were band-pass filtered at around the excitation frequency and cropped to extract the exact level at the excitation frequency. Results obtained using the single-parameter model recommended by the measurement standard show that the effective flow resistivity decreases with the increase of the sea state. [Work supported by ONR N00014-23-1-2352.]

4:40

5pPAa11. Simulated free field measurements for active cancellation of noise entering a window. Robert Nelson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, rwn5136@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Environmental noise propagating through an open window may be cancelled by a sparse array of loudspeakers placed in the window frame. This approach towards global noise cancellation mitigates noise pollution at its source while retaining the light and ventilation received from an open window. To perform global noise cancellation measurements in a non-anechoic environment, a simulated free field approach was developed. Using repeated randomized stepped sine waves, where each frequency has an integer number of wavelengths, discrete frequencies are time gated to negate reflections and then are concatenated together to yield a pure tone. With a dense concentration of discrete frequencies, intermittent data may be interpolated via

a spline interpolation. Results for this approach will be presented in two parts: First, the transfer functions of individual loudspeakers are measured and are compared to conventional measurements performed in an anechoic chamber. Second, the global noise cancellation produced by the sparse loudspeaker array was measured with and without the time gating approach to determine the effectiveness of the method.

5:00

5pPAa12. Emerging topics in aeroacoustics: Interdisciplinary perspectives. John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Holmes 302, 2540 Dole St., Honolulu, HI 96822, alleniii@hawaii.edu)

Recent studies on use of drone self-noise for improved control algorithms and sound generated by flapping flight of insects are presented in dedication. The flapping flight studies encompasses experimental and computational investigations of invasive insect species Coconut Rhinoceros Beetle (Oryctes rhinoceros) and the Oriental Flower Beetle (Protaetia orientalis). Three dimensional, unsteady flow simulations were conducted using compressible flow software (CAESIM, Adaptive Research, Inc.) with a TVD methodology. Flapping wing motion uses mesh deformation with a rotation with a prescribed bending coupled to rotation and translation of the wing's hinge position. Sound generation by trailing edge and tip vortices is discussed with respect to second harmonic sounds. Benchmarks and validations of sound propagation are highlighted.

FRIDAY AFTERNOON, 23 MAY 2025 GALERIE 1, 1:00 P.M. TO 5:20 P.M.

Session 5pPAb

Physical Acoustics, Engineering Acoustics and Structural Acoustics and Vibration: Topological Aspects of Acoustic Waves

Georgios Theocharis, Chair
CNRS-LAUM, Le Mans Université Av. Olivier Messiaen, Le Mans, 72000, France

Vincent Pagneux, Cochair
CNRS, LAUM - Le Mans Université, av. O. Messiaen, Le Mans, 72085, France

Vassos Achilleos, Cochair
CNRS-LAUM, Av. Olivier Messiaen, 72085 Le Mans cedex 9, France, Le Mans, 72085, France

Invited Papers

1:00

5pPAb1. Su-Schrieffer-Heeger model in Nano-opto-electro-mechanical device for high optomechanical coupling rate. Yan Pennec (iemn, IEMN - Université de Lille Sci. et Technologies, Villeneuve d'Asq 59652, France, yan.pennec@univ-lille.fr), Abdellatif Gueddida, Gaetan Leveque, and Bahram Djafari-Rouhani (iemn, Villeneuve d'Asq, France)

The Nano-Opto-Electro-Mechanical device consists of an optomechanical (OM) cavity within a phoXonic nanobeam, connected to inter-digitated transducers (IDTs) at both ends to excite and detect acoustic waves. The OM cavity's optical modes are activated by an external waveguide ending in a Bragg mirror made from tapered holes. The phoXonic cavity, designed with periodic holes and stubs, generates both an acoustic bandgap in the GHz range and an optical bandgap around 1550 nm for telecommunications. The device supports acoustic modes from 2 to 4 GHz and optical modes with high-quality factors, demonstrating a substantial optomechanical coupling rate (g) on the MHz scale. We also explored phononic cavity modes using IDTs placed in front of the nanobeam. In this work, we investigate high-frequency cavities in the 6–10 GHz range. To achieve this, the design of OM cavities with localized phonon modes is modified by introducing long-length stubs to adjust the frequency and enhance OM coupling. These innovations build on previous designs that controlled phonon frequency by altering hole and stub sizes. Additionally, the study explores topological effects, using the Su-Schrieffer-Heeger (SSH) model to create simultaneous localized acoustic and optical modes at the interface between two different phoXonic crystals, resulting in both photonic and phononic topological interface modes.

