JASA

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184th Meeting of the Acoustical Society of America

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8-12 MAY 2023

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ACOUSTICS 23
Sydney

4–8 December 2023

Acoustical Society of America
Australian Acoustical Society
Western Pacific Acoustics Conference
Pacific Rim Underwater Acoustics Conference

The Acoustical Society of America (ASA) and the Australian Acoustical Society (AAS) invite acousticians from around the world to participate in the joint meeting to be held 4–8 December 2023 in Sydney, Australia. A broad range of topics in acoustics will be covered in technical sessions and keynote lectures. Presentations on emerging topics are especially encouraged. The meeting is cosponsored by the Western Pacific Acoustics Conference and the Pacific Rim Underwater Acoustics Conference. Social events, student events, and an accompanying persons program will be organized. The best features of meetings of all organizations will be combined to offer a premier venue for presenting your work to an international audience.

Sydney is located on the east coast of Australia and is the state capitol of New South Wales. Situated on Darling Harbor, Sydney was established in 1788, and is best known for its harbor front Sydney Opera House, with a distinctive sail-like design.

Please Join Us!

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The Acoustical Society of America was founded in 1929 to generate, disseminate, and promote the knowledge and practical applications of acoustics. Any person or corporation interested in acoustics is eligible for membership in this Society. Information concerning membership may be obtained from Elaine Moran, Director of Operations, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300, T: 516-576-2360, E-mail: elaine@acousticalsociety.org. Options for joining the Society can be found at https://acousticalsociety.org/asa-membership/

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OF AMERICA

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SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Sun, 7 May, 6:30 p.m.	Executive Council Dinner	Navy Pier
Mon, 8 May, 9:00 a.m.	Executive Council	Kane
Mon, 8 May, 1:00 p.m.	Technical Council	Kane
Tue, 9 May, 7:00 a.m.	Member Engagement	Old Town
Tue, 9 May, 7:30 a.m.	Editorial Board	Kane
Tue, 9 May, 2:00 p.m.	Meetings	River North
Tue, 9 May, 4:00 p.m.	Newman Fund	Old Town
Tue, 9 May, 5:00 p.m.	Women in Acoustics	Kane
Wed, 10 May, 7:00 a.m.	Regional and Student Chapters	Kane
Wed, 10 May, 7:30 a.m.	Finance	Old Town
Wed, 10 May, 11:00 a.m.	Public Relations	River North
Thu, 11 May, 1:00 p.m.	Investments	Old Town
Fri, 12 May, 8:00 a.m.	Technical Council	Kane
Fri, 12 May, 12:00 noon	Executive Council	Kane

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 9 May, 4:45 p.m.	Engineering Acoustics	Northwestern/Ohio State
Tue, 9 May, 5:30 p.m.	Signal Processing in Acoustics	Purdue/Wisconsin
Tue, 9 May, 7:30 p.m.	Acoustical Oceanography	Michigan/Michigan State
Tue, 9 May, 7:30 p.m.	Animal Bioacoustics	Chicago FG
Tue, 9 May, 7:30 p.m.	Architectural Acoustics	Chicago AB
Tue, 9 May, 7:30 p.m.	Physical Acoustics	Los Angeles/Miami/ Scottsdale
Tue, 9 May, 7:30 p.m.	Psychological and Physiological Acoustics	Indiana/Iowa
Wed, 10 May, 7:30 p.m.	Biomedical Acoustics	Armitage
Wed, 10 May, 7:30 p.m.	Structural Acoustics and Vibration	Chicago H
Thu, 11 May, 4:30 p.m.	Computational Acoustics	Lincolnshire 1/2
Thu, 11 May, 7:30 p.m.	Musical Acoustics	Lincolnshire 1/2
Thu, 11 May, 7:30 p.m.	Noise	Chicago C
Thu, 11 May, 7:30 p.m.	Speech Communication	Los Angeles/Miami/ Scottsdale
Thu, 11 May, 7:30 p.m.	Underwater Acoustics	Michigan/Michigan State

STANDARDS COMMITTEES AND WORKING GROUPS

Tue, 9 May, 8:00 a.m.	Standards Plenary including TAGs	McHenry
Tue, 9 May, 9:30 a.m.	ASC S1 Acoustics	McHenry
Tue, 9 May, 11:00 a.m.	ASC S2 Mechanical Vibration and Shock	McHenry
Tue, 9 May, 12:45 p.m.	ASC S3 Bioacoustics	McHenry
Tue, 9 May, 2: 15 p.m.	ASC S3/SC1 Animal Bioacoustics	McHenry
Tue, 9 May, 3:45 p.m.	ASC S12 Noise	McHenry

ASACOS and ASACOS Steering meet virtually before meetings. Refer to https://asastandards.org/#meetings

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon–Thu, 8–11 May 7:00 a.m. –5:00 p.m. Fri, 12 May 7:30 a.m.–1:00 p.m.	Registration	Halstead (4th Fl)
Mon–Thu, 8–11 May 7:00 a.m.–5:00 p.m. Fri, 12 May 7:00 a.m.–12:00 noon	A/V Preview	Minnesota (6th Fl)
Mon–Thu, 8–11 May 8:00 a.m.–5:00 p.m. Fri, 12 May 8:00 a.m.–12:00 p.m.	Mothers' Room	7th Fl Registration Office
Mon–Fri, 8–12 May, 9:30 a.m.–11:00 a.m.	Morning Coffee Breaks	Chicago D/E
Tue, 9 May 2:30 p.m.–3:30 p.m.	Afternoon Coffee Break	Chicago D/E
Mon–Fri, 8–12 May 8:00 a.m.–10:00 a.m.	Accompanying Persons	Navy Pier
Mon, 8 May 1:00 p.m.–4:00 p.m.	Solutions Showcase	McHenry
Mon, 8 May 4:00 p.m.–5:00 p.m.	Keynote Lecture	Salon II
Mon, 8 May 5:30 p.m.–7:00 p.m.	Exhibit Opening Reception	Chicago D/E
Mon, 8 May 5:30 p.m.–6:00 p.m.	First-Time Attendee Orientation	Chicago A/B
Mon, 8 May 6:00 p.m.–7:30 p.m.	Student Meet and Greet	Los Angeles/Miami/ Scottsdale
Tue, 9 May 9:00 a.m.–5:00 p.m. Wed, 10 May 9:00 a.m.-12:00 p.m.	Exhibit	Chicago D/E
Tue, 9 May 11:30 a.m.–1:00 p.m.	Student Career Mixer	Salon III
Tue, 9 May 6:00 p.m.–7:30 p.m.	Social Hour	Salons II/III
Wed, 10 May 11:45 a.m.–1:30 p.m.	Women in Acoustics Luncheon	Salon III
Wed, 10 May 2:30 p.m.–3:30 p.m.	Women in Acoustics Roundtable	Kane
Wed, 10 May 3:45 p.m.–5:00 p.m.	Plenary Session/Awards Ceremony	Salon II
Wed, 10 May 5:00 p.m.–7:00 p.m.	Affinity Room	McHenry
Wed, 10 May 6:00 p.m.–8:00 p.m.	Student Reception	Salon III
Wed, 10 May 8:00 p.m.–12:00 midnight	ASA Jam	Salon II
Thu, 11 May 12:00 noon–2:00 p.m.	Society Luncheon and Lecture	Salon III
Thu, 11 May 6:00 p.m.–7:30 p.m.	Social Hour	Salons II/III
Fri, 12 May 3:00 p.m.–8:00 p.m.	Interagency DEI	Avenue

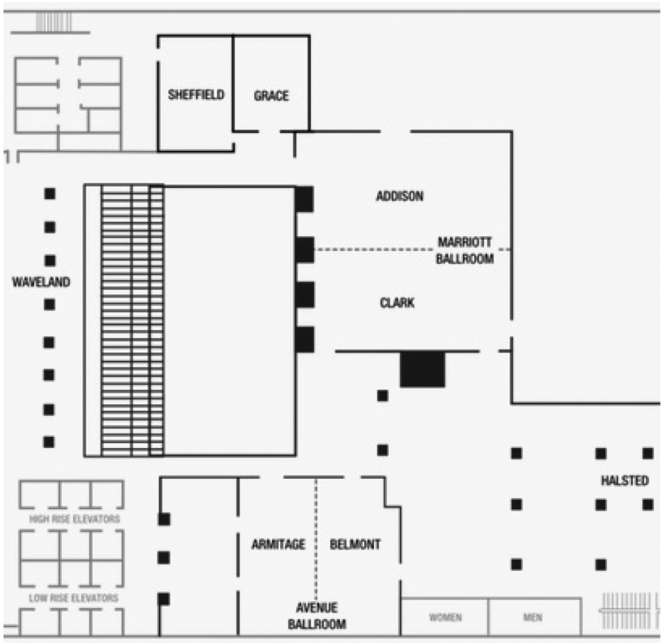
Chicago Downtown Marriott Magnificent Mile Hotel
Room Directory

Room	Floor
Armitage	4
Belmont	4
Chicago A/B	5
Chicago C	5
Chicago D/E	5
Chicago F/G	5
Chicago H	5
Denver/Houston/Kansas City	5
Dupage	3
Halsted	4
Huron	10
Indiana/Iowa	6
Kane	3
Los Angeles/Miami/Scottsdale	5
McHenry	3

Room	Floor
Michigan/Michigan State	6
Navy Pier	10
Northwestern/Ohio	6
O'Hare	10
Old Town	2
Printers Row	2
Purdue/Wisconsin	6
Registration Office North	7
River North	2
Salon I	7
Salon II	7
Salon III	7
Streeterville	2
Watertower	10
Wrigleyville	2

Chicago Marriott Downtown Magnificent Mile Hotel

4th Floor



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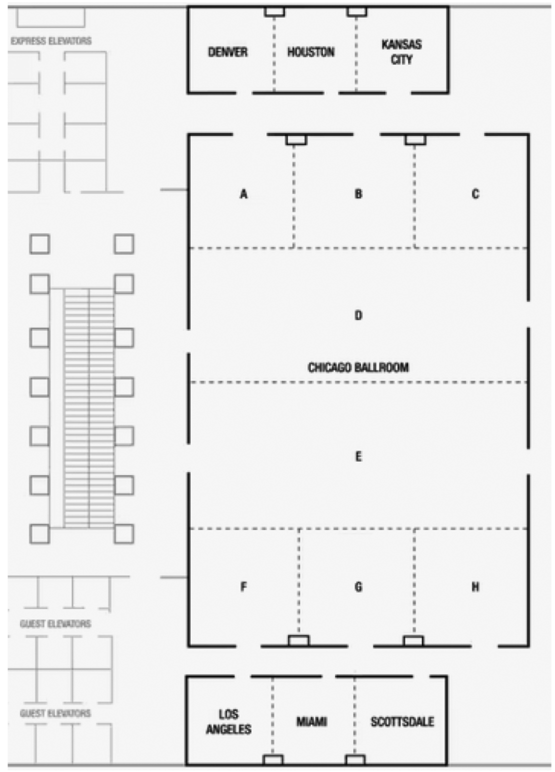

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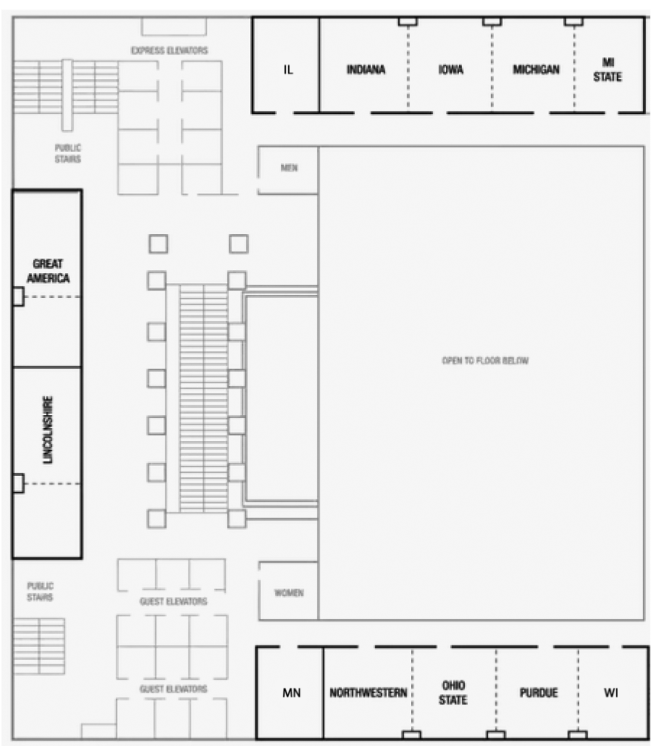

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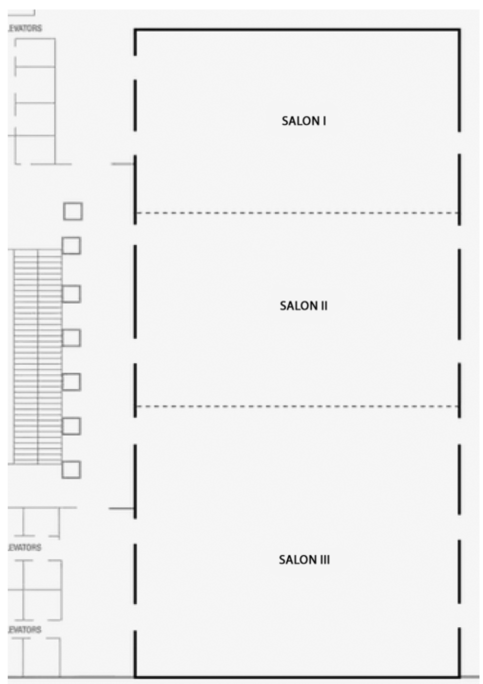
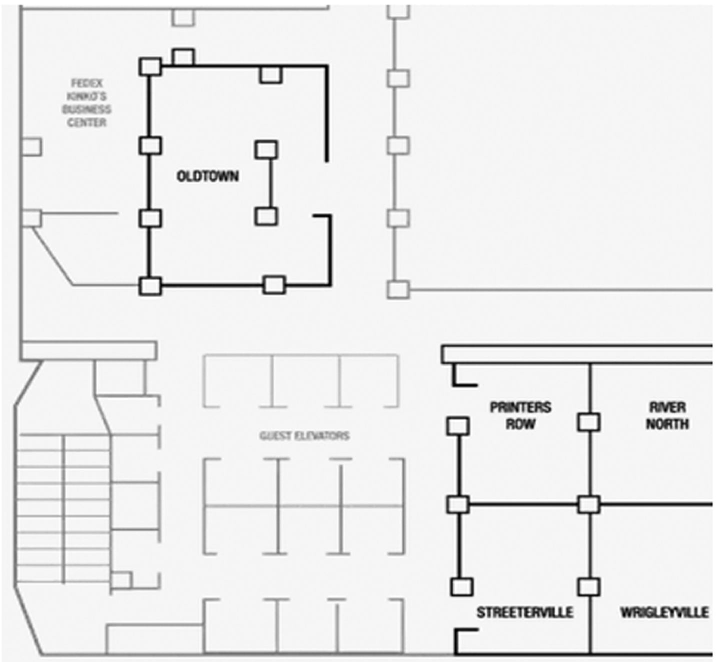
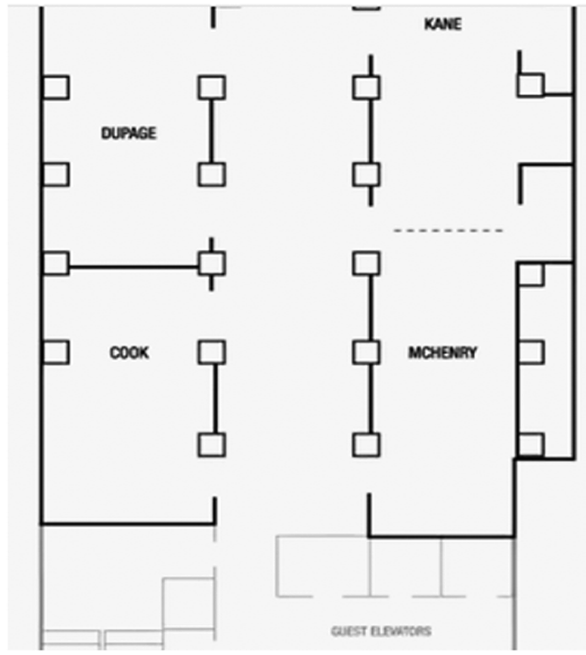