1:20

5pPAb2. Topological valley phononic crystals in surface acoustic wave microfluidics. Chen Shen (Dept. of Mech. Eng., Rowan Univ., 201 Mullica Hill Rd., Glassboro, NJ 08028, shenc@rowan.edu), Shuaiguo Zhao, Zhenhua Tian (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), Steven Cummer (Dept. of Elec. and Comput. Eng., Duke Univ., Durham, NC), and Tony Huang (Thomas Lord Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Recent years have witnessed the surge of topological wave phenomena as a versatile platform to engineer exotic wave energy transport which is robust to defects and disorders. Most demonstrations in acoustics remain in a single phase of matter such as solid or air. Here, we introduce the realization of valley phononic crystals for surface acoustic waves and their interaction with fluids in an acoustofluidic setup. It is shown that the interplay between megahertz elastic waves and hydrodynamics where two phases of materials are involved offers rich physics and new engineering potentials of topological matter. By electroplating hexagonal copper pillars on a lithium niobate substrate and adding a liquid layer on top of it, the excited elastic valley spin is transferred at the interface of solid-fluid domains. The interactions lead to valley streaming vortices in the fluid domain that support backward-immune particle transport. In addition, it is found that pressure wells are formed around the small pillars, which enable the concentration of DNA molecules in the nm size range. The studies may open new avenues for applying topological acoustic waves in particle manipulation and life sciences.

1:40

5pPAb3. New tools to shape topological boundary states: Hidden symmetry and nonlinearity. K. Prabith, Udbhav Vishwakarma, Murthaza Irfan (Aerosp. Eng., Indian Inst. of Sci., Bangalore, India), Georgios Theocharis (LAUM, CNRS-UMR 6613, Le Mans Univ., Le Mans, France), and Rajesh Chaunsali (Aerosp. Eng., Indian Inst. of Sci., Indian Inst. of Sci., Bangalore 560012, India, rchaunsali@iisc.ac.in)

Topological insulators have generated renewed interest in robust wave propagation across various physical systems. In this talk, we introduce two powerful tools—hidden symmetry and nonlinearity—that can shape novel topological boundary states in mechanical lattices. First, we explore spinner dimer lattices, where hidden chiral symmetry protects topological edge states with distinct localization profiles at opposite ends. Then, we investigate how introducing nonlinearity into a Kagome lattice allows for the tuning of higher-order topological states' frequency and stability, unveiling new families of states absent in linear systems. Together, these tools offer fresh insight into controlling the shape and stability of topological boundary states.

2:00

5pPAb4. Observation of mechanical kink control and generation via phonons. Kai Qian (Dept. of Mech. and Aerosp. Eng., Univ. of California, San Diego, San Diego, CA, k3qian@ucsd.edu), Nan Cheng (Dept. of Phys., Univ. of Michigan, Ann Arbor, MI), Francesco Serafin, Kai Sun (Dept. of Phys., Univ. of Michigan, Ann Arbor, MI), Georgios Theocharis (CNRS-LAUM, Le Mans, France), Xiaoming Mao (Phys., Univ. of Michigan, Ann Arbor, MI), and Nicholas Boechler (Dept. of Mech. and Aerosp. Eng., Univ. of California, San Diego, San Diego, CA)

Kinks are localized transitions between distinct ground states that are associated with a topological charge, significant in fields ranging from condensed matter to cosmology. Theoretical and computational studies have shown that phonons can interact with mechanical kinks to trigger their motion, offering a pathway for kink control. However, the discreteness of most kink-supporting systems introduces a Peierls-Nabarro (PN) barrier, requiring extra energy for kinks to move locally. Here, we report the first experimental observation of mechanical kink control and generation via phonons. To achieve this, we create an elastically-coupled realization of the Kane-Lubensky chain model, which supports a single, topologically protected kink that requires zero energy to deform quasi-statically, resulting in a zero PN barrier. In addition to finding strong agreement between our experiments and numerical simulations, we also numerically observe unique kink dynamics distinct from other nonlinear discrete systems, including continuously smoothly varying set of kink solutions between the neighboring onsite-centered kinks (and associated "shape" mode evolution) and long-duration kink motion, the latter of which has important implications for kink control. Given the topological polarization of our system and the associated features, this work has implications for remote material stiffness control, locomotion, shape-shifting materials, as well as robust signal processing and transmission.

2:20

5pPAb5. Breakdown of conventional winding number calculation in Su-Schrieffer-Heeger Lattices. Amir Rajabpoor Alisepahi (Mech. Eng., Univ. of Vermont, Burlington, VT), Siddhartha Sarkar, Kai Sun (Phys., Univ. of Michigan, Ann Arbor, MI), and Jihong A. Ma (Mech. Eng., Univ. of Vermont, 33 Colchester Ave., Votey 201B, Burlington, VT 05405, Jihong.Ma@uvm.edu)

Topological insulators promise to realize exotic quantum phenomena in electronic, photonic, and phononic systems. Conventionally, topological indices, such as winding numbers, have been used to predict the number of topologically protected domain-wall states (TPDWSs) in topological insulators, a signature of the topological phenomenon called bulk-edge correspondence. In this talk, I will demonstrate theoretically and experimentally that the number of TPDWSs in a mechanical Su-Schrieffer-Heeger (SSH) model can be higher than the winding number depending on the strengths of beyond-nearest-neighbor interactions, revealing the breakdown of the winding number prediction. Alternatively, we resort to the Berry connection to accurately characterize the number and spatial features of TPDWSs in SSH systems, further confirmed by the Jackiw-Rebbi theory proving that the multiple TPDWSs correspond to the bulk Dirac cones. Our findings deepen the understanding of complex network dynamics and offer a generalized paradigm for precise TPDWS prediction in potential applications involving localized vibrations, such as drug delivery and quantum computing.

2:40–3:00 Break

3:00

5pPAb6. Backscattering-free edge states below all bands in two-dimensional auxetic media. Wenting Cheng (Phys., Univ. of Michigan, 1811 Willowtree Ln., Ann Arbor, MI 48105, cwenting@umich.edu), Kai Qian (Mech. and Aerosp. Eng., Univ. of California San Diego, San Diego, CA), Nan Cheng (Phys., Univ. of Michigan, Ann Arbor, MI), Nicholas Boechler (Mech. and Aerosp. Eng., Univ. of California San Diego, San Diego, CA), Xiaoming Mao, and Kai Sun (Phys., Univ. of Michigan, Ann Arbor, MI)

Unidirectional and backscattering-free propagation of sound waves is of fundamental interest in physics and highly sought-after in engineering. Current strategies utilize topologically protected chiral edge modes in bandgaps, or complex mechanisms involving active constituents or nonlinearity. Here, we propose a new class of passive, linear, one-way edge states based on spin-momentum locking of Rayleigh waves in two-dimensional media in the limit of vanishing bulk to shear modulus ratio, which provides perfect unidirectional and backscattering-free edge propagation that is immune to any edge roughness and has no limitation on its frequency (instead of residing in gaps between bulk bands). We further show that such modes are characterized by a new topological winding number that protects the linear momentum of the wave along the edge. These passive and backscattering-free edge waves have the potential to enable a new class of phononic devices in the form of lattices or continua that work in previously inaccessible frequency ranges.

3:20

5pPAb7. The acoustic Dirac equation as a model of topological insulators. Pierre A. Deymier (Mater. Sci. and Eng. New Frontiers of Sound Sci. and Technol. Ctr., Univ. of Arizona, 1235 E. James E. Rogers Way, Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ 85721, deymier@arizona.edu), Keith Runge (Mater. Sci. and Eng. New Frontiers of Sound Sci. and Technol. Ctr., Univ. of Arizona, Tucson, AZ), and Abhirup Basu (Mathematics New Frontiers of Sound Sci. and Technol., Ctr., Univ. of Arizona, Tucson, AZ)

The dynamical equations of motion of a discrete one-dimensional harmonic chain with side restoring forces is analogous to the relativistic Klein-Gordon equation. Dirac factorization of that discrete Klein-Gordon equation introduces two equations with time reversal (T) and parity (P) symmetry breaking conditions. The Dirac-factored equations enable the exploration of the properties of the solutions of the dynamical equations under P and T symmetry breaking conditions. The spinor solutions of the Dirac factored equations describe two types of acoustic waves, one with a conventional topology (Berry phase equal to 0) and the other one with a non-conventional topology (Berry phase of π). In this latter case, the acoustic wave is isomorphic to the quantum spin of an electron, also known as an acoustic pseudospin, which requires a closed path corresponding to two Brillouin zones to recover the original spinor. The interface between topologically conventional and non-conventional chains supports topological surface states. The Dirac-factored equations of motions of the one-dimensional harmonic chain with side springs can serve as a model for the investigation of the properties of acoustic topological insulators. [Work supported by NSF Award No. 2242925.]