Chicago Marriott Downtown Magnificent Mile Hotel

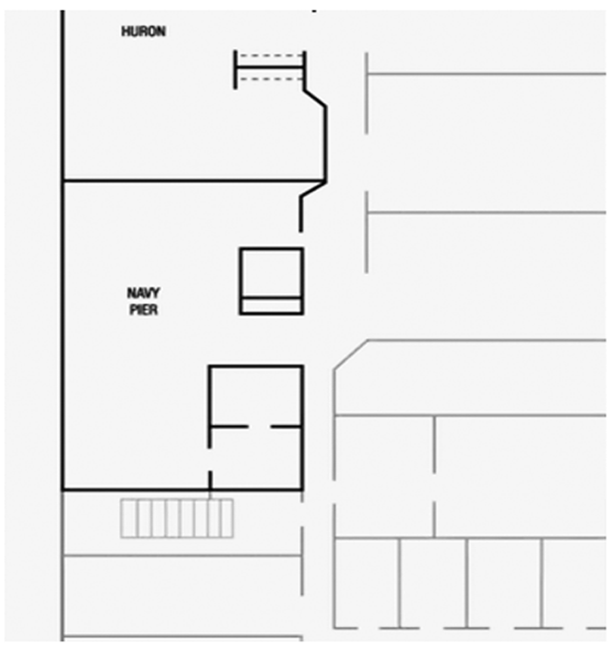
2nd Floor



3rd Floor



10th Floor



TECHNICAL PROGRAM CALENDAR
184th Meeting of the Acoustical Society of America
8-12 May 2023

Please refer to the Itinerary Planner and Mobile App for Updated Information

Monday Morning

8:00 1aAA **Architectural Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Application of Statistical and Computational Methods in Building Acoustics. Chicago A/B

8:00 1aAB **Animal Bioacoustics, Acoustical Oceanography, Noise, Underwater Acoustics, and Signal Processing in Acoustics:** Climate Change and Sound: How the Sound of the Planet Reflects the Health of the Planet I (Hybrid Session). Chicago F/G

8:00 1aBAa **Biomedical Acoustics**: Ultrasound Brain and Super-Resolution Imaging I. Belmont

8:20 1aBAb **Biomedical Acoustics and Physical Acoustics**: Ultrasound Induced Cell Responses. Armitage

10:00 1aCA **Computational Acoustics, Architectural Acoustics, Musical Acoustics, and Physical Acoustics**: Real-Time Computing I (Hybrid Session). Lincolnshire 1/2

9:00 1aEA **Engineering Acoustics**: Acoustic Transducers, Devices, and Systems. Northwestern/Ohio State

9:00 1aMU **Musical Acoustics**: Musical Acoustics General Topics I. Indiana/Iowa

8:00 1aNS **Noise, Computational Acoustics, and Physical Acoustics:** Rocket Noise Part I. Chicago C

8:10 1aPA **Physical Acoustics**: Acoustic Remote Sensing in Urban Environments. Great America 1/2

8:00 1aPP **Psychological and Physiological Acoustics**: Psychological and Physiological Acoustics Poster Session I. Salon I

9:00 1aSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, Signal Processing in Acoustics and Noise**: Acoustic Excitation of Structures I. Chicago H

8:30 1aSP **Signal Processing in Acoustics, Biomedical Acoustics, Underwater Acoustics, Physical Acoustics, and Architectural Acoustics**: Acoustical Imaging, Reconstruction, and Localization I: General Applications I (Hybrid Session). Purdue/Wisconsin

10:00 1aUW **Underwater Acoustics, Acoustical Oceanography and Physical Acoustics**: Exploring Fine-Grained Sediments in the Variable Ocean I (Hybrid Session). Michigan/Michigan State

Monday Afternoon

1:00 1pAA **Architectural Acoustics, Musical Acoustics, and Noise**: Evaluation of Completed Performance Spaces: Goals and Methods. Chicago A/B

1:00 1pAB **Animal Bioacoustics, Acoustical Oceanography, Noise, Underwater Acoustics, and Signal Processing in Acoustics**: Climate Change and Sound: How the Sound of the Planet Reflects the Health of the Planet II (Hybrid Session). Chicago F/G

1:00 1pAO **Acoustical Oceanography**: Topics in Acoustical Oceanography. Indiana/Iowa

1:00 1pBAa **Biomedical Acoustics**: Ultrasound Brain and Super-Resolution Imaging II. Belmont

1:00 1pBAb **Biomedical Acoustics**: Ultrasound for Ocular Therapy. Armitage

1:00 1pCA **Computational Acoustics, Structural Acoustics and Vibration, and Physical Acoustics**: Computational Methods for Modeling Acoustic Damping. Lincolnshire 1/2

1:00 1pNS **Noise, Computational Acoustics, and Physical Acoustics:** Rocket Noise Part II. Chicago C

1:00 1pPA **Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics**: Advances in Sonochemistry. Great America 1/2

1:00 1pSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, Signal Processing in Acoustics and Noise**: Acoustic Excitation of Structures II. Chicago H

1:00 1pSC **Speech Communication**: Speech Perception I (Poster Session). Salon I

1:00 1pSP **Signal Processing in Acoustics, Biomedical Acoustics, Underwater Acoustics, Physical Acoustics, and Architectural Acoustics**: Acoustical Imaging, Reconstruction, and Localization II: General Applications II (Hybrid Session). Purdue/Wisconsin

1:00 1pUW **Underwater Acoustics, Acoustical Oceanography and Physical Acoustics**: Exploring Fine-Grained Sediments in the Variable Ocean II (Hybrid Session). Michigan/Michigan State

4:00 1eID **Interdisciplinary**: Keynote Lecture. Salon II

Tuesday Morning

8:00 2aAA **Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration**: Classroom Acoustics I (Hybrid Session). Denver/ Houston/Kansas City

8:00 2aAB **Animal Bioacoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics**: Session in Honor of James A. Simmons I (Hybrid Session). Great America 1/2

8:30 2aAO **Acoustical Oceanography, Underwater Acoustics, and Signal Processing**: Memorial Session for Jeffrey A. Nystuen I (Hybrid Session). Michigan/Michigan State

7:40 2aBAa **Biomedical Acoustics and Physical Acoustics**: Clinical Perspective of Biomedical Acoustics I (Hybrid Session). Armitage

8:00 2aBAb **Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics**: New Technology Developments for use in Focused Ultrasound Therapy I. Belmont

10:05 2aCA **Computational Acoustics, Architectural Acoustics, Musical Acoustics, and Physical Acoustics**: Real-Time Computing II (Hybrid Session). Lincolnshire 1/2

9:00 2aEA **Engineering Acoustics, Signal Processing in Acoustics, Noise, Underwater Acoustics, Architectural Acoustics, and Physical Acoustics**: Microphones: Design, Development, and Characterization I. Northwestern/Ohio State

7:40 2aMU **Musical Acoustics**: Musical Acoustics General Topics II. Lincolnshire 1/2

7:45 2aNS **Noise**: Assorted Topics on Noise. Chicago C

8:00 2aPAa **Physical Acoustics and Structural Acoustics and Vibration**: Acoustics and Elasticity of Consolidated and Unconsolidated Granular Materials I (Hybrid Session). Chicago F/G

9:45 2aPAb **Physical Acoustics, Education in Acoustics, and Structural Acoustics and Vibration**: Acoustics Demonstration Extravaganza. Chicago A/B

7:55 2aPP **Psychological and Physiological Acoustics**: David Green and Psychoacoustics (Hybrid Session). Indiana/Iowa

8:00 2aSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, and Computational Acoustics**: Acoustic Metamaterials I. Chicago H

9:00 2aSC **Speech Communication and Psychological and Physiological Acoustics**: Clear Speech(es) Across People, Places, and Time I. Los Angeles/Miami/ Scottsdale

8:40 2aSP **Signal Processing in Acoustics**: Signal Processing Potpourri. Purdue/Wisconsin

Tuesday Afternoon

1:35 2pAA **Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration**: Classroom Acoustics II (Hybrid Session). Denver/ Houston/Kansas City

1:00 2pAB **Animal Bioacoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics**: Session in Honor of James A. Simmons II (Hybrid Session). Great America 1/2

1:30 2pAOa **Acoustical Oceanography**: Acoustical Oceanography Prize Lecture. Chicago A/B

2:40 2pAOb **Acoustical Oceanography, Underwater Acoustics, and Signal Processing**: Memorial Session for Jeffrey A. Nystuen II (Hybrid Session). Michigan/Michigan State

1:00 2pBAa **Biomedical Acoustics and Physical Acoustics**: Clinical Perspective of Biomedical Acoustics II (Hybrid Session). Armitage

1:00 2pBAb **Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics:** New Technology Developments for use in Focused Ultrasound Therapy II. Belmont

1:00 2pCA **Computational Acoustics, Structural Acoustics and Vibration, Noise, Musical Acoustics, Physical Acoustics, and Acoustical Oceanography**: Wind Acoustics. Lincolnshire 1/2

1:00 2pEA **Engineering Acoustics, Signal Processing in Acoustics, Noise, Underwater Acoustics, Architectural Acoustics, and Physical Acoustics**: Microphones: Design, Development, and Characterization II. Northwestern/Ohio State

1:00 2pED **Education in Acoustics, Physical Acoustics, Noise, Architectural Acoustics, and Engineering Acoustics**: Assessment of Acoustics Education. Indiana/Iowa

4:30 2pMU **Musical Acoustics**: Concert Session: Harp Duo Julie Spring and Ellie Kirk. Chicago A/B

1:00 2pNS **Noise**: Incorporating Tones in Noise Criteria. Chicago C

1:00 2pPA **Physical Acoustics, Biomedical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics**: Interaction of Electromagnetic Waves with Acoustic Waves. Chicago F/G

1:00 2pPP **Psychological and Physiological Acoustics**: Best Student Paper Award: Psychological and Physiological Acoustics (Poster Session). Salon I

1:00 2pSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, and Computational Acoustics**: Acoustic Metamaterials II. Chicago H

1:00 2pSCa **Speech Communication and Psychological and Physiological Acoustics**: Clear Speech(es) Across People, Places, and Time II (Poster Session). Salon I

1:00 2pSCb **Speech Communication**: Speech in Different Listening Conditions (Poster Session). Salon I

1:00 2pSP **Signal Processing in Acoustics and Computational Acoustics**: Feature Extraction, Dimensionality Reduction, and Learning in Ocean Acoustics. Purdue/Wisconsin

Wednesday Morning

8:00 3aAA **Architectural Acoustics, Noise, ASA Committee on Standards, Engineering Acoustics, and Structural Acoustics and Vibration**: Application and Development of Standards Used in Noise and Architectural and Structural Acoustics. Denver/Houston/Kansas City

8:00 3aABa **Animal Bioacoustics**: Acoustic Ecology and Biological Soundscapes. Great America 1/2

10:30 3aABb **Animal Bioacoustics**: Animal Vocal Communication and Physiology. Great America 1/2

8:00 3aBAa **Biomedical Acoustics**: Lung Ultrasound. Belmont

8:00 3aBAb **Biomedical Acoustics and Education in Acoustics**: Best Practices in Mentoring for Biomedical Acousticians. Armitage

8:45 3aCA **Computational Acoustics**: Topics in Computational Acoustics. Lincolnshire1/2

9:00 3aEA **Engineering Acoustics, Signal Processing in Acoustics, Biomedical Acoustics, Physical Acoustics, and Underwater Acoustics**: Additive Manufacturing of Acoustic Transducers. Northwestern/Ohio State

8:40 3aMU **Musical Acoustics**: Acoustics of Stringed Instruments. Indiana/Iowa

8:00 3aNS **Noise, Architectural Acoustics, Acoustical Oceanography, Engineering Acoustics, and Underwater Acoustics**: Post Pandemic Soundscapes Part I (Hybrid Session). Chicago C

8:25 3aPA **Physical Acoustics and Structural Acoustics and Vibration**: Acoustics and Elasticity of Consolidated and Unconsolidated Granular Materials II (Hybrid Session). Chicago F/G

8:00 3aPP **Psychological and Physiological Acoustics and Speech Communication**: Sensory and Non-Sensory Influences on Auditory Development. Chicago A/B

8:30 3aSA **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics**: Historical Perspectives in Structural Acoustics (Hybrid Session). Chicago H

8:00 3aSC **Speech Communication and Education in Acoustics**: Infusing Social Justice in Speech and Hearing Acoustics Pedagogy: Principles and Case Studies (Hybrid Session). Los Angeles/Miami/Scottsdale

8:00 3aSP **Signal Processing in Acoustics, Education in Acoustics, and Physical Acoustics**: My Favorite Signal Processing Homework Problems. Purdue/Wisconsin

8:00 3aUW **Underwater Acoustics, Acoustical Oceanography, Computational Acoustics, Physical Acoustics, Noise, and Architectural Acoustics**: 3D Acoustic Propagation. Michigan/Michigan State

Wednesday Afternoon

1:00 3pAA **Architectural Acoustics**: Hot Topics: Measuring Acoustic Properties and Beyond. Denver/Houston/Kansas City

1:00 3pAB **Animal Bioacoustics**: General Topics in Animal Bioacoustics. Great America 1/2

1:00 3pBAb **Biomedical Acoustics**: Biomedical Acoustics Student Paper Competition. Salon I

1:00 3pCA **Computational Acoustics, Architectural Acoustics, Physical Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics**: Boundary Conditions Across Acoustics. Lincolnshire 1/2

1:00 3pED **Education in Acoustics**: Acoustics Education Prize Lecture. Armitage

1:00 3pID **Interdisciplinary**: Hot Topics in Acoustics. Northwestern/Ohio State

1:00 3pMU **Musical Acoustics, Signal Processing in Acoustics, and Physical Acoustics**: Music Synthesis Techniques Applied to Sound Design and Sonification (Hybrid Session). Indiana/Iowa

1:00 3pNS **Noise, Architectural Acoustics, Acoustical Oceanography, Engineering Acoustics, and Underwater Acoustics**: Post Pandemic Soundscapes Part II (Hybrid Session). Chicago C

1:00 3pPAa **Physical Acoustics and Structural Acoustics and Vibration**: Acoustics and Elasticity of Consolidated and Unconsolidated Granular Materials III (Hybrid Session). Chicago F/G

1:00 3pPAb **Physical Acoustics**: Ultrasound: Radiation Force and Spectroscopy. Los Angeles/Miami/Scottsdale

1:30 3pPP **Psychological and Physiological Acoustics**: Auditory Neuroscience Prize Lecture. Chicago A/B

1:00 3pSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, Noise, and Computational Acoustics**: Real World Case Studies for Damping. Chicago H

1:00 3pSP **Signal Processing in Acoustics, Biomedical Acoustics, Underwater Acoustics, Physical Acoustics, and Architectural Acoustics**: Acoustical Imaging, Reconstruction, and Localization III: Biomedical Acoustics (Hybrid Session). Purdue/Wisconsin

1:30 3pUW **Underwater Acoustics: General Topics in Underwater Acoustics**: Calibration and Laboratory Measurements. Michigan/Michigan State