3:40

5pPAb8. Experimental realization of higher-dimensional non-Hermitian acoustic lattices. Yun Jing (Grad. Prog. in Acoust., Pennsylvania, 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu) and Jiaxin Zhong (Grad. Prog. in Acoust., Pennsylvania, State College, PA)

We designed and implemented an active acoustic crystal to experimentally realize higher-dimensional non-Hermitian lattices. This platform enables precise tuning of both onsite potentials and hoppings using active components comprising microphones, loudspeakers, and a centralized digital controller. As a first demonstration, we implemented a nonreciprocal non-Hermitian Kagome lattice and used both real and complex frequency excitations to observe the higher-order non-Hermitian skin effect. In the topologically nontrivial phase, acoustic energy was observed to localize at a corner of the lattice, even when the source was positioned far from that corner. Subsequently, we realized both nonreciprocal and reciprocal two-dimensional single-band non-Hermitian lattices. Through direct measurements of energy spectra and eigenstates, we experimentally confirmed that energy spectra are highly sensitive to boundary conditions in both cases but are influenced by lattice geometry only in reciprocal systems. Moreover, we observed significant skewness between the left and right eigenstates in nonreciprocal lattices, while in reciprocal lattices, the eigenstates were nearly identical. These experimental results align well with theoretical predictions, validating the effectiveness of our platform. This versatile experimental setup holds broad potential for advancing studies in non-Hermitian physics and exploring uncharted theoretical predictions.

4:00

5pPAb9. The analogue of the Hatano-Nelson model in active acoustic lattices. Vassos Achilleos (CNRS, Av. Olivier Messiaen, 72085 Le Mans cedex 9, France, Le Mans 72085, France, vassosnbi@gmail.com), Vincent Pagneux, Yves Aure?gan (CNRS, Le Mans, France), Guillaume Penelet, and Anis Maddi (Laboratoire d'Acoustique de l'Universite ´ du Mans (LAUM), UMR 6613, Institut d'Acoustique - Graduate School (IA-GS), CNRS, Le Mans Universite ´, France., Le Mans, France)

In this work, we show how an exact mapping to the Hatano-Nelson model can be achieved using acoustic waveguides with periodically arranged active (electro-acoustic) elements. Due to the exact mapping, we are able to exhibit experimentally the interesting physics predicted by the model, i.e., the skin-modes, the formation of a loop-like spectrum in the case of periodic boundaries, and finally the exponential sensitivity to boundary conditions with respect to the number of unit cells. Extensions of the model to more complex scenario, especially the extension to a non-reciprocal topological 1-D lattice are also discussed. Among various implementations already existing in the literature, we believe that our method is advantageous since it is broadband (no resonances needed to achieve the discrete model) and can be made stable even with periodic boundaries which constitute an important element in the study of non-Hermitian periodic systems.

4:20

5pPAb10. Group velocity control in PT-symmetric acoustic systems. Kangkang Wang (Inst. of Acoust., Nanjing Univ., Nanjing, China), Felix Langfeldt (Univ. of Southampton, Southampton, United Kingdom), Chen Shen (Dept. of Mech. Eng., Rowan Univ., Glassboro, NJ), Haishan Zou (Inst. of Acoust., Nanjing Univ., Nanjing, China), Sipei Zhao (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, Botany, New South Wales, Australia), Jing Lu (Inst. of Acoust., Nanjing Univ., Nanjing, China), and Lea Beilkin (Mech. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, leabeilkin@tauex.tau.ac.il)

Achieving a group velocity that exceeds the speed of sound in a waveguide presents a significant challenge in acoustic wave engineering. This task becomes even more demanding when aiming to enhance velocity without altering the waveguide's geometry, such as by narrowing or widening, and without obstructing the passage with passive or active inclusions. In this work, we address this problem by leveraging principles from non-Hermitian physics, implemented through active elements seamlessly integrated into the waveguide walls. These elements operate in a real-time feedback loop, introducing local pressure gain and loss, as well as non-local couplings based on pressure integration. By carefully tuning the control couplings, informed by lattice theory and adapted to the waveguide configuration, we achieve a stable parity-time-symmetric regime that governs the wave dynamics. Through numerical simulations and experimental validation in an air-filled waveguide, we demonstrate the accelerated propagation of a wave packet. Additionally, we explore the balance between system stabilization and the extent of achievable velocity enhancement. This study paves the way for innovative approaches to wave transmission in continuous media, enabled by short- and long-range active couplings implemented via embedded real-time feedback mechanisms.

Contributed Papers

4:40

5pPAb11. Tunability of superradiant scattering. Lauryn Schilling (Dept. of Mech. and Process Eng., ETH Zürich, Zürich, Switzerland), Tiemo Pedergnana (Dept. of Mech. and Process Eng., ETH Zürich, Sagenmattstrasse 11, Luzern 6003, Switzerland, pedergnanatiemo@gmail.com), and Nicolas Noiray (Dept. of Mech. and Process Eng., ETH Zürich, Zürich, Switzerland)

A low-Mach airflow passing over a cavity can cause aeroacoustic instabilities, resulting in whistling sounds. These instabilities take place when the hydrodynamic mode of the fluid and the acoustic mode of the cavity form constructive feedback exceeding the dissipative losses. The resulting limit cycle has been observed to enable tunable compensation of transmission losses, which is desirable in a broad range of phenomena from lossless non-reciprocal transmission to acoustic cloaking. The general method works by scattering harmonic incident waves with sufficiently large amplitude by a cavity limit cycle oscillating in the weakly non-linear regime, forcing the limit cycle to synchronize with the incident waves. At suitable conditions, the resulting non-linear wave-mode coupling leads to superradiant amplification of the outgoing waves. This work is focused on the tunability of superradiant scattering with regard to the incident wave amplitude and the bifurcation parameter. To investigate the tunability, experiments were performed on a superradiant aeroacoustic meta-atom, which has previously been demonstrated to amplify harmonic acoustic waves. These experimental validations show excellent agreement with corresponding theoretical predictions. As a key result, it is demonstrated that superradiant amplification can be achieved at small incident wave amplitudes, given that the bifurcation parameter is also small enough.

5:00

5pPAb12. Vibration control in rotor chain structures with dual topological states. Soroush Soltani (Mech. Eng., Univ. of Vermont, 135 Centennial Court, Burlington, VT 05401, Soroush.Soltani@uvm.edu) and Jihong A. Ma (Mech. Engineering/Phys., Univ. of Vermont, Burlington, VT)

Rotor chains are mathematical-mechanical models used to analytically investigate wave propagation in periodic systems, such as folding-based mechanical metamaterials. These models facilitate the exploration of topological states in mechanical systems, enabling vibration localization and enhanced vibration isolation in specific structural regions. In this study, we present a one-dimensional rotor-chain system (RCS) model that employs bar-hinge elements while considering rotational stiffness and varying angles between the bars. By tuning key parameters, we realize dual types of topological states at both zero and finite frequencies. The analysis demonstrates that adjusting rotational stiffness and bar angles enables tunable vibration localization in different regions of the lattice. These findings provide valuable insight into controlling wave propagation and vibration isolation, laying the groundwork for designing robust mechanical systems, such as origami-based structures, with applications in aerospace, deployable systems, and noise and vibration control technologies.

Session 5pUWa

Underwater Acoustics and Acoustical Oceanography: Bill Kuperman (1943-2024): Contributions to the Field of Underwater Acoustics II

Kevin D. Heaney, Cochair
Applied Ocean Sciences, Fairfax Station, VA 22039

Christopher Verlinden, Cochair
Applied Ocean Physics LLC, Fairfax Station, VA 22039

David R. Barclay, Cochair
Dept. of Oceanogr., Dalhousie Univ., P.O. Box 15000, Halifax B3H 4R2, Canada

Contributed Paper

1:00

5pUWa1. A geoacoustic Bayesian inversion of seafloor sediment properties using a large calibrated single-beam echosounder dataset. Andrew P. Niedbala (Ctr. for Acoust. Res. and Education, and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, 50 Cswy. St., 1409, Boston, MA 02114, andrew.niedbala@unh.edu), Jenna Hare (Ctr. for Acoust. Res. and Education, and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH), Amy Geist (U.S. Geological Survey, Woods Hole, MA), and Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, and Dept. of Civil and Environ. Eng., Univ. of New Hampshire, Durham, NH)

Seabed characterization is of interest to those studying benthic environments, pursuing offshore development, and improving national defense capability, however, direct sampling is costly even for a limited geographic scope. The National Oceanic and Atmospheric Administration (NOAA) maintains a database containing more than 200 000 km of calibrated, multifrequency, single beam echosounder (SBES) data taken over the last 15 years. Bayesian geoacoustic inversion using these data provides a promising and impartial framework by which the posterior probability distributions (PPDs) of model parameters are estimated given the information content in the data. Here, an established time domain scattering model is used, containing scattering contributions from the bulk properties, interface roughness, and heterogeneity in the sediment. The forward model is augmented, coupling the bulk geoacoustic model parameters using empirical relationships based on index of impedance, and allowing for a depth-dependent bulk density. Efficient sampling of model parameter space is employed using parallel tempering methods and principal component analysis. To provide proof-of-concept, sandy, silty, and muddy sites are chosen that contain both NOAA SBES returns and ground truth. PPDs presented agree well with ground truth, demonstrating promise for seabed characterization and associated uncertainty at unprecedented spatial scales. [Work sponsored by ONR, N000142312819.]