Thursday Morning

8:20 4aAA **Architectural Acoustics, Signal Processing in Acoustics, Computational Acoustics, Physical Acoustics, Structural Acoustics and Vibration, and Noise**: Artificial Intelligence and Machine Learning. Chicago A/B

8:00 4aBAa **Biomedical Acoustics, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics**: Advances in Elastography I. Belmont

8:30 4aBAb **Biomedical Acoustics, Signal Processing in Acoustics, Physical Acoustics**: Making and Using Cavitation Images for Therapeutic Ultrasound I. Armitage

9:00 4aCA **Computational Acoustics, Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, Musical Acoustics, Underwater Acoustics, Engineering Acoustics, and Biomedical Acoustics**: Showcases of HPC Across Acoustics I. Lincolnshire 1/2

8:00 4aED **Education in Acoustics, Physical Acoustics, Architectural Acoustics, and Noise**: When Doing it Right Goes Wrong. Great America 1/2

8:05 4aNSa **Noise**: Aircraft Noise. Chicago C

10:00 4aNSb **Noise and Architectural Acoustics**: Flanking Paths: Finding them, Solving Them and Improvement in ASTC When You Do. Chicago C

8:05 4aPAa **Physical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration**: Acoustic Sensing in Planetary Environments. Chicago F/G

9:00 4aPAb **Physical Acoustics: General Physical Acoustics I**: Time Reversal Technique and Source Characterization. Los Angeles/Miami/Scottsdale

8:00 4aPP **Psychological and Physiological Acoustics and Speech Communication:** Perception Beyond Tones and Speech in Normal and Impaired Hearing: Voice, Emotions, and Music Perception. Indiana/Iowa

10:00 4aSA **Structural Acoustics and Vibration**: General Topics in Structural Acoustics. Chicago H

8:00 4aSC **Speech Communication**: Speech Production I (Poster Session). Salon I

8:30 4aSP **Signal Processing in Acoustics, Biomedical Acoustics, Underwater Acoustics, Physical Acoustics, and Architectural Acoustics**: Acoustical Imaging, Reconstruction, and Localization IV: Underwater Applications (Hybrid Session). Purdue/Wisconsin

8:30 4aUW **Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, Physical Acoustics and Engineering Acoustics**: Acoustic Methods for Unmanned Mobile Platforms: Sensing, Localization and Communications I. Michigan/Michigan State

Thursday Afternoon

1:10 4pAA **Architectural Acoustics, Noise, ASA Committee on Standards, and Engineering Acoustics:** Sound Data for Sound Design. Chicago A/B

1:00 4pAB **Animal Bioacoustics:** Contributions of Expert Subjects to Animal Bioacoustics. Great America 1/2

1:30 4pBAa **Biomedical Acoustics, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics:** Advances in Elastography II. Belmont

1:30 4pBAb **Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics:** Making and Using Cavitation Images for Therapeutic Ultrasound II. Armitage

1:30 4pCA **Computational Acoustics, Structural Acoustics and Vibration, Physical Acoustics, Signal Processing in Acoustics, Musical Acoustics:** Showcases of HPC Across Acoustics II. Lincolnshire 1/2

1:30 4pEA **Engineering Acoustics, Signal Processing in Acoustics, Physical Acoustics, Architectural Acoustics, Underwater Acoustics:** Acoustic Measurement in Extreme Environments. Purdue/Wisconsin

1:30 4pID **Interdisciplinary and Student Council:** Guidance From the Experts: Applying for Grants and Fellowships. Indiana/Iowa

1:00 4pMUa **Musical Acoustics:** Acoustics of Percussion Instruments. Denver/Houston/Kansas City

4:20 4pMUb **Musical Acoustics:** Acoustics of Percussion Instruments Steelpan Ensemble Performance. Los Angeles/Miami/Scottsdale

1:00 4pNS **Noise Engineering Acoustics, Computational Acoustics, Structural Acoustics and Vibration, and Physical Acoustics:** Validation of Environmental Noise Modeling. Chicago C

1:30 4pPA **Physical Acoustics:** Acoustic Propagation Effects. Chicago F/G

1:00 4pPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics Poster Session II. Salon I

1:30 4pSA **Structural Acoustics and Vibration, Education in Acoustics, Physical Acoustics, and Engineering Acoustics:** Tutorial on Structural Acoustics and Vibration. Chicago H

2:00 4pSC **Speech Communication:** Speech Perception II - Bilingualism and Second Language Acquisition (Poster Session). Salon I

1:30 4pUW **Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, Physical Acoustics and Engineering Acoustics:** Acoustic Methods for Unmanned Mobile Platforms: Sensing, Localization, and Communications II. Michigan/Michigan State

Friday Morning

8:00 5aAA **Architectural Acoustics**: Heavy-Handed Recommendations: When Less is More. Chicago A/B

7:30 5aBAa **Biomedical Acoustics**: General Topics in Biomedical Acoustics: QUS and Beamforming. Northwestern/Ohio State

7:30 5aBAb **Biomedical Acoustics**: General Topics in Biomedical Acoustics: Bubbles and More. Purdue/Wisconsin

9:00 5aED **Education in Acoustics, Physical Acoustics, Engineering Acoustics, and Musical Acoustics**: Resources for Teaching Waves in a Physics Class (Hybrid Session). Lincolnshire 1/2

9:00 5aPA **Physical Acoustics**: General Physical Acoustics II: Application, Measurements, and Novel Effects. Chicago F/G

8:00 5aPP **Psychological and Physiological Acoustics and Speech Communication:** Environmental Sounds: Perception, Cognition, Applications. Indiana/Iowa

8:00 5aSC **Speech Communication**: Speech Production II - Speech Articulation | Sociophonetics (Poster Session). Salon I

8:20 5aUW **Underwater Acoustics**: General Topics in Underwater Acoustics. Michigan/Michigan State

184th Meeting of the Acoustical Society of America

The 184th meeting of the Acoustical Society of America will be held Monday through Friday, 8-12 May 2023 at the Chicago Marriott Downtown Miracle Mile Hotel, Chicago, Illinois, USA.

SECTION HEADINGS

1. REGISTRATION
2. TECHNICAL SESSIONS
3. TECHNICAL SESSION DESIGNATIONS
4. EXHIBIT AND EXHIBIT OPENING RECEPTION
5. SOLUTIONS SHOWCASE
6. HOT TOPICS SESSION
7. ACOUSTICS DEMONSTRATION EXTRAVAGANZA
8. PRIZES AND PRIZE LECTURES
9. TECHNICAL COMMITTEE OPEN MEETINGS
10. WOMEN IN ACOUSTICS ROUNDTABLE
11. PROVIDING AND RESPONSING TO CONSTRUCTIVE REVIEWS WORKSHOP
12. AFFINITY LISTENING SPACE
13. PLENARY SESSION AND AWARDS CEREMONY
14. ANSI STANDARDS COMMITTEES
15. COFFEE BREAKS
16. A/V PREVIEW ROOM
17. MOTHERS ROOM
18. SOCIALS
19. SOCIETY LUNCHEON AND LECTURE
20. STUDENT EVENTS: NEW STUDENTS/FIRST-TIME ATTENDEE ORIENTATION, MEET AND GREET, STUDENT CAREER MIXER, STUDENT RECEPTION
21. WOMEN IN ACOUSTICS LUNCHEON
22. JAM SESSION
23. ACCOMPANYING PERSONS PROGRAM
24. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)
25. TECHNICAL PROGRAM ORGANIZING COMMITTEE
26. MEETING ORGANIZING COMMITTEE
27. PHOTOGRAPHING AND RECORDING
28. ABSTRACT ERRATA
29. GUIDELINES FOR ORAL PRESENTATIONS,
30. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS
31. 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 Monday, 8 May, at 7:00 a.m. in Halstead (4th floor) (see floor plans on pages A7 and A8).

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 $795 for members of the Acoustical Society of America; $1095 for non-members, $250 for Emeritus members (Emeritus status pre-approved by ASA), $500 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $200 for ASA Student members, $300 for students who are not members of ASA, $25 for Undergraduate Students, and $250 for accompanying persons.

One-day registration is available at $500 for members and $600 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions). A nonmember who pays the $1095 nonmember registration fee and simultaneously applies for Associate Membership in the Acoustical Society of America will be given a $50 discount off their dues payment for 2023 dues.

Invited speakers who are members of the Acoustical Society of America are expected to pay the Member full-week or one-day registration fees. Nonmember invited speakers who participate in the meeting only on the day of their presentation may register without charge. The registration fee for nonmember invited speakers who wish to participate for more than one day is $500 and includes a one-year Associate Membership in the ASA upon completion of an application form.

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 1200 abstracts.

Floor plans of the Chicago Downtown Marriott appear on pages A7 and A8. 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, 8 May
2-Tuesday, 9 May
3-Wednesday, 10 May
4-Thursday, 11 May
5-Friday, 12 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. EXHIBIT AND EXHIBIT OPENING RECEPTION

An instrument and equipment exhibition will be located in Chicago D/E on the 5th floor and will open on Monday, 8 May, with an evening reception serving a complimentary drink. Exhibit hours are Monday, 8 May, 5:30 p.m. to 7:00 p.m., Tuesday, 9 May, 9:00 a.m. to 5:00 p.m., and Wednesday, 10 May, 9:00 a.m. to 12:00 noon.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

5. SOLUTIONS SHOWCASE

In an effort to provide industry members and supporting companies better visibility at the meetings, ASA is hosting the second Solutions Showcase in Chicago on Monday, 8 May, 1:00 p.m. to 4:00 p.m. in the McHenry room. This will be an opportunity to present a product, service, or solution in a setting similar to a technical session, but without restricting the commercial character of the talk.

6. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 10 May, at 1:00 p.m. in Northwestern/Ohio State.

7. ACOUSTICS DEMONSTRATION EXTRAVAGANZA

The Physical Acoustics Technical Committee and the Committee on Education in Acoustics are pleased to present a showcase of demonstrations and apparati to inspire and challenge your understanding of acoustics phenomena presented in Session 2aPAb on Tuesday, 9 May, at 9:45 a.m. in Chicago A/B. This event will be similar to the "Circus of Acoustics" demonstration show from the June 2002 ASA 143 Pittsburgh meeting [*J. Acoust. Soc. Am.* **111**(5–2), 2451 (2002), session 4pPAb].

8. PRIZES AND PRIZE LECTURES

The 2023 Hartmann Prize in Auditory Neuroscience, Medwin Prize in Acoustical Oceanography, and the Rossing Prize in Acoustics Education will be presented at the Plenary session on Wednesday, 10 May, at 3:45 p.m. in Salon II.

The Auditory Neuroscience Prize Lecture will be presented on Wednesday, 10 May, in session 3pPP at 1:30 p.m. in Chicago A/B. The Acoustical Oceanography Prize Lecture will be presented on Tuesday, 9 May, in session 2pAOa at 1:30 p.m. in Chicago A/B. The 2022 Acoustical Education Prize Lecture will be presented on Wednesday, 10 May, in session 3pED at 1:00 p.m. in Armitage.

9. 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 A5.

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.

10. WOMEN IN ACOUSTICS ROUNDTABLE

A Roundtable sponsored by the Women in Acoustics Committee will be held on Wednesday, 10 May, 2:30 p.m. to 3:30 p.m. in Kane that will include six to eight topics related to mentorship, work-life balance, and navigating careers in academia, industry, and government settings. Topics will be facilitated by volunteers from Women in Acoustics. Attendees will have the opportunity to switch tables/topics at the 30-minute mark.

11. PROVIDING AND RESPONDING TO CONSTRUCTIVE REVIEWS WORKSHOP

This workshop, sponsored by the Member Engagement Committee, is intended for writers and reviewers alike and will be held on Wednesday, 10 May, 9:00 a.m. to 10:30 a.m. It will give examples of scientific writing submissions and reviewer responses for participants to practice writing reviews and responding to them. Experienced authors, reviewers and editors, including James Lynch, Editor in Chief of ASA, will provide assistance and insight.

12. AFFINITY LISTENING SPACE

ASA values providing a safe space for all its members. To facilitate that, the Member Engagement Committee will provide an Affinity Listening Space on Wednesday, 10 May, 5:00 p.m. to 7:00 p.m. in the McHenry room for members of affinity groups to meet and discuss important issues with other ASA members from underrepresented identities. All groups will meet together for some initial networking and then participants will break out into discussion groups specific to their identities.

13. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 10 May, at 3:45 p.m. in Salon II. The Hartmann Prize in Auditory Neuroscience, Medwin Prize in Acoustical Oceanography, the Rossing Prize in Acoustics Education, the R. Bruce Lindsay Award, the Helmholtz-Rayleigh Interdisciplinary Silver Medal, and the Gold Medal will be presented. Certificates will be presented to Fellows elected at the meeting. See page A248 for a list of fellows and award recipients.

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

14. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be held at the Chicago meeting as noted on page A5.

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, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; T: 631-390-0215; E: asastds@acousticalsociety.org

15. COFFEE BREAKS

Morning coffee breaks will be held daily from 9:30 a.m. to 11:00 a.m. and an afternoon break will be held on Tuesday from 2:00 p.m. to 3:00 p.m. in Chicago D/E on the 5th floor.

16. A/V PREVIEW ROOM

The A/V preview room will be available on Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 3:00 p.m in Minneapolis (6th floor).

17. MOTHERS ROOM

A Mothers Room for ASA meeting attendees will be available Monday to Friday, 8-12 May, in the 7th floor registration office. The hours are Monday to Thursday, 8:00 a.m. to 5:00 p.m. and Friday, 8:00 a.m. to 12:00 noon.

18. SOCIALS

Socials will be held on Tuesday and Thursday evenings, 6:00 p.m. to 7:30 p.m. in Salons II and III on the 7th floor.

The ASA hosts these social hours to provide 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. A second goal of the socials is to provide a sufficient meal so that meeting attendees can attend the open meetings of Technical Committees that begin immediately after the socials.

19. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 11 May, at 12:00 noon in Salon III.

This luncheon is open to all attendees and their guests. Purchase your tickets at the Registration Desk before 10:00 a.m. on Thursday, 11 May. The cost is USD $40.00 per ticket.

20. 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, 8 May, from 5:30 p.m. to 6:00 p.m. in Chicago A/B. This will be followed by the Student Meet and Greet from 6:00 p.m. to 7:30 p.m. in Los Angeles/Miami/Scottsdale where refreshments and a cash bar will be available.

The Student Career Mixer will be held on Tuesday, 9 May, 11:30 a.m. to 2:00 p.m. in Salon III. 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 Wednesday, 10 May, from 6:00 p.m. to 8:00 p.m. in Salon III. 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.

21. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 10 May, in Salon III. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Wednesday, 10 May. The fee is USD $35 for non-students and USD $15 for students.

22. JAM SESSION

You are invited to Salon II on Wednesday night, 10 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.

23. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Chicago meeting. The on-site registration fee for accompanying persons is USD $250. A hospitality room for accompanying persons will be open in Navy Pier (10th floor), 8:00 a.m. to 10:00 a.m. Monday through Friday. This entitles you access to the accompanying persons room, socials on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

24. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Chicago 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.

25. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Brandon Cudequest, Technical Program Chair; Christopher Bassett, Acoustical Oceanography; Shane Guan, Animal Bioacoustics; Brandon Cudequest, David Manley, Architectural Acoustics; Kang Kim, Libertario Demi, John Cormack, Biomedical Acoustics; Amanda Hanford, Computational Acoustics; Daniel Russell, Education in Acoustics; Thomas Blanford, Engineering Acoustics; Kurt Hoffman, Stephen Tufte, Musical Acoustics; Aaron Vaughn, James Phillips, Hales Swift; Noise; Ralph Herman, Joel Lonzaga, Physical Acoustics; Gregory Ellis, Virginia Best, Psychological and Physiological Acoustics; Trevor Jerome, Signal Processing in Acoustics; Pasquale Bottalico, Matthew Masapollo, Benjamin Tucker, Kelly Berkson, Speech Communication; Anthony Bonomo, Stephanie Konarski, Structural Acoustics and Vibration; David Dall'Osto, Underwater Acoustics; Zane Rusk, Student Council.

26. MEETING ORGANIZING COMMITTEE

Shane Jerome Kanter, Chair; Brandon Cudequest, Technical Program Chair; Jennifer Nelson Smid, Laura Brill, Signs.

27. PHOTOGRAPHING AND RECORDING

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

28. ABSTRACT ERRATA

This meeting program is Part 2 of the April 2023 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*.

29. 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.

30. 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.

31. 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, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; Telephone: 516-576-2360; E-mail: asa@acousticalsociety.org

185th Meeting, joint meeting with the Australian Acoustical Society, WESPAC, PRUAC, Sydney, Australia, 4-8 December 2023

186th Meeting, Ottawa, Canada 13-17 May 2024

187th Meeting – Virtual Meeting, fall 2024

188th Meeting – joint with the International Congress on Acoustics, New Orleans, Louisiana 19-23 May 2025.

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Session 1aAA

Architectural Acoustics, Signal Processing in Acoustics, and Computational Acoustics: Application of Statistical and Computational Methods in Building Acoustics

Benjamin M. Shafer, Cochair
PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Samantha Rawlings, Cochair
Veneklasen Associates, 1711 16th Street, Santa Monica, CA 90404

Invited Papers

8:00

1aAA1. Noise in your web browser: An online listening study regarding the perceived annoyance due to impact sounds. Markus Mueller-Trapet (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, Markus.Mueller-Trapet@nrc-cnrc.gc.ca), Sabrina Skoda (none, Duesseldorf, Germany), Young-Ji Choi (Architectural Acoust. Lab, Kangwon National Univ., Gangwon-do, South Korea), Iara Batista da Cunha, and Jeffrey Mahn (National Res. Council Canada, Ottawa, ON, Canada)

Laboratory listening studies are typically limited in the number of people who can participate because of the effort that is involved in setting up and carrying out the experiment for each participant. This effort has significantly increased since the start of the COVID-19 pandemic due to the heightened public health requirements for in-person studies. This increase in effort and the resulting limit on the number of participants can be avoided by implementing the listening study as an online interface, where participants can then run the experiment from the comfort of their home and anyone who has access to a computer and headphones is able to participate. Such an approach supports the goal of involving the general public, who are the ultimate target audience for the research outcome. This contribution presents the preliminary results of an online listening study assessing the perceived annoyance due to impact sounds in residential buildings. The interface was previously validated and presented (Internoise 2021) with limited data. Since then, the survey has been published online for worldwide access. The results are discussed in relation to the results of previous laboratory studies.

8:20

1aAA2. Using a simulation model to guide a new measurement method for floor ceiling assembly impact performance. Sunit Girdhar (Mech. Eng., Michigan Technolog. Univ., Michigan Tech University, Houghton, MI 49931, sgirdhar@mtu.edu), Andrew Barnard (Acoust., Penn State, University Park, PA), Jason R. Blough (Mech. Eng., Michigan Technolog. Univ., Houghton, MI), John Lo Verde, and Wayland Dong (Veneklasen Assoc., Santa Monica, California, CA)

The impact performance of a floor-ceiling assembly is calculated as a single number rating using the ASTM standard. The radiated SPL in the receiving room below is measured, either at discrete points or by scanning in parts of the room, and compared to a reference curve. This method suffers from non-reproducibility issues, especially in low frequencies where the sound field is non-diffuse and SPL measurements may give a large error for different locations in the room. A new method is needed that improves the reproducibility of the measurement method. We are using a simulation model to guide a new measurement method that will have a higher reproducibility. In this presentation, we will discuss some of the advances made so far and discuss the future scope of this work. This presentation provides an update on the long-term project of proposing a new measurement method for impact noise in floor-ceiling assemblies.

8:40

1aAA3. The effect of loudspeaker type and position on field airborne sound isolation measurements. Wayland Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com) and John Lo Verde (Veneklasen Assoc., Santa Monica, CA)

Airborne sound isolation measurement per ASTM E336 uses random noise excitation generated by loudspeakers in the source room. For smaller rooms such as those in residences, loudspeakers are typically located in and pointed into a corner. There is some indication that this loudspeaker position results in systematic reductions in the airborne sound isolation. To investigate the reported reductions and other possible effects due to loudspeaker characteristics, we performed a gauge repeatability and reproducibility study (GRR) on a typical demising wall between units in a multi-family residential building. A GRR uses analysis of variations (ANOVA) on an appropriately designed experiment to separate and quantify the components of the overall uncertainty. This adds to the GRR analyses previously completed (see papers by the authors at Internoise 2018, 2019, and 2021). This study used loudspeaker type, position, and level as the study variables. The results are presented, and the effect on noise isolation testing procedures is discussed.

9:00

1aAA4. Statistical evaluation of noise due to changes in pandemic traffic patterns. John Lo Verde, Wayland Dong (Veneklasen Assoc., Santa Monica, CA), Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), and McCall Edwards (Veneklasen Assoc., Santa Monica, CA)

Vehicular traffic noise for free-flowing roadways and highways generally follows statistically predictable flow rates and vehicular mix. The authors' earlier work in this area examined long-term variations in traffic noise level with the purpose of establishing maximum hourly levels for vehicular sources ("Defining vehicular noise levels to manage risk associated with exterior façade design," LoVerde, Dong, Rawlings, Internoise 2014 Melbourne) and general average sound level ("Noise prediction of vehicle sources on arterials using measured sound data," LoVerde, Dong, Rawlings, ASA 2014 Providence; "Methods for estimating the variance in traffic noise distribution from short-duration measurements," LoVerde, Dong, Rawlings, ICSV 2015 Florence). When the COVID-19 pandemic resulted in lockdowns, traffic patterns at the measurement location in Southern California were affected. The authors' examination of traffic data revealed that the changes in traffic patterns did not affect noise level significantly, but variability in sound level across the 24-h period increased substantially ("Changes in statistical traffic noise descriptors during COVID-19," LoVerde, Dong, Edwards, Rawlings, ASA 2021). For this paper, the authors have undertaken comparison of pre-, during-, and post-lockdown vehicular traffic noise levels over a several-month period for the purpose of understanding how measurement of traffic noise is implemented and interpreted.

Contributed Papers

9:20

1aAA5. Assessing the acoustic impedance of indoor wall surfaces through a phase array sensor. Cengiz Yilmazer (CSY R&D and Architecture Eng., Ankara, Turkey), Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Faculty of Art, Design and Architecture, Dept. of Interior Architecture and Environ. Design, Ankara 06800, Turkey, semiha@bilkent.edu.tr), and Ela Fasllija (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

Reliable data on the acoustic absorption behavior of building materials are crucial for predicting and, therefore, designing desired acoustic conditions in the built environment. The existing methods, such as impedance tube and reverberation room, might not always be feasible due to their own respective limitations such as angle of incidence, large materials samples, and reproducibility issues. In this light, this study is concerned with incorporating a method based on a phased array sensor to predict the random acoustic impedances of an indoor wall surface. Considering multipath propagation in a reverberant field, it focuses on accurately designing the array shape and the sensor number. As a case study for validation, it uses parallel arranged micro-perforated panels (MPP) placed as a wall finishing material in a $10.2\text{m} \times 5.47\text{m} \times 3.6\text{m}$ sized room. Comparisons between theoretical and predicted results through the implemented method show a fair agreement. The material was also previously assessed in the impedance tube for its normal absorption coefficient. Further comparisons between results obtained through geometrical acoustics software and impedance tube are also discussed for their discrepancies regarding the proposed method.

9:35

1aAA6. Parametric analysis of the impact of geometry on room modes at low frequencies using wave-based simulation. Alaa Algargoosh (Media Lab, Massachusetts Inst. of Technol. (MIT), 75 Amherst St., Cambridge, MA 02139, alaas@mit.edu) and Ahmed Allam (Media Lab, Massachusetts Inst. of Technol. (MIT), Cambridge, MA)

Room modes at low frequencies strongly influence the perception of sound in a room, which can enhance or deteriorate the acoustic experience. Previous research showed that low-frequency modes are also linked to the emotional impact of the acoustic environment; this link calls for further understanding of the effect of resonance at low frequencies on sound perception. However, most auralization software programs adopt geometric-based simulations, which are not accurate at low frequencies. In this work, we adopt wave-based simulations to characterize room modes at low frequencies depending on the room geometry through a parametric analysis. The study outlines the contribution of architectural elements, such as domes and sub-volumes, in shaping the acoustic environment by manipulating low frequencies.

9:50–10:05 Break

10:05

1aAA7. Application of active noise control for plenum window in a domestic setting. Johann K. Tan (Dept. of Architecture, National Univ. of Singapore, 4 Architecture Dr., Singapore 117566, Singapore, johann.tan@nus.edu.sg) and Siu-Kit LAU (Dept. of Architecture, National Univ. of Singapore, Hong Kong, Hong Kong)

Plenum windows, also known as ventilation windows allow for natural ventilation in living spaces while providing noise reduction to improve the health and living comfort of occupants, especially in noise polluted and tropical/sub-tropical cities. However, while plenum windows can provide acceptable levels of natural ventilation, the passive noise performance of plenum windows when fully opened is still lacking compared to the conventional closed windows at low frequencies. Active noise control (ANC) has been shown to provide good performance in several applications in low-frequency control. Thus, the application of active noise control through a single-channel and dual-channel ANC implementation is to be developed and demonstrated in a field test for a plenum window set in a domestic home built to reproduce the environment of a high-rise apartment. The experimental application will be based on the optimal configurations for single and dual-channel ANC informed by prior simulation studies.

10:20

1aAA8. Treating seat-dip with acoustic metamaterials and detuned acoustic resonators. Alex Maurer (JBL Loudspeakers, Harman, 60 Charlesgate E, Boston, MA 02215, alexander.maurer@intertek.com)

As sound passes over regularly spaced audience seating areas in performance spaces at a grazing angle, the low frequency response of performing arts venues and auditoria is altered. This is due to diffraction based on the regular spacing of the seating. One effect is what is referred to as seat-dip attenuation, which is a reduction in low frequency energy across a bandwidth. Being a low-frequncy phenomenon, it is difficult to treat using conventional acoustic materials such as porous absorbers or Schroeder diffusers. Modern acoustic metamaterials can alter wave propagation using relatively thin layers. This project investigates whether acoutic metamaterials applied to the tops of the seats, the rear of the seats, or the floor can reduce the low-frequency attenuation brought about by the seat-dip effect. This novel approach demonstrates varying levels of succes for remediating low-frequency seat-dip attenuation. The study is carried out using a computer model that is evualted using boundary element method (BEM) in COMSOL Multiphysics. Additionally, a novel approach for reducing the prominence of the seat-dip effect is explored using detuned Helmholtz resonatos placed at the top of the seats.

10:35

1aAA9. Comparing acoustical worship ambience of churches built by Spanish Franciscans: Nossa Senhora do Pilar (Goa) and Mission Concepcion (Texas). Menino A. Tavares (Acoust. of Revitalising Environment, Society of Pilar, Pilar Deepti Sadan, Institutional Zone, P.O.Alto Porvorim, Dist. North Goa, Porvorim, Goa 403521, India, allan.wholysound50@gmail.com) and Susan Wiseman (retired, Waco, TX)

Spanish Franciscan Friars established Nossa Senhora do Pilar" church at Pilar (Goa) in 1613 and Mission Concepción" at San Antonio (Texas) in 1711. This cross-cultural comparative acoustical study of these churches connects sound decay measures (ISO 3382) with auditory, affective, and meta-physical perceptions (complementing ISO/TS 12913-1/2/3). At $p \leq 0.05$, Mission Concepcion" was found more reverberant than Pilar; sacred music from the choir-loft" of Mission Concepcion" was perceived louder" and clearer" than from the nave floor; whereas, in Pilar," sacred music from the nave floor" was perceived louder" than from the choir-loft; perception of pleasantness" and eventfulness" in both churches showed no significant difference for sacred music from either source; sacred music from the choir-loft" of Mission Concepcion" evoked better perception of heightened awareness," stillness," and inspiration; perceived loudness" of sacred music from the choir loft" in Pilar showed positive correlation with RT60, D50, TS, and STI and negative correlation with C80; perception of inspiration" by sacred music from the choir loft" in Pilar" was positively correlating with RT60, TS, and STI and negatively correlating with C80 and D50. This study identified character defining acoustical differences between Pilar" and Mission Concepcion" which accentuate each of this historic church as a unique living heritage.

10:50

1aAA10. Experimental study of instantaneous sound intensities in rectangular enclosures. Daniel Tay (Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, daniel-tay@uiowa.edu) and Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Instantaneous and time-averaged intensities are relevant field quantities in architectural acoustics investigations. Modeling of intensity flows in coupled volumes and the effect of absorptive material on its trajectory in reverberation chambers have been studied recently. This work focuses on vortical energy flows with a derivation of instantaneous intensity in enclosed spaces and an experimental effort for investigating the changes over time in all field quantities of interest—instantaneous intensity, pressure, and velocity simultaneously. In this study, the scale modeling technique is applied to investigate instantaneous intensity flows experimentally measured using pressure-3D velocity sensors in a rectangular room. Experimental results suggest the presence of vortical intensity modes at specific frequencies correlated to the room dimensions. The resulting simulation of intensity vortices has potential applications in the acoustical design and analysis of complex room geometries where sound energy flows play a major role in the acoustics of the space such as that of coupled-volume spaces and small rooms for critical listening.

11:05

1aAA11. Using crowd-sourced data to optimize conversational sound levels in hospitality venues. Erin L. Dugan (ELD Acoust., PO Box 139, Greeley, PA 18425, eld.acoustics@gmail.com)

One of the top complaints about a restaurant is usually related to noise. The sound level in a venue can significantly affect whether or not a customer will return. Statistical analysis of the crowdsourced data collected with the SoundPrint smartphone app at more than 10,000 U.S.-based hospitality venues (restaurants, coffee shops, and bars) provides a comparison of the objective sound level data versus the patrons' subjective ratings for conversational ability. These insights show a customer's tolerance for noisy environments can be highly dependent on the venue type, specifically when communication is important. The findings can also help venue managers know what interior sound levels to target when optimizing the acoustics for their venue, whether by reducing background noise levels or improving acoustical treatments, in order to attract and keep their patrons happy.

11:20

1aAA12. Assessing cognitive effects of transportation noise on office workers with physiology and task performance. Heui Young Park (The Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hkp5188@psu.edu), Michelle C. Vigeant (The Pennsylvania State Univ., University Park, PA), and Andrew Dittberner (GN Group, GN Adv. Sci., Bloomington, MN)

Noise typically present in offices can negatively impact workers' mental workload and task performance. This study investigated how transportation noise in open offices when windows are open can influence workers' task performance, physiology (brain activity, heart rate variability, and skin resistance), and subjective ratings, as previous literature works often show inconclusive results or use unrealistic, simulated acoustic conditions. Subjects were given two cognitive tasks, mental arithmetic and digit span, while wearing three physiological measurement devices: an electroencephalography (EEG) headset, electrocardiogram (ECG) sensors, and electrodermal activity (EDA) sensors. After a baseline measurement, both the physiological markers and task performance were recorded for a training session, a quiet condition, and two transportation noise conditions with sound pressure levels of 50 and 70 dBA, respectively. Recorded rail noise was auralized in an anechoic chamber with a 32-loudspeaker array to simulate transportation noise heard when sitting in an office with an open window. Subjective rating questions regarding the acoustic environment and self-perception of mental workload and performance were administered after each acoustic condition. The effects of different acoustic conditions on cognitive task performance, physiological data, and subjective ratings were analyzed with a repeated measures MANOVA. Findings from this study will be presented.