Invited Papers

1:20

5pUWa2. A journey with Bill Kuperman from ocean acoustics to structural health monitoring. Sandrine T. Rakotonarivo (Aix-Marseille Univ. and Lab. of Mech. and Acoust., LMA - UMR 7031 AMU - CNRS - Centrale Marseille, 4 impasse Nikola Tesla, Marseille 13453, France, sandrine.rakotonarivo@univ-amu.fr)

Bill Kuperman had a tremendous impact on ocean acoustics through his research focusing on sound propagation, analysis, and sources localization. His work would combine modeling, signal processing, and experimental developments. Eager to learn and interact with other fields, he has also contributed to the development of source localization and structure characterization methods for other fields, such as structural acoustics, ultrasonic non-destructive testing, or structural health monitoring. In particular, he was a pioneer in developing Matched Field processing for source localization or for structure characterization in diffuse or reverberating environments. This presentation will talk about his extended contributions to source localization and structure characterization in such environments in the fields of underwater acoustics, structural acoustics, non-destructive testing, and structural health monitoring.

1:40

5pUWa3. An homage to Bill Kuperman's wizardry: Turning noise into signal. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Bill Kuperman's contributions to the field of underwater acoustics have been numerous and tremendous. I would like to share some memories of my time working with Bill starting in 2003 as a Postdoc under his mentoring at the Marine Physical Laboratory (Scripps Institute of Oceanography, UC San Diego). At that time, Bill was investigating the use of coherent processing of ocean and seismic ambient noise as a novel means to perform passive remote sensing, i.e., without using conventional active sources. Bill's intuition,

charisma, leadership, scholarly experience, and constant encouragement to explore new research avenues created a unique work environment at the Marine Physical Laboratory, which I greatly benefited from, and I am most grateful for.

2:00

5pUWa4. Time reversal-based underwater acoustic communications. Heechun Song (SIO, UCSD, 8820 Shellback Way, Spiess Hall, Rm 448, La Jolla, CA 92093-0238, hcsong@ucsd.edu) and William Hodgkiss (SIO, UCSD, San Diego, CA)

Over more than a decade beginning in 1996, we conducted a series of time reversal mirror (TRM) experiments in collaboration with the NATO Undersea Research Centre in coastal waters. The method leverages spatial diversity to achieve both spatial and temporal focusing in complex environments, offering a promising alternative to conventional multichannel equalizers (M-DFE) for underwater communications. Temporal focusing enables self-equalization and mitigation of intersymbol interference (ISI), which can then be followed by an adaptive decision feedback equalizer (DFE), known as TR-DFE. Additionally, spatial focusing facilitates multiuser communications, with adaptive TR further reducing crosstalk among users. Our findings reveal that TR-DFE outperforms M-DFE, particularly when high-order constellations, such as 32-QAM, are employed, significantly enhancing data throughput and spectral efficiency. This presentation will showcase examples inspired by Bill's invaluable contributions and insights.

2:20

5pUWa5. Wisdom, waveguide invariant theory, and whales: Recollections of Bill Kuperman as a graduate advisor. Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0206, La Jolla, CA 92093, athode@ucsd.edu)

I was one of Bill's first PhD students when he moved to Scripps, and he was instrumental in advancing my career. I see Bill as a blend of the physicist Niels Bohr, catcher Yogi Berra, and Jedi Master Yoda. Like Niels Bohr, he has been a keystone scientist and beloved role model and mentor for generations of students and colleagues. Like his fellow New Yorker Yogi Berra, whom he loved to (mis)quote, he had an uncanny ability to distill complex concepts, physical and political, into Zen-like zingers ("Be the first with the worst"), handy to keep in your pocket. Like Yoda, he had a deep physical insight into the universe (or at least the wet parts that wobble) that bordered on mystical intuition. I was fond of him beyond words, so I never said much, but from him I've learned to distrust large numerical computations without insight ("big computer, small mind"), revere the Pekeris waveguide ("it explains nearly everything"), and firmly believe in intellectual generosity and honesty ("it's best to be honest, because my father told me it gets too hard to keep track of all your lies"). I will also discuss whales (a little) and waveguide invariant theory (a lot).