11:35

1aAA13. The effects of speech intelligibility on reading comprehension under task-switching behaviour in open-plan offices. Yuanyuan Zhang (School of Architecture, Huaqiao Univ., 668 Jimei Ave., Jimei District, Xiamen City, Fujian Province, Xiamen, Fujian 361021, China, 253908074@qq.com), Siu-Kit LAU (College of Design and Eng., National Univ. of Singapore, Hong Kong, Hong Kong), Dayi Ou (School of Architecture, Huaqiao Univ., Xiamen, Fujian, China), and Guanhua Qu (College of Design and Eng., National Univ. of Singapore, Singapore, Singapore)

Numerous investigations showed that speech noise is the most disturbing factor in work efficiency. Work efficiency decreases with speech intelligibility. The results of a recent study show that the effect of speech intelligibility on work efficiency is significantly correlated with the effect of speech intelligibility on reconcentration. To examine the effects of speech intelligibility on reading comprehension under task-switching behaviour, a total of 30 students were recruited to participate reading comprehension test, which is interrupted by a creative task. Task performance, psychology, and physiology indicators were carried out under eight experimental conditions with speech transmission index = 0.00, 0.17, 0.26, 0.36, 0.45, 0.52, 0.57, and 0.71. The experimental results showed that the task-switching behaviour had no evidence to support a significant impact on task performance for each speech intelligibility. However, task-switching behaviour significantly impacts participants' fatigue and workload as the value of the speech transmission index increases.

Session 1aAB

Animal Bioacoustics, Acoustical Oceanography, Signal Processing in Acoustics, Underwater Acoustics, and Noise: Climate Change and Sound: How the Sound of the Planet Reflects the Health of the Planet I (Hybrid Session)

Laura Kloepper, Cochair
Department of Biological Sciences, University of New Hampshire, 230 Spaulding Hall, Durham, NH 03824

Megan Ballard, Cochair
University of Texas at Austin, 10000 Burnet Road, Austin, TX 78665

Edward J. Walsh, Cochair
VA Loma Linda Healthcare System, Loma Linda, CA 92357

Chair's Introduction—8:00

Invited Papers

8:05

1aAB1. Passive acoustic monitoring of biological soundscapes in a changing climate. Lauren A. Freeman (NUWC Newport, 1176 Howell St., Newport, RI 02841, lauren.a.freeman@navy.mil), Daniel Duane (NUWC Newport, Newport, RI), and Simon Freeman (Dept. of Energy, ARPA-E, Washington, DC)

Passive acoustic monitoring of biological soundscapes offers a long-term view into ecosystem state. This is particularly well studied for coral reefs and tropical littoral systems with evidence for similar capability in temperate and deep ocean biologically rich ecosystems. Monitoring ecosystem state under both climate change impacts and changing human usage is a critical piece of understanding how climate change and human use impact ecosystems. Passive acoustics allow for wide area coverage of an ecosystem heartbeat, and changes in key bioacoustic metrics in coral reefs indicate shifts from healthy coral dominated systems to more degraded systems with increased macroalgal cover. These shifts are typically associated with increased ocean temperatures and/or increased human use. The primary controls on coral reef soundscapes are time of day and year. Here, broad comparisons between warmer and cooler years at long term coral reef monitoring sites in Hawaii will be discussed, as well as summer versus winter biological soundscapes at temperate sites in New England. Reef soundscapes encompass contributions from a wide variety of marine flora and fauna, some of which can be identified to species level through characteristic calls and tracked with a high degree of fidelity through passive acoustics alone.

8:25

1aAB2. Estimation of rain amount over global oceans using ambient sound recorded on Argo floats relating to global water cycle studies. 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), and Eric I. Thorsos (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Rain over the ocean is a central process in the global freshwater cycle. The freshening of the ocean surface due to rain can be seen in global maps of sea surface salinity, where areas with high precipitation are broadly coincident with areas of low salinity and regions of high salinity occur in regions with low precipitation and high evaporation. The transfer of water from these evaporating regions to the precipitating regions drives the global water cycle, and understanding its dynamics is essential in determining how weather patterns will respond to changes in global climate. Rain is difficult to measure over the ocean due to its spatial and temporal variability and the limitations imposed by rain gauges when mounted on moving platforms. However, the loud and distinctive underwater sound generated by raindrops on the ocean surface can be used to detect and quantify rainfall and to track climate change impacts. The work here will focus on deriving time series of rain rate with temporal resolution on the order of minutes using ambient noise recorded by Passive Aquatic Listener (PAL) on both Argo floats and deep ocean moorings (Nystuen, J., Atmos. Oceanic Tech, **13**, 74–84, 1996). [Work supported by NOAA and NASA.]

8:45

1aAB3. Combining passive acoustic monitoring and species distribution models to monitor species responses to climate change. Marconi Campos-Cerqueira (Sci., Rainforest Connection, PO BOX 1180, Barranquitas 00794, Puerto Rico, marconi@rfcx.org)

Climate change is altering the spatial distribution of many species worldwide, even in pristine and protected areas. Therefore, we need to identify and protect suitable areas for a large proportion of the fauna so that they persist through time. Here, we combined passive acoustic monitoring, semi-automatic species identification models, and species distribution models to document shifts in bird and frog distributions along an elevational gradient. In addition, we used acoustic data from 674 sites from Puerto Rico's main island to create species distribution models based on past (1980–1989), present (2005–2014), and future (2040–2060) climate scenarios to determine how species distributions relate to the current distribution of protected areas. Our results suggest that species are shifting toward high-elevation areas. We also showed that Puerto Rico is projected to become dryer by 2040–2060, and precipitation in the warmest quarter was among the most important variables affecting bird distribution across the entire island. In addition, a large portion of always-suitable areas for birds is outside of protected areas (>75%), indicating that present protected areas will not suffice to safeguard bird species under climate change. We must be creative and proactive in protecting species in climate change scenarios.

9:05

1aAB4. Monitoring the ocean climate with acoustics. Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Jahnebakken 3, Bergen 5007, Norway, Hanne.Sagen@nersc.no), Matthew Dzieciuch (SIO, UCSD, La Jolla, CA), and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

Underwater acoustics has developed into a powerful tool for monitoring the climate of ocean basins. Acoustic travel-times are a direct indication of ocean temperature and give a large-scale average of heat content. A recent experiment called the Coordinated Arctic Acoustic Thermometry Experiment (CAATEX) is a prime example of the method in a region known to be experiencing the most rapid response to anthropogenic forcing. In CAATEX, two 35 Hz acoustic transceiver moorings, one in the Nansen Basin and one in the Beaufort Sea, were deployed along with four other receiving moorings. The travel-times between these moorings provide an accurate baseline of the environment during the 2019–2020 deployment year that can be compared to past, present, and future measurements and climatologies. Furthermore, other acoustic observables, such as transmission loss and the ambient soundscape, provide important insight into the ice conditions and the changing environment.

9:25

1aAB5. Unraveling the link between sediment acoustics and organic carbon. Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 8 College Rd., Durham, NH 03824, g.venegas@unh.edu), W. Cyrus Clemo, Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Abdullah F. Rahman (School of Earth, Environ., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX), Kevin M. Lee, Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), and Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The top meter of marine sediments is estimated to store a total of 2322 Pg of carbon [Atwood *et al*., 2020. Front. Mar. Sci., vol. 7 p. 165], which is twice that of terrestrial soils. Physical disturbance and remineralization of these carbon stocks could further accelerate climate change. Rapid and accurate quantification of sediment carbon stocks can (1) better inform carbon budgets and management of human activities in the ocean to minimize carbon remineralization and (2) monitor changes in sediment carbon stocks due to anthropogenic and natural disturbances. Sediment acoustics, which have been linked with sediment total organic carbon/content in mud banks, seagrass beds, and estuarine environments, can be measured *in situ* and offer a scalable solution toward rapid estimation of carbon stocks. However, before this modality can be applied broadly, a fundamental understanding between sediment acoustic properties and sediment organic/inorganic constituents must be realized. Here, direct *ex situ* measurements of sediment acoustic properties, sediment organic carbon, and grain size distribution will be compiled and compared across a variety of sediment types and locations from our own measurements and from published datasets. The development of a non-site-specific constitutive-based relationship will be discussed. [Work supported in part by ONR.]

9:45–10:00 Break

10:00

1aAB6. Odontocete foraging vocalizations during various Cold Pool states in the Bering Sea. Kerri D. Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), and Jennifer J. Johnson (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Cold Pool is a subsurface layer (<2°C) that is formed in the summer from stratification and is characterized by previous winter conditions. This bottom layer of relatively cooler temperatures impacts zooplankton dynamics, driving two different energetic pathways. The cold regime pathway favors larger zooplankton species, which increases forage fish biomass while the warm regime pathway favors smaller zooplankton species, which decreases forage fish biomass. Since odontocetes rely on forage fish for survival and use echolocation to find them, tracking foraging vocalizations could lead to a better understanding of the implications of a variable Cold Pool on the Bering Sea's food web. The Cold Pool was more variable in warm regime years (2014–2017) relative to cold regime years (2007–2013) and was absent in 2018 at this study site. Concurrently, acoustic data with odontocete vocalizations were collected. This presentation will discuss whether odontocete vocal presence is related to Cold Pool dynamics and if odontocete foraging behavior changes as a function of Cold Pool variability. Since a reduced presence of the Cold Pool could occur more often with climate change, it is important to monitor the higher trophic levels' foraging activities to predict what ecosystem changes could occur in the future.

Contributed Papers

10:20

1aAB7. Climate-induced changes in temperature, salinity, and sound speed in the global ocean. Madusanka Madiligama (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, Dept. of Phys. and Astronomy, 108 Lewis Hall, University, MS 38677, mbabeyko@go.olemiss.edu) and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

Oceans play a significant role in climate change by absorbing the earth's excess heat and transporting heat from one location to another. Melting and reducing ice coverage, rising sea levels, ocean acidification, and changes in underwater sound propagation are a few impacts of global warming on oceans. Marine lives have been threatened by many of these alterations, such as interference of their communications and reduction of their habitable places, resulting in the ocean ecosystem imbalance. Ocean datasets containing temperature and salinity are necessary to measure the warming rates and places, access the magnitude and extent of the impacts, and consider mitigation policies. Data collected by traditional *in situ* measurement techniques are insufficient for this purpose because of the scarcity of the measurements in the vast ocean. Here, an idea to overcome the limitation is proposed by using machine learning to infer the ocean temperature and salinity from satellite-observed sea surface temperature and salinity data that are broadly available. Following the predicted temperature and salinity values, global warming indicators, such as ocean heat content and mixed layer depth, are examined. Also examined are the underwater sound speed variations.

10:35

1aAB8. I know what you did last winter: Bowhead whale anomalous winter acoustic occurrence patterns in the Beaufort Sea, 2018–2020. Nikoletta Diogou (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, Victoria, BC V8P 3E6, Canada, niki.diogou@gmail.com), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Xavier Mouy (Passive Acoust. Res. Group, NOAA, Miami, FL), Andrea Niemi (DFO, Winnipeg, MB, Canada), and Stephen Insley (Wildlife Conservation Society Canada, Whitehorse, YT, Canada)

Global warming is affecting the Arctic at a faster pace than the rest of the world, causing an urgent need to monitor ecosystems to detect possible climate-related changes. To this end, five passive acoustic datasets were recorded in the southern Amundsen Gulf (eastern Beaufort Sea) from September 2018 to September 2020 and analyzed for Bering-Chukchi-Beaufort (BCB) bowhead whale calls using a combination of automated and manual detections. Results indicate a large variation in bowhead occurrence patterns between the two years. For 2018–2019, we obtained the first evidence of bowheads overwintering in what is typically their summer foraging ground. Examination of the following year's recordings sheds light on whether this interruption in bowhead annual migration was an anomaly or part of an ongoing phenological shift due to climate change. Time series of remotely sensed sea ice concentration at the study area were considered over the last seven years in interpreting differences in migratory behavior of the whales. Statistical quantification of seasonal patterns and habitat preferences of bowheads, based on the 2018–2019 acoustic data, are presented to provide context to BCB bowhead ecology. Passive acoustic monitoring is an indispensable tool in discerning whale responses to a changing ocean in the Arctic.

10:50

1aAB9. Ocean acoustics and the UN decade of ocean science for sustainable development. Kyle M. Becker (Office of Naval Res., 875 N Randolph St., Arlington, VA 22203, kyle.becker1@navy.mil), Heather Spence (Water Technologies Power Office, US Dept. of Energy, Washington, DC), and Grace Smarsh (AAAS S&T Policy Fellowship Program, Arlington, VA)

The United Nations Decade of Ocean Science for Sustainable Development (Ocean Decade) was initiated in 2021 and runs until 2030. The Ocean Decade seeks transformative ocean science solutions that connects people to our oceans to bring about positive change. This motivated an idea that ocean acoustics has a role to play among the larger ocean sciences as they relate to climate change and the emerging blue economy. On World Ocean Day 2021 (June 8), the Ocean Decade Research Programme on the Maritime Acoustic Environment (OD-MAE) was included among the first Ocean Decade actions endorsed by the United Nations Intergovernmental Oceanographic Commission of UNESCO (IOC). Inspired by Lindsay's "wheel of acoustics," the OD-MAE program is envisioned as a hub for coordinating studies involving rigorous and principled used of sound to address questions relating to all aspects of ocean science and engineering, development, policy, and management. The program seeks to support the development of both people and capabilities that enable a quantitative linkage between an acoustic environment and the physical and biological components and processes occurring within that environment. This presentation will introduce the OD-MAE program, describe some of the initiative underway within it, and provide information on how to get involved.

Invited Papers

11:05

1aAB10. Temporal dependence of acoustic propagation in a seagrass meadow over diurnal and annual timescales. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kyle Capistrant-Fossa (Marine Sci. Inst., The Univ. of Texas at Austin, Port Aransas, TX), Andrew R. McNeese, Colby W. Cushing, Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kenneth H. Dunton (Marine Sci. Inst., The Univ. of Texas at Austin, Port Aransas, TX)

Seagrasses serve major ecological roles in biodiversity promotion, coastal protection, and nutrient cycling. Furthermore, seagrasses have been proposed as a nature-based solution to mitigate effects of climate change due to their capacity for sequestering marine carbon. Current global estimates of seagrass coverage are uncertain; therefore, developing improved methods to assess seagrass coverage and rates of decline are critical to promote sustainable seagrass conservation efforts. Acoustic propagation in seagrass meadows is highly sensitive to oxygen bubble production via photosynthesis and gas volumes encapsulated within seagrass tissue, both acting as biophysical markers. This paper discusses an acoustic method to monitor seagrass oxygen production and biomass with high temporal resolution and over long time-scales. An 18-month acoustic propagation experiment was conducted in a seagrass meadow located in a shallow bay on the Texas Gulf of Mexico Coast. A piezoelectric sound source transmitted broadband frequency-modulated chirps (0.5–100 kHz) every ten minutes, and the signal was measured on horizontal hydrophone array. Dissolved oxygen, photosynthetically active radiation, water temperature, salinity, and depth were concurrently measured with oceanographic probes. Additionally, cores were collected for point-estimates of seagrass biomass. Our work demonstrates that acoustic propagation offers a valuable alternative to experimental measurements of photosynthesis. [Work sponsored by NSF.]

11:25

1aAB11. Decadal scale trends in temperature, salinity, and sound speed in the Beaufort Sea since 1976. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu) and John E. Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA)

Profiles of conductivity and temperature as a function of depth (CTD) were collected in the Beaufort Sea during two U.S. Navy Ice Exercises in 2016, 2018, and 2020 (ICEXYY) in the month of March. The data show a significant departure from climatological values and significant variance at the interface between the Arctic Surface Layer (ASL) and the underlying Pacific Summer Water (PSW). Comparisons to older ICEX data collected over a 45 year period and to ice-tethered profiler (ITP) data since 2007 confirm both the subsurface warming trend and the variance at the ASL-PSW interface. Statistical analysis reveals a 0.4 degreeCelsius per decade increase in the near-surface temperature maximum (NSTM) near 75 m, which establishes two persistent acoustic propagation features beginning in the year 2000—a near-surface acoustic duct above 50 m and a subsurface acoustic duct, known as the Beaufort Lens, centered on 150 m water depth.

MONDAY MORNING, 8 MAY 2023 BELMONT, 8:00 A.M. TO 11:40 A.M.

Session 1aBAa

Biomedical Acoustics: Ultrasound Brain and Super-Resolution Imaging I

Chengzhi Shi, Chair
GWW School of Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Dr NW, Atlanta, GA 30332-0001

Chair's Introduction—8:00

Invited Papers

8:05

1aBAa1. Motion model ultrasound localisation microscopy: Preclinical findings and clinical challenges. Fabian Kiessling (Experimental Molecular Imaging, RWTH Aachen Univ., Forckenbeckstrasse 55, n.a., Aachen 52074, Germany, fkiessling@ukaachen.de)

Superresolution ultrasound represents a major breakthrough for noninvasive tissue characterization. It mainly comprises ultrafast and motion model driven approaches. This talk will focus on motion model Ultrasound Localisation Microscopy (mULM), an approach working at frame rates of most clinical ultrasound devices. It realizes vascular tracks by subwavelength localization of microbubbles and estimating their probability to have moved from one voxel to another. mULM reliably depicts vascular pattern of different tumors and provides functional vascular characteristics that cannot be obtained by other imaging modalities. We will also report on our experiences in clinically translating mULM and applying it to track breast cancer response to neoadjuvant chemotherapy. Currently, we are exploring mULM for other indications, i.e., chronic kidney diseases preclinically and clinically. Here, it became obvious that microbubble dose and injection speed must be carefully adapted to each application to ensure that vascular architectures are sufficiently displayed and to avoid microbubble overload in the images, which complicates correct assignments of microbubbles to tracks. Motion compensation is another challenge for clinical translation, potentially demanding 3D transducers. Thus, we are highlighting the significant potential of superresolution ultrasound and pinpointing challenges that need to be addressed to make it a robust and reliable clinical tool.

8:25

1aBAa2. Small animal cerebral microvascular imaging with super-resolution ultrasound: Techniques and applications. Pengfei Song (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave., Beckman Inst. 4041, Urbana, IL 61801, songp@illinois.edu)

Super-resolution ultrasound (SR-US) is an emerging microvascular imaging technology that provides a micron-scale spatial resolution with tens of millimeters of depth of imaging penetration. The unmatched combination of high spatial resolution and deep imaging penetration opened new doors for many brain imaging applications that benefit from cerebrovascular biomarkers. Recently, SR-US has found its niche in small animal brain applications thanks to its unique capabilities of extending the optical-imaging-level spatial resolution to subcortical, deep brain regions. In this presentation, I will first introduce existing SR-US imaging techniques that are tailored to small animal brain imaging applications. Topics include fast SR-US methods, 3D SR-US imaging, and phase aberration correction for intact skull imaging. I will then focus on reviewing brain imaging applications that use SR-US to characterize cerebral mirovasculature in the aging, stroke, and Alzheimer's Disease brain. Finally, I will introduce the new functional super-resolution imaging technique that combines SR-US with functional ultrasound (fUS) to realize whole-brain neural activity recording at a micron-scale spatial resolution.

8:45

1aBAa3. Super-resolution imaging with modulation of point spread function. Jian-Yu Lu (Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606, jian-yu.lu@ieee.org)

The spatial resolution of an imaging system using waves is limited by the spatial bandwidth of the point spread function (PSF) of the system, which is related to the wavelength. However, when the PSF is modulated either in amplitude or phase or in both, the resulting spatial bandwidth of the PSF is increased. In this study, the PSF-modulation method is used to obtain super-resolution imaging of objects and to distinguish wave sources that are closely located in space and are not normally separable due to diffraction limit. In imaging using waves, such as ultrasound, acoustics, optics, electromagnetics, radar, and sonar, the PSF can be modulated in their respective fields. For example, in ultrasound, shear wave in biological soft tissues has a low wave speed and thus has a small wavelength. A ring-shaped shear wave can be generated locally (remotely) deep in the tissue by the radiation force of a focused Bessel beam, X wave, or other limited-diffraction beam at their focuses [Lu, 2021 IEEE IUS, 2021 ASA POMA] to produce a sharp peak at the center of the ring (due to shear wave focusing with a small wavelength). This sharp peak of the shear wave modulates the center of a conventional focused beam transmitted after the shear wave ring is produced. The modulated focused beam is then used to scan through an object to obtain a super-resolution image after removing the contribution of the original beam. In this talk, the theory, computer simulation, and experiment results of the super-resolution method will be presented.

9:05

1aBAa4. Dynamic 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) can map the vasculature at large depth with unprecedented resolution by localizing millions of injected microbubbles in hundreds of thousands of images acquired over a few minutes. The current state of the art in ULM is to use low concentrations to achieve the best possible spatial resolution without providing temporal information, which limits the development of functional biomarkers such as pulsality or the imaging of moving organs like the heart. In this work, we will present dynamic ultrasound localization microscopy (DULM), which enables the generation of dynamic images of the vasculature of periodic phenomena by using a combination of enhanced image formation and processing techniques to drastically increase the number of microbubbles that can be detected in each image. Specifically, we will describe how the detection of microbubbles directly in space time along with novel aberration correction algorithms and a motion-invariant Lagrangian beamforming approach can be used to increase the concentration of microbubbles 5-fold with a limited degradation in resolution. Examples of application for the mapping of pulsatility in the brain and the dynamics of the intramyocardial blood flow in 2D + t and 3D + t will be shown.

9:25

1aBAa5. Nonlinear ultrasound imaging of the microcirculation. Matthew Bruce (Appl. Phys. Lab., Univ. of Washington, 8817 Interlake Ave. N, Seattle, WA 98103, mbruce@uw.edu), Jonah Harmon, Anton Arkadevitch, and Zin Khaing (Dept. of Neurosurgery, Univ. of Washington, Seattle, WA)

Blood flow at and near the tissue level is a physiological parameter of significant experimental and clinical importance, as it reflects the adaptive response of organs to their normal biological environment, to disease, trauma, and the malignant progression of cancer. The use of microbubbles gives ultrasound imaging access to microvascular hemodynamics, which are beyond the ability of Doppler ultrasound methods. However, expanded clinical use of microbubbles has been limited due in part to a lack of quantification and providing only relative differences in microvascular flow. Past and more recent approaches to the quantification of microvascular blood flow will be presented. Past approaches included the use of focused nonlinear pulsing sequences to isolate microbubble from tissue signals followed by analysis of bolus kinetics or flash replenishment dynamics. Although successfull to some extent, their clinical limitations prohibited extended use and the overall application of contrast enhanced ultrasound (CEUS). Newer approaches include higher frame-rate plane wave acquisitions combined with nonlinear pulsing schemes, which enable the tracking of microbubble flow through microvascular networks. Different approaches of these elevated frame-rate acquisitions will be presented, including use of ultrasound localized microscopy.

9:45

1aBAa6. Super-resolution cellular ultrasound imaging via localization of nanodroplets. John Kim (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, jkim3286@gatech.edu), Phoebe J. Welch, and Chengzhi Shi (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ultrasound localization microscopy (ULM) had made it possible to differentiate microbubble contrast agents that are separated only by a few micrometers and, thus, opening the path for super-resolution imaging. However, due to the innate limits of acoustic waves and microbubble imaging, similar levels of resolution to optical imaging (i.e., submicron resolution) are still very challenging. In this paper, we utilize ULM to achieve cellular imaging by using phase-change perfluorocarbon nanodroplets (PFCnDs) as a contrast agent that were fabricated with spontaneous nucleation method. To identify the point spread function (PSF) of a single nanodroplet, gaussian fit function was applied for reconstruction. Nanodroplets were injected into cells through patch clamping and later captured with two transducers: a single element transducer for focused ultrasound and linear array transducer for imaging. Nonlinear imaging (NLO) was used to maximize the sound to noise ratio (SNR), which enables optimal amplitude. With the PSF of a phase-transitioned nanodroplet known, stochastic activation of multiple nanodroplets within a cell was accumulated to image a full cell morphology. These findings can lead researchers to develop effective ultrasound imaging paradigms that can visualize intracellular level of organs in deep tissue *in vivo*.

10:00–10:15 Break

10:15

1aBAa7. Direct spatiotemporal localization of microbubble trajectories for highly resolved hemodynamics in ultrasound localization microscopy. Alexis Leconte (Polytechnique Montréal, 782 Ave. de l'Épée, Montréal, QC H2V3T9, Canada, alexis.leconte@polymtl.ca), Jonathan Porée, Brice Rauby, Paul Xing, Chloé Bourquin, Nin Ghigo (Polytechnique Montréal, Montréal, QC, Canada), Gerardo Ramos, Abbas F Sadikot (McGill Univ., Montréal, QC, Canada), and Jean Provost (Eng. Phys., Polytechnique Montreal, Montreal, QC, Canada)

Using high concentrations of microbubbles in ultrasound localization microscopy can reduce acquisition time but is also associated with reduced localization precision and accuracy of blood flow measurements. To address these limitations, we introduce Ultrasound Spatio-Temporal Localization (USTL), a novel approach that localizes microbubble trajectories using a spatiotemporal physiological constraint, in contrast to standard approaches that detect, pair, and track microbubbles over time without a priori. We tested USTL *in vivo* in the brains of rats and mice using a 15 MHz linear array probe and a Vantage system at different concentrations of microbubbles. Overall, USTL increased the number of detected microbubbles while reducing processing time and susceptibility to signal attenuation at depth. USTL provided coherent velocity profiles in vessels regardless of microbubble concentration, making it a valuable tool for studying brain hemodynamics in various conditions. Overall, USTL offers a new approach for the non-invasive measurement of dynamic brain function with high spatial and temporal resolution.

10:30

1aBAa8. Deep learning-based super-resolution ultrasound imaging for a wide range of microbubble density. Qiyang Chen (Univ. of Pittsburgh, 3550 Terrace St., 624, Pittsburgh, PA 15261, qic41@pitt.edu), Kayhan Batmanghelich, Roderick Tan, and Kang Kim (Univ. of Pittsburgh, Pittsburgh, PA)

Super-resolution ultrasound (SRU) imaging that can identify the vasculature with a high spatial resolution has a great potential for assessing and monitoring the progression of diseases associated with microvascular changes. One limitation of the conventional SRU is that the image quality is highly sensitive to the microbubbles (MBs) concentration at the imaging site, an aspect that is difficult to control *in vivo* due to the varying local perfusion and vessel sizes in the different organs. Deep learning (DL) techniques have been explored in the field of medical ultrasound for various applications, including image reconstruction, segmentation, and disease classifications. In this study, we developed a DL-based SRU imaging that works in end-to-end fashion without parameter tuning and is robust over a wide range of MB density while preserving the spatial resolution. The DL network containing U-net and interpolation layers was trained by the simulated MB images to replace the conventional MB center localization algorithms. The developed algorithm was evaluated *in vivo* on the mouse kidney for different MB injection doses and shown consistent performance, which support the robustness of this approach to varying MB density. The end-to-end operation that requires minimum parameter tuning would also potentially benefit the clinical applications.

10:45

1aBAa9. Ultrasonic subwavelength imaging with blind structured illumination. Jinuan Lin (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 4715 Sheboygan Ave. Apt 219, Madison, WI 53705, jlin328@wisc.edu) and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

The resolution of ultrasonic imaging systems suffers from the diffraction limit. To solve this problem, we develop a blind structured illumination microscopy (blind-SIM) system that achieves far-field subwavelength resolution in underwater ultrasonic imaging systems. By illuminating the object with multiple structured illumination (SI) patterns generated by scattering media with subwavelength features, the spatial frequency mixing between the object and the SIs converts evanescent waves to propagate waves that can reach sensors in the far field. In this work, the acoustic SIs are generated by randomly distributed micro particles in water. The image of the object is then reconstructed from multiple measurements (each produced by a SI pattern) utilizing computational reconstruction algorithms where the precise knowledge of the SIs is not required. The developed ultrasonic blind-SIM system has the potential in improving the ultrasonic imaging resolution in medical diagnoses, underwater acoustic imaging, and communications. Our system largely reduces the imaging hardware complexity and improves the imaging speed compared with existing ultrasonic subwavelength technologies that mostly rely on the precise localization of time-varying contrast agents.

11:00

1aBAa10. Resolution and contrast improved ultrafast power Doppler microvessel imaging with null subtraction imaging. Michael L. Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, oelze@illinois.edu) and Zhengchang Kou (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

In this work, we demonstrate a novel non-linear beamforming technique, called null-subtraction imaging (NSI), to improve spatial resolution and image contrast of ultrafast, power Doppler microvessel imaging without significantly increasing computational cost compared to traditional delay and sum (DAS). NSI is a non-linear beamforming technique that operates on receive data by subtracting envelopes of beamforming results from three apodization windows. An array transducer and Verasonics Vantage 256 system were used to capture 1600 frames of raw ultrasonic radio frequency (RF) channel data by scanning a rat brain using nine plane waves spanning angles between -4° and 4° at a pulse repetition frequency of 1000 Hz. A singular value decomposition filter was applied on the raw RF channel data to filter out the tissue signal. Coherent compounding was performed individually on the beamformed result for each apodization window. Next, the subtraction of the envelopes for each apodization was conducted to form the NSI image. A total of 1,600 frames of NSI images were accumulated to form the final image. The NSI based ultrafast power Doppler microvessel imaging provided a ten-fold improvement in spatial resolution and a three-fold improvement in image contrast compared with the traditional DAS-based ultrafast power Doppler microvessel imaging.

11:20

1aBAa11. Acousto-optical camera for super-resolution imaging. Bogdan-Ioan Popa (Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu)

I will present a hybrid acousto-optical imaging system that probes the environment with acoustic waves, converts the back-scattered sound into a coherent optical field on an aperture using an acousto-optical metasurface, and focuses the latter into an image with existing optical lenses. Preliminary theoretical analysis and experiments show that this approach can produce images of significantly higher resolution than possible with conventional methods, thus enabling high quality images produced with low frequency probing sound. Our imaging approach takes advantage of the best properties of optic and acoustic waves, namely, (1) the high image quality produced by existing optical systems and (2) the ability of low frequency acoustic waves to penetrate most materials. An additional advantage of our method is that it puts essentially no constraints on the sound source. For example, the source could have low directivity and may consist of a single transducer or it could even miss when imaging noisy environments. In the latter scenarios, our imaging method becomes purely passive.

Session 1aBAb

Biomedical Acoustics and Physical Acoustics: Ultrasound Induced Cell Responses

Yun Jing, Cochair
Acoustics, Penn State University, 201 Applied Science Building, State College, PA 16802

Qifa Zhou, Cochair
Bioengineering, Univ. of Southern California, Los Angeles, CA 90089

Costas Arvanitis, Cochair
School of Mechanical Engineering, Dept. of Biomedical Engineering,
Georgia Institute of Technology, 311 Ferst Drive Northwest, Atlanta, GA 30332

Invited Papers

8:20

1aBAb1. Ultrasound modulation of neurons by sonogenetics. Hong Chen (Washington Univesity in St. Louis, 6338 Washington Ave., University City, MO 63130, chenhongxjtu@gmail.com)

Focused ultrasound (FUS) can noninvasively deliver ultrasound energy to target any area in the whole brain with combined depth penetration and spatial focusing that cannot be achieved with other external stimulations. Sonogenetics, integrating focused ultrasound with genetic engineering, can achieve noninvasive control of neurons genetically modified with ultrasound-sensitive ion channels. Crucial to the development of sonogenetics has been the identification of ultrasound-sensitive probes to control neurons. FUS propagation through the tissue can generate mechanical and thermal effects within its focal region. This talk will present recent progress in the development of mechanosensitive and thermosensitive ion channels as sonogenetic probes. Recent success in using sonogenetics for noninvasive and cell-type-specific behavior control of freely moving mice will also be presented. This noninvasive and cell-type-specific neuromodulation approach has the promise to advance the study of the intact nervous system and uncover new ways to treat neurological disorders.