2:40

5pUWa6. In memory of Bill. Earl G. Williams (Acoust., Naval Res. Lab., Naval Res. Lab., 4555 Overlook Ave., Code 7106, Washington, DC 20375, earl.williams@nrl.navy.mil)

Our paths joined in 2012 when I spent a six month sabbatical with him at MPL. He and Gaby were the most welcoming of hosts, and this started off a 12 year bicoastal collaboration that was unique and inspirational. Bill was humble, always non-threatening, welcoming new ideas and always presenting his side of the story with a way of thinking that was rare and perceptive. I would return from my week-long trips to UCSD mentally exhausted, but with an onslaught of new ideas. Our intellectual partnership was one of a kind—a once in a lifetime association. And I am so thankful and fortunate that he was part of my professional life over the past 12 years. He became like an older brother to me. It is so hard to accept that he is no longer with us. However, in his memory, we can all strive to emulate his selfless, unassuming character—supporting and inspiring a new generation of scientists. [Work supported by the Office of Naval Research.]

Contributed Paper

3:00

5pUWa7. Efficient and robust 3D perception of seafloor targets using a single acoustic camera for low-visibility marine environments. Xiaoteng Zhou (The Univ. of Tokyo, 5-1-5, Kashiwanoha, Kashiwa, Chiba 277-8583, Japan, zhouxiaoteng@g.ecc.u-tokyo.ac.jp), Yusheng Wang, and Katsunori Mizuno (The Univ. of Tokyo, Kashiawa, Japan)

Accurate and reliable three-dimensional (3-D) perception in low-visibility seafloor environments is essential for advancing underwater investigations. This study introduces a novel and efficient approach for rapid 3D seafloor target perception using a single acoustic camera (a forward-looking sonar). Unlike previous methods, the proposed approach eliminates the reliance on complex 3-D reconstruction theories and assumptions about target geometry, making it more robust and rapid. The information required for 3-D geometric solving is directly derived from the interaction between sonar motion and the acoustic imaging mechanisms. By integrating visual clues such as highlights and shadows in sonar images with sensor motion information, the 3-D bounding box of each seafloor target can be effectively inferred. The effectiveness of the proposed approach was validated through both simulation and real water tank experiments. The results demonstrate its robustness and accuracy in perceiving both structured and unstructured targets, highlighting the practicality of this approach in real-world underwater environments. This study achieves a 3-D perception of seafloor targets through rapid estimation of their bounding boxes, significantly improving robustness and efficiency while meeting the requirements of most marine engineering tasks. Potential applications include marine environment monitoring in dark conditions and autonomous navigation of marine robots in turbid water environments.

Session 5pUWb

Underwater Acoustics: Underwater Acoustic Propagation

Gavin Dies, Cochair
Ocean Eng., Univ. of Rhode Island, 281 Kingstown Rd., Narragansett, RI 02882

Megan Ballard, Cochair
Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665

Contributed Papers

3:40

5pUWb1. Acoustic propagation influenced by an anticyclonic eddy near Jan Mayen Island. Megan Ballard (Appl. Res. Labs, Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu), Robert T. Taylor, Jason D. Sagers (Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX), Timothy F. Duda, J. T. Farrar (Woods Hole Oceanographic Inst., Woods Hole, MA), Alejandra Sanchez-Rios, Jennifer MacKinnon (Scripps Inst. of Oceanogr., San Diego, CA), Andrew J. Lucas (Scripps Inst. of Oceanogr., La Jolla, CA), Harper Simmons, and Leah Johnson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

This paper presents measurements collected on a vertical line array (VLA) of hydrophones moored near Jan Mayen Island, an area where warm, salty Atlantic waters mix with fresher polar waters from the Greenland Basin. The dataset compares signals simultaneously collected along two propagation paths: a moored source located 42 km north of the array and a drifting source that originated near the array and then moved northeast. Water-column properties were measured extensively using sensors on the moorings, on the buoy of the drifting source, and on ship deployed instruments. The water-column sound-speed profile included a warm surface mixed layer and a subsurface acoustic duct with an axis near 200 m. Both the moored and drifting sources were located at a depth of 100 m, and signals from both sources presented a ducted arrival measured on the VLA. While the sound-speed conditions between the moored source and VLA were more quiescent, the drifting source became trapped in an eddy with a deep mixed layer, intensifying the downward refraction of sound, and lengthening the ray cycle distance. As the source opened to a range of 43 km, the ducted path became bathymetrically blocked, and the signal could no longer be detected.