8:40

1aBAb2. Ultrasound mediated control of neurons and immune cells. Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., 311 Ferst Dr. Northwest, Atlanta, GA 30332, costas.arvanitis@gatech.edu), Pradosh Pritam Dash, and Chulyong Kim (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ultrasound holds great promise for realizing noncontact stimulation and control of cell function. Here, we will present our ongoing efforts in controlling cell responses using two distinct approaches. The first is based on engineered cytotoxic immune cells with thermal switches, where closed-loop controlled MRgFUS-hyperthermia (20-mins at 42 °C) is employed to drive the local production of key transgenes and potentiate anti-tumor responses in brain tumors. The responses are assessed using bioluminescence, cell viability, and cell cytotoxicity assays. The second is based on US pulse sequences designed to promote mechanical effects on cultured cells with the goal to decode US neuromodulation. For our investigations, we employed a high throughput ultrasonic platform designed to monitor and locally control sound and vibration in combination DRG sensory neurons. The later are the most sensitive neuronal type to mechanical stretch and present an excellent experimental system to examine and identify mechanosensitive ion channels sensing sonication-induced membrane stretch. Cell responses are quantified and analyzed for different pulse durations and amplitude, cell culture conditions, and degassing protocols. Together, our findings show that ultrasound thermal and mechanical effects can be employed to control the function of different cell types, albeit cells with thermal switches provide more robust responses.

9:00

1aBAb3. Drug-mediated acoustic reporter genes for mammalian cell ultrasound imaging. Phoebe J. Welch (GWW School of Mech. Eng., Georgia Inst. of Technol., 315 Ferst Dr. NW, IBB 2A, Atlanta, GA 30332, pwelch8@gatech.edu), Alessandro R. Howells (Biomedical Eng., Penn State Univ., State College, PA), John Kim, Craig R. Forest (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Xiaojun L. Lian (Biomedical Eng., Penn State Univ., State College, PA), and Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The newly established field of genetically encoded ultrasound contrast agents in the form of gas vesicles could expand the uses of medical ultrasound imaging for cell specific deep tissue imaging. However, current gene constructs encoding for these gas vesicles require significant cell processing to ensure sufficient gas vesicles are formed within the cell to produce ultrasound contrast. Here, we describe a drug-inducible and drug-selectable acoustic reporter gene construct that can enable gas vesicle expression in mammalian cell lines, which we demonstrate in wild type HEK293T cells. Plasmid integration was validated using fluorescence microscopy, and flow cytometry was used to establish single cell clones of the cells. Gas vesicle expression was optically and ultrasonically verified, with an 80% improved signal to noise ratio in cells expressing gas vesicles compared to negative controls. This technology presents a new reporter gene paradigm by which ultrasound can be harnessed to visualize specific cell types *in vitro* and *in vivo* for various applications, including cellular reporting and cell therapies.

9:15

1aBAb4. Cellular response to photo-mediated ultrasound therapy. Madhumithra Subramanian Karthikesh (Univ. of Kansas, 2720 Brittany Pl, 101, Lawrence, KS 66049, m787s254@ku.edu) and Xinmai Yang (Univ. of Kansas, Lawrence, KS)

Photo-mediated ultrasound therapy (PUT) is a novel technique that combines relatively low-pressure focused ultrasound and low-fluence laser to enhance cavitation inside blood vessels. Due to the increased shear and circumferential stresses, the enhanced cavitation activity can achieve anti-vascular effect. The role of endothelial cells in the antivascular effect of PUT was explored by quantifying the release of vasodilators-nitric oxide and prostacyclin from endothelial cells after PUT, laser-only, and ultrasound-only therapies in an *in vitro* vessel mode. PUT was able to suppress both the vasodilators by more than 5% with concurrently applied 0.5MPa ultrasound pressure at 500 kHz and 15 mJ/cm^2 laser fluence at 532 nm. Both the laser-only and ultrasound-only therapies at the similar levels resulted only in elevation of these vasodilators with respect to control. The suppression of vasodilators by PUT was justified with cavitation signals observed using passive cavitation detection and apoptosis observed with Annexin V -fluorescein isothiocyante (FITC), at respective levels. The endothelial cellular response to PUT further motivated us to evaluate the effect of PUT on melanoma cells and evaluate cytotoxic effect through MTT assay. PUT was able to destroy cancer cells through the cavitation effect when compared to laser-only and ultrasound-only therapies at the similar energy levels.

9:30

1aBAb5. Assessing the effects of ultrasound parameters on P-selectin expression for active targeting of nanotherapeutics. Mark Burgess (Dept. of Medical Phys., Memorial Sloan Kettering Cancer Ctr., 321 E 61st St., New York, NY 10065, burgesm1@mskcc.org), Daniel Tylawsky (Molecular Pharmacology Program, Memorial Sloan Kettering Cancer Ctr., New York, NY), Quincey LaPlant (Dept. of Radiation Oncology, Memorial Sloan Kettering Cancer Ctr., New York, NY), and Daniel Heller (Molecular Pharmacology Program, Memorial Sloan Kettering Cancer Ctr., New York, NY)

P-selectin is a nanotherapeutic target for enhanced passage of nanoscale drug delivery systems across the vascular barrier and into tumors via transcytosis. While many tumors express P-selection, radiation can be used to induce expression in tumors devoid of P-selectin. This study aims to test whether ultrasound can elicit a similar response through mechanical or thermal bioeffects. Preliminary experiments were performed *in vitro* using brain tissue derived mouse endothelial cells. A Vevo SoniGene system was used to sonicate cells with low-intensity (2 W/cm^2) pulsed ultrasound in a 12-well cell culture plate. Upregulation of P-selectin was seen using a 60 second treatment duration at 50% and 100% duty cycles via flow cytometry analysis. The mean fluorescence intensity, normalized to nontreated cells, was 32 +/- 17.4 and 103 +/- 37.2 for 50% and 100% duty cycles, respectively. Viability remained high with 72.6 +/- 14.5 and 77.5 +/- 8.3% of cells viable for both 50% and 100% duty cycle groups, respectively. These results highlight the potential of ultrasound for active targeting using non-ionizing energy and for tumors where radiation is not feasible. Future studies will identify associated bioeffects and compare ultrasound with radiation for nanotherapeutic delivery in small animal tumor models.

9:45

1aBAb6. Ultrasound-spectroscopic-imaging study on the effects of local mechanical stimulation on living human induced pluripotent stem cells. Natsumi Fujiwara (Graduate School of Eng., Osaka Univ., Yamadaoka2-1-M1-523, Suita-city, Osaka 565-0871, Japan, fujiwara@qm.prec.eng.osaka-u.ac.jp), Takaki Matsumoto, Akira Nagakubo, Masahiro Kino-oka, and Hirotsugu Ogi (Graduate School of Eng., Osaka Univ., Suita, Japan)

Living cells sense various mechanical stimuli and respond to them by adjusting their tissue morphogenesis, self-renewal, and differentiation. Because stem cells seem to be more sensitive to the stimuli, many studies have been conducted to control their behavior. However, many of these mechanisms remain unexplored, because stimulus extends throughout the cells or causes cell-damaging. In this study, we have developed a focused ultrasound spectroscopic technique for noninvasively applying local mechanical stimulation in a single cell and combined it with an optical microscopy to systematically investigate its effects on the function and morphology of the cell. Our original system allows ultrasonically monitoring living cells for longer than 24 h. Appling this technique for human iPS cells, we find out an ultrasound absorption band at ~160 MHz, which we attribute to the resonance of the nucleus. This indicates that we can stimulate cell nucleus selectively and effectively. Furthermore, we have established a method to evaluate the height distribution of cell colonies and revealed that human iPS cell colonies show very characteristic changes in height distribution during their growth.

10:00

1aBAb7. Tuning microbubble-mediated endothelial drug delivery outcome with monodisperse microbubbles of different sizes. Yuchen Wang, Hongchen Li (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Bram Meijlink (Biomedical Eng., Erasmus MC, Utrecht, Utrecht, Netherlands), Robert Beurskens, Antonius F. van der Steen (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Benjamin Johnson (Molecular and Nanoscale Phys. Group, School of Phys. and Astronomy, Univ. of Leeds, Leeds, United Kingdom), and Klazina Kooiman (Biomedical Eng., Erasmus MC, Wytemaweg 80, Rm. Ee2302, Rotterdam 3015 CN, Netherlands, k.kooiman@erasmusmc.nl)

Microbubble-mediated drug delivery using polydisperse microbubbles has shown differences in drug delivery outcome due to the microbubble's differences in size and, thus, acoustic response. The aim of this study was to investigate whether monodisperse microbubbles can achieve controllable sonoporation and tunnel formation. Using the Horizon microfluidics platform, monodisperse phospholipid-coated microbubbles were produced with radii of 1.25–3.5 μm. Single microbubble-endothelial cell interactions (n = 82) upon insonification at 2 MHz (200 kPa PNP, 10 cycles) were investigated *in vitro* using confocal microscopy and ultra-high-speed imaging (10 Mfps). No cellular response was observed for the 3.5 μm microbubbles having excursion amplitudes (R_{max}-R_0) ranging from 0.3 to 0.6 μm. PI uptake and resealing pores were observed more often for the 1.5 μm microbubbles (50%) with excursion amplitudes ranging from 0.5 to 0.9 μm than for the other microbubble sizes. PI uptake and tunnel formation was most

observed for the 1.25 μm microbubbles (46.6%; 0.4–0.6 μm excursion amplitudes) and least for the 3 μm microbubbles (4.2%; 1.0 μm excursion amplitude). No other cellular responses were observed for the 3 μm microbubbles. The excursion ratio (R_{max}-R_{min}/R_0) better separated the tunnel formation events (>0.6) from other cellular responses (0.1–0.6). This study shows the importance of monodisperse microbubbles for tuning drug delivery.

10:15–10:30 Break

10:30

1aBAb8. An *in silico* model of clot degradation under the action of histotripsy and thrombolytic drugs. Kenneth B. Bader (Univ. of Chicago, 5835 South Cottage Grove Ave. Dept. of Radiology, MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu)

For venous thrombosis patients, catheter-directed thrombolytic drugs remain the standard-of-care to restore vascular flow. Improved outcomes are observed when 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. A Monte Carlo-like simulation was used to gauge the extent of histotripsy ablation. The local tissue properties that dictate bubble nucleation were adjusted based on annotated histological sections of venous thrombi. The distribution of thrombolytic drug within the clot was computed using a finite-difference time domain solution of the perfusion-diffusion equation. Fibrin degradation was computed using the known reaction rate of thrombolytic drug. Predictions of ablation were dependent on the clot subgroup, with a reduction in the extend of ablation as the concentration of fibrin increased. Thrombolytic drug was more uniformly distributed throughout the clot when the effects of histotripsy ablation were consistent, consistent with recent *in vitro* measurements. Overall, the findings here indicate histotripsy primarily enhances the activity of thrombolytic drugs is via debulking red blood cells within a clot.

10:45

1aBAb9. Ultrasounds trigger reversible permeability of the cerebrovascular endothelium in a novel 3D-bioprinted model of the human blood-brain barrier. Mona Mirheydari (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, mirheyma@mail.uc.edu), Sirjana Pun (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Daniel Pomeranz Krummel (Dept. of Neurology and Rehabilitation, Univ. of Cincinnati, Cincinnati, OH), and Riccardo Barrile (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

The blood-brain barrier (BBB) restricts the penetration of therapeutic agents to the brain. Focused ultrasound (FUS) insonation of microbubbles has enhanced drug delivery via a temporary opening of the tight junctions of the BBB. Rodents are often used for testing novel therapeutic strategies. However, biological discrepancies existing between rodents and humans impedes the translation of some results. We investigated the effect of FUS transient disruption of a human microphysiological model of the BBB consisting of human brain endothelial cells cultured within a 3D-bio printed scaffold designed to reconstitute the architecture of a perfused blood vessel. Cavitation activity was detected in the model when perfused with Lumason microbubbles and exposed to 500 kHz ultrasound at 0.4 MPa for 10 ms pulse duration and 2 Hz pulse repetition frequency. An increase in extravasation of a dye beyond the endothelial layer was detected within 2–3 h of ultrasound insonation. Our results support that this system could be used for targeted BBB opening with a human cell culture model. Moreover, this system holds a promise to further the translational study on US-mediated drug delivery and the development of personalized therapies for patients with brain cancer and other neurovascular disorders.

11:00

1aBAb10. Investigation of the ultrasound-mediated toxicity mechanisms of various sonosensitive drugs. Kritika Singh (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd., Headington, Oxford OX3 7LD, United Kingdom, kritika.singh@magd.ox.ac.uk), Alexandra Vasilyeva, Luna Hu, Jia-Ling Ruan, Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

There is a current pressing need to develop novel platform cancer therapeutics that are efficient, reduce side effects, and are minimally invasive. One of these platforms is photodynamic therapy where light is used as an external stimulus to activate drugs at a target location; however, clinical applications are limited due to poor light penetration. Previously, it has been shown that ultrasound with and without cavitation nuclei can be used to activate photodynamic drugs (sonodynamic therapy – SDT); however, the mechanism of this activation remains unclear. Recently, sonoluminescence has been detected in real time using a photomultiplier tube setup, up to millisecond temporal resolution; however, the spectra of light produced as well as the associated intensity has yet to be characterized. A proposed mechanism for SDT is that this light produced can activate photodynamic drugs at the target location resulting in reactive oxygen species (ROS) production (which leads to cell death). However, through temporal uncoupling of ultrasound application and compound administration *in vitro* and *in vivo*, this work shows that intracellular uptake of compound via cell permeabilization (sonoporation) plays an important role in SDT-induced cell death and potentially explains the high levels of cell death observed for comparatively low concentrations of ROS.

11:15

1aBAb11. Tracking macrophages with ultrasound. Ashley Alva (Elec. and Comput. Eng., Georgia Inst. of Technol., 901 atlantic Dr., Atlanta, GA 30318, aalva3@gatech.edu), Chulyong Kim, Hohyun Lee (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)

Macrophages play a key role among the immune cells in host defense. They are also known to be involved in various complex processes like angiogenesis and can infiltrate disease environments like in the case of cancer and atherosclerosis. Hence, imaging macrophages and their trafficking patterns could provide important prognostic and diagnostic markers of various human diseases. Previous attempts at imaging macrophages have been limited in depth of penetration, spatial resolution, and sensitivity. We hypothesize that labelling macrophages with microbubble (MB) contrast agents could render them visible to ultrasound and potentially overcome these limitations to enable imaging of macrophage trafficking at high resolution and sensitivity deep into tissues. Here, we show that macrophages can be labeled with MB to make them echogenic. Subsequently, we demonstrate that there are ultrasound exposure settings where these macrophages can retain high echogenicity and viability without affecting their functionality. We also confirmed that these MB-labeled macrophages can be imaged *in vivo* (7-fold increase in CNR as compared to unlabeled macrophages) following intratumoral injection into subcutaneous tumors. In summary, our research findings demonstrate a highly sensitive approach to study macrophage interactions with disease environments that may lead to novel biomarkers for disease diagnosis.