4:00

5pUWb2. Spatial sound field variability and mode coupling during upslope propagation in a coastal wedge. Marina Yarina (Marine Geoscience, Univ. of Haifa, Haifa, Israel) and Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave. Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il)

Wedge-like shallow water model is often being used for the sound field propagation in a near-coastal zone of the ocean. The canonical model, which has an analytical solution and is used in many applications, assumes a constant sound speed in the water layer. However, it was shown (Jiang *et al.*, JASA EL, 2023) that the presence of a thermocline significantly changes the structure of the sound field when propagating downslope due to the emerging specific mode coupling. In this paper, the problem of the sound field in such a wedge is considered for upslope propagation and is studied depending on the sound speed profile, in other words, depending on the thermocline parameters. The sound field is constructed both using an expansion over adiabatic modes and using a parabolic equation. It is shown that the mode coupling arising due to local violation of adiabaticity, leads to noticeable variability of the spatial distribution of the field, the field decomposition by modes, and also to the frequency dependence. The results are illustrated by calculations for real parameters in shallow water. [Work was supported by ISF (Grant No. 946/20)]

4:20

5pUWb3. Approximating underwater sound propagation with neural networks: The case of normal mode theory. Arthur Varon (ALSEAMAR, 11 Rue des Mathématiques, Saint-Martin-d'He?res, Ise?re 38400, France, arthur.varon@gipsa-lab.grenoble-inp.fr), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Rémi Emmetiere (ALSEAMAR, Rousset, France), and Jérome Mars (GIPSA-Lab, Saint-Martin-d'He?res, France)

In shallow water environments and at low frequencies, normal mode theory provides an adequate description of the acoustic propagation; however, conventional normal mode simulation codes can be computationally intensive. To address this, we propose using neural networks (NNs) to approximate the modal wavenumbers and the modal depth functions. Predicting modal parameters using NNs is a relatively straightforward task compared to predicting more variable acoustic quantities, such as those involving modal interference like transmission loss. This approach allows NNs to be trained to approximate modal parameters across diverse environments and at different frequencies, thereby eliminating the need for expensive retraining of the NNs models. Once trained, the predicted modal wavenumbers and modal depth functions can be used according to the normal mode theory to obtain the acoustic field for any source and receiver positions. This approach divides the computation time to obtain the modal parameter by a factor of thirty compared to traditional normal mode codes, making it a promising solution for applications with limited computational resources, such as simulation on autonomous underwater vehicles.

4:40

5pUWb4. Sound speed data categorization. Luke Clark, Tassia Jones, Gabi Kojder, Archie Soares-Mullen (Univ. of Bristol, Bristol, United Kingdom), and Duncan Williams (Dstl, Dstl Porton Down, Salisbury SP4 0JQ, United Kingdom, dpwilliams@dstl.gov.uk)

Categorising sound speed data across different geographic regions can be useful for describing the general distribution of sound speed data and for identifying variations in sound speed data, which can be used to reduce the computational costs associated with propagation loss modelling. Sound speed data changes spatially and temporally over different length and time scales making it costly to calculate and store the propagation loss between all points in a region of the ocean. This paper presents a new parameterization and categorization scheme for sound speed data, which employs proper orthogonal decomposition to organize individual sound speed profiles into a small number of categories based on the structural similarity of sound speed data in each category. We show how, using this scheme, the likelihood of occurrence of each category, and the likelihood of any individual sound speed profile fitting into each category, can be calculated and then used to

identify global and local variations in sound speed data and consequently improve the efficiency and prediction accuracy of ocean acoustic modelling.

5:00

5pUWb5. Modeling Coastal Virginia Offshore Wind bubble curtain effectiveness. Gavin Dies (Ocean Eng., Univ. of Rhode Island, 281 Kingstown Rd., Narragansett, RI 02882, gavin.dies@uri.edu), Ying-Tsong Lin (Univ. of California, San Diego, La Jolla, CA), Gopu R. Potty, Jennifer Amaral, James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Anwar A. Khan (HDR, Inc., Fort Lauderdale, FL)

The Coastal Virginia Offshore Wind (CVOW) consists of two turbines roughly 40 km off the coast of Virginia Beach, Virginia. Water- and seabed-borne acoustic signals from impact pile driving at CVOW were recorded during the installation of these turbines in May 2020. In-water pressure signals were measured using various methods at multiple ranges. The water depth at the wind turbines and the measurement locations were about 26 m. During installation, one of the piles utilized a double bubble curtain during construction, while the other did not. Bubble curtains are used to reduce the acoustic impacts of pile driving by creating a barrier of bubbles around the source. This arrangement has allowed for an in-depth analysis of the bubble curtain's noise control characteristics. Analysis of acoustic measurements found that bubble curtains are effective at frequencies above 200 Hz and can reduce the Sound Power Level by more than 25 decibels at certain frequencies and depths. CVOW monopile installation was modeled using a coupled Finite Element and Parabolic Equation model to replicate the insertion loss of the bubble curtain. The modeled and measured insertion loss were compared. Frequency analysis of seabed-borne propagation was also performed to contrast water-borne propagation. [Work sponsored by BOEM.]

AUTHOR INDEX

to papers presented at

188th Meeting of the ASA//ICA25 New Orleans