11:30

1aBAb12. Mechanistic study of blood brain barrier opening by microbubbles and focused ultrasound. Lu Na Hu (Biomedical Eng., BUBBL Group, Univ. of Oxford, Brasenose College, Radcliffe Square, Oxford, Oxfordshire OX1 4AJ, United Kingdom, luna.hu@eng.ox.ac.uk), Olive Jung (National Ctr. for Advancing Translational Sci., Bethesda, MD), Jia Ling Ruan, Kristoffer Petersson (CRUK-MRC Oxford Inst. for Radiation Oncology, Oxford, United Kingdom), Marc Ferrer (National Ctr. for Advancing Translational Sci., Bethesda, MD), and Eleanor P. Stride (Biomedical Eng., BUBBL Group, Univ. of Oxford, Oxford, United Kingdom)

Drug delivery to the brain is restricted by the blood brain barrier (BBB), limiting up to 98% small molecule drugs and almost 100% biologics. This severely limits the treatment options for diseases of the CNS. The use of microbubbles (MBs) in combination with focused ultrasound (FUS) has successfully delivered multiple classes of therapeutics through the BBB, although the molecular mechanism of the BBB opening, and associated adverse effects, are yet to be fully understood or characterised. The aim of this study was to perform simultaneous imaging of microbubble dynamics and cellular response under different ultrasound exposure conditions in a two-dimensional *in vitro* endothelial model with and without MBs present. The results indicate that changes in cell-cell junctions dominated over sonoporation and/or other mechanisms of cellular uptake of a fluorescent drug surrogate. Investigation was also made of the signalling pathways associated with paracellular permeability and sterile inflammation.

11:45

1aBAb13. Numerical simulation and experimental studies on acoustofluidic separation of microparticles using 3D printed chips. Alara Karaman (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey), Barbaros Çetin (Mech. Eng., Bilkent Univ., Ankara, Turkey), and Mehmet Bülent Özer (Mech. Eng., Middle East Tech. Univ., Orta Doğu Teknik Üniversitesi, Makina Mühendisliği Bölümü, Ankara 06800, Turkey, ozerb@metu.edu.tr)

Principles of acoustofluidics can be used to manipulate and separate microparticles and cells in microchannels. In this study using numerical simulations, we will be investigating the limits of fundamental assumptions used in acoustofluidic such as a limited acoustic scattered field or fluid flow interaction between the particles/cells. A device-level numerical simulation tool will be used which accounts for the effect of the piezoelectric materials, interactions between the microparticles/cells fluid flow-field and the acoustic-field in the microchannel. Also, in this study, different numerical acoustic and mechanical models of biological cells will be investigated and the conditions (frequency, channel dimensions, etc.), which favor the acoustofluidic separation of cells with similar mechanical properties will be studied numerically. Finally, the results of the numerical simulations will be utilized, and experimental studies will be demonstrated, which separate microparticles/cells in a 3D printed polymeric chip.

Session 1aCA

Computational Acoustics, Architectural Acoustics, Physical Acoustics, and Musical Acoustics: Real-Time Computing I (Hybrid Session)

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

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Chair's Introduction—10:00

Invited Papers

10:05

1aCA1. What would a Webchuck Chuck? Chris Chafe (Music, Stanford Univ., CCRMA/Music, Stanford University, Stanford, CA 94305, cc@ccrma.stanford.edu), Ge Wang, Mike Mulshine, and Jack Atherton (Music, Stanford Univ., Stanford, CA)

Webchuck is a new platform for real-time web-based music synthesis. Combining expressive music programming power with the ubiquity of web browsers is a game changer, one which we've experienced in recent teaching at Stanford's CCRMA. The paper will describe how this has fulfilled a long-sought promise for accessible music making and simplicity of experimentation. The Chuck music programming language now runs anywhere a browser can and a link is all it takes. Sample-synchronous live coding alongside full-on studio functionality is what a Webchuck can Chuck, thanks to advances in the web audio API.

10:25

1aCA2. Low-complexity equalizers and applications. Udo Zoelzer (Elec. Eng., Helmut Schmidt Univ. Hamburg, Holstenhofweg 85, Hamburg 22043, Germany, zoelzer@hsu-hh.de)

Equalizers for audio signal processing play an important role in microphones, mixing systems, mastering processors, monitor loudspeakers, and consumer devices such as smart tablets, smart phones, wireless headphones, and hearing devices. We will introduce low-complexity filter designs for equalizers based on first- and second order recursive filters. Low-complexity filter design means having the lowest number of parameters and design equations for first- and second-order difference equation realizing prescribed filter transfer functions such as low-pass, high-pass, band-pass, and all-pass filters. Based on these basic filters, we derive all-pass realizations of these standard filters and then apply them to design parametric low- and high-frequency shelving filters and peak filters. These last two versions of weighting filters are based on three parameters, namely, the cut-off or center frequency, the bandwidth or Q factor, and the gain in dB for a low-, mid-, or high frequency band and are, therefore, named parametric equalizers. These parametric equalizers' PEQs occur in a channel-strip of a mixing console and are an integral part of the mixing process. We will describe several further applications of PEQs for amp and loudspeaker modeling, headphone equalization and head-related transfer function modeling, and audio coding. All applications allow a low-complexity filter design, parameter update, and efficient recursive filter implementations with much lower complexity compared to non-recursive filters.

10:45

1aCA3. Real-time musical performance across and within extended reality environments. Rob Hamilton (Arts, Games and Simulation Arts and Sci., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hamilr4@rpi.edu)

The design of real-time digital musical instruments, both novel as well as the emulation of traditional musical performance practices using computer systems, has always required nearly imperceptible latencies between controlling gesture and expressed sound. Sound and music based experiences built within immersive visual rendered spaces often struggle to provide seamless articulate control of rendered dynamic output. Working across and within extended reality environments, the complexity of the challenge continues to evolve with current generations of hardware and software pushing core and graphics processors to their respective limits. The introduction of networked performance spaces has only added to the complexity, resulting in the need for additional latency mitigation strategies from both the technical and artistic domains. This paper will discuss recent creative and technical strategies and concerns in the design, development and use of real-time virtual musical instruments.

11:05–11:20 Break

11:20

1aCA4. Real-time considerations for a source-time dominant auralization scheme. Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov) and Randall Ali (Dept. of Elec. Eng., ESAT-STADIUS, KU Leuven, Leuven, Belgium)

A well-designed recording system can capture a moving source without risk of distortions, knowledge of the source or path, or transmission of information back to the source (i.e., a smartphone can reasonably record a plane flying overhead). This necessarily happens in real time. It would be good if signal processing schemes for auralization possessed these properties. Recent work on the NoTAP method of auralization proposed an asynchronous sample rate conversion scheme that keeps track of the (nonuniform) rate of incoming samples to formulate an effective incoming sampling frequency. This value allows the method to predict what frequency regions at the receiver are vulnerable to aliasing or imaging artifacts. Strategies of oversampling and filtering can be used to eliminate these problem regions while preserving as much of the original content as possible given the desired receiver sampling frequency. This approach creates a situation where the receiver processing can run independently of the source/path processing making it attractive for real-time implementation. This presentation discusses the challenges associated with producing a truly real-time scheme. A three-way tradeoff emerges between an interpolation mechanism that generates decorrelated noise, the computational burden, and the nearness to absolute real-time with which one wants the scheme to run.

11:40

1aCA5. Low-delay interactive rendering of virtual acoustic environments with extensions for distributed low-delay transmission of audio and bio-physical sensor data. Giso Grimm (Dept. of Medical Phys. and Acoust., Carl von Ossietzky Universitaet, Marie-Curie-Str. 2, Oldenburg 26129, Germany, g.grimm@uni-oldenburg.de), Angelika Kothe, and Volker Hohmann (Dept. of Medical Phys. and Acoust., Carl von Ossietzky Universitaet, Oldenburg, Germany)

In this study, we present a system that enables low-delay rendering of interactive virtual acoustics. The tool operates in the time domain based on a physical sound propagation model with basic room acoustic modelling and a block-wise update and interpolation of the environment geometry. During the pandemic, the tool was extended by low-delay network-transmission of audio and sensor data, e.g., from motion sensors or bio-physical sensors such as EEG. With this extension, distributed rendering of turn-taking conversations as well as ensemble music performances with individual head-tracked binaural rendering and interactive movement of directional sources is possible. Interactive communication requires a low time delay in sound transmission, which is particularly critical for musical communication, where the upper limit of tolerable delay is between 30 and 50 ms, depending on the genre. Our system can achieve latencies between 7 (dedicated local network) and 100 ms (intercontinental connection), with typical values of 25–40 ms. This is far below the delay achieved by typical video-conferencing tools and is sufficient for fluent speech communicationand music applications. In addition to a technical description of the system, we show here example measurement data of head motion behaviour in a distributed triadic conversation.

Session 1aEA

Engineering Acoustics: Acoustic Transducers, Devices, and Systems

Thomas E. Blanford, Chair
The Pennsylvania State University, State College, PA 16804

Contributed Papers

9:00

1aEA1. Design of sonar receivers for shallow water buried object imaging. Thomas E. Blanford (The Penn State Univ., State College, PA 16804, teb217@psu.edu), Jason Philtron (The Penn State Univ., University Park, PA), and Daniel C. Brown (The Penn State Univ., State College, PA)

Downward looking synthetic aperture sonar can detect and localize proud and buried objects, such as unexploded ordinance (UXO), in very shallow water environments. These sensors generate three-dimensional imagery from a two dimensional array that is a hybrid of a synthetic and real aperture sonar. Achieving high-resolution imaging with these sensors places stringent requirements on the sonar hardware. The acoustic signals generated by the sensor are wide bandwidth, low frequency, and high dynamic range. Receiver spacing and directivity also play an important role in image quality but create competing design constraints at low frequency. This work will present the results of a recent design study to develop receivers for a buried UXO imaging system. Achieving the stringent hardware requirements necessitated that the transducers, baffle, housing, and electronics were designed in concert. Finite element and one-dimensional models were used to evaluate several candidate designs based upon piezocomposite, spherical, and hemispherical transducers and identify tradeoffs with each. The analysis will focus on both simulated and experimental results of the receiver components. Results from the component level also informed system level models to quantify the point spread function and resolution improvement of the sensor.

9:15

1aEA2. Underwater low frequency subwoofer systems. Andrey K. Morozov (TWR, Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

Experiments with a gas-filled adiabatic bubble resonator of large diameter have shown that it is an efficient emitter of low-frequency seismic waves. The resonator has a single resonant frequency with a narrow bandwidth. The internal acoustic resonant structure changes the frequency response of the entire source and makes it possible to expand its bandwidth. An internal Helmholtz resonator converts the system into a double resonant system with a wide bandwidth. The bass-reflex resonator separates the pressure in front and behind the speaker cone, and the speaker works as an internal monopole driver. Any other known loudspeaker enclosure design can be used to create the desired multi-pole frequency response. A specific of the underwater enclosure is that it is loaded into the inner chamber of the bubble resonator. The bubble resonator works as an efficient impedance converter, matching the impedance of the internal resonant system to the water radiation impedance. Examples of various systems and the results of their tests in water are considered. Experiments have shown that a gas-filled underwater bubble covered with an elastic membrane and driven by an audio subwoofer can achieve source levels of 180–185 dB re 1 μPa@1 m over a wide frequency band.

9:30

1aEA3. Implementation of lumped element multimode modeling for balanced-armature receivers. Mohammad Mohammadi (Knowles Electronics, LLC, 1151 Maplewood Dr., Itasca, IL 60143, mohammad.mohammadi@knowles.com)

Lumped element (LE) modeling is one of the simplest and fastest tools for designing and predicting the performance of miniature speakers. Tweeters are being incorporated more frequently to extend the high-frequency range of earphones in the commercial earphone market with balanced armature (BA) receivers being commonly used as tweeters. To support this trend, it is necessary to develop LE models of BAs that are accurate at high frequencies. Developing accurate models has been challenging due to several mechanical and acoustic modes interacting at frequencies over 10 kHz. Based on the work of Sun and Hu ["Lumped element multimode modeling of balanced-armature receiver using modal analysis," J. Vib. Acoust. 2016 Dec 1;138(6)], a multimode LE modeling method using finite element (FE) modal analysis is implemented and compared with older LE models. The simulation results are compared with experimental data for several production models of BA. The multimode LE method shows significant improvements compared to the old single-mode LE models in predicting BA receiver acoustic response, especially at frequencies above 10 kHz.

9:45

1aEA4. Testing and modeling of nozzle and eartip acoustics for balanced armature tweeter implementation in earphones. Brenno R. Varanda (Knowles Electronics, LLC, 1151 Maplewood Dr., Itasca, IL 60143, brenno.varanda@knowles.com)

Balanced armature (BA) receivers can be used as tweeters in earphones to increase their treble and ultra sound bandwidth. Key factors that influence the 9–20 kHz response of earphones when implementing BA tweeters are shown through experimental validation and comparison to newly-developed SPICE lumped parameter models (LPM). While the BA location and acoustic passage inside the earphone are important factors, the study focuses on how changes to the design of the earphone's nozzle and the choice of eartips affect the 9–20 kHz frequency response. Improvements to existing SPICE LPM elements are provided, based on finite element analysis of the acoustic passageway that is formed between the compressed eartip and ear-simulator couplers. The model is compared to measurements from various nozzle and eartip configurations, validating its effectiveness to capture the earphone's high frequency response due to changes to the earphone's nozzle and eartip geometry.

10:00

1aEA5. Aluminum foams with complex pore topologies for acoustical applications. Pulitha G. Kankanamalage (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Aerosp. Eng., Wichita, KS 67260, pxgodakawelakankanamalage@shockers.wichita.edu), Jake Puppo, Denver Schaffarzick (ERG Aerosp. Corp., Oakland, CA), and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Metal foams are used in various industries due to the great variety of properties they possess such as high strength-to-weight ratio, high energy absorption, and the ability to endure extreme conditions. However, despite their desirable properties, traditional metal foams lack acoustic absorption properties because of their stochastic open porous structure—a function of the foaming process. Additive manufacturing (AM) can allow the fabrication of more complex foams; however, current metal AM methods provide significant processing and scalability challenges, especially in printing aluminum parts. Here, we present an alternative method for fabricating open-celled aluminum sound absorbers with controlled cellular architectures. The method relies on modeling the cellular templates using an implicit, field-based modeling method. The templates are then fabricated by combining polymer-based AM techniques and converted into aluminum Duocel® foams using ERG Aerospace Corporation's proprietary foaming technology. The acoustical properties of the fabricated foams are then measured using a normal incidence impedance tube method. Our results show that this method allows the fabrication of highly complex cellular architectures that may be optimized to obtain application-specific multifunctional performance.

10:15–10:30 Break

10:30

1aEA6. Recent developments regarding the 15 dB improvement in SNR in noise available from four talkers in noise with wireless signals provided by simple clip-on-collar Companion Mics Remote Microphones. Mead C. Killion (MCK Audio, 61 Martin Ln., Elk Grove Vlg, IL 60007, abonso@aol.com), Frank Dunn (Fdpdcs LLC, Austin, TX), Michael Cevette (Mayo Clinic Arizona, Scottsdale, AZ), and Andrew Haapapuro (MCK Audio, Arlington Heights, IL)

The Companion Mics® (CMICS^{TM}) by Etymotic Research, and more recently by the author' company MCK Audio, have frequently demonstrated a 10–20 dB improvement in SNR in noisy groups and restaurants for up to four talkers, each of whom wears a CMIC wireless microphone unit on the collar or held near the chin. Until recently, however, the listener would need to wear an earphone. (Experimentally, two ears listening to the same signal provide only a 2 dB advantage.) The disadvantage was that the hearing aid audiologist would have to say: "If you have trouble in noise, take off your hearing aids and put on these earphones." While waiting for the incorporation of Bluetooth circuits, two other successful alternates have been introduced. The first is an open-ear HearHook© sound tube which hooks over the ear and delivers sound near the earcanal, which is also near the microphone of a hearing aid. The second is the use of a neckloop connected to the CMIC, with the hearing aid switched to "tcoil" mode. An amplified tcoil HHearbud with built-in *hum filter* allows that use even when electrical hum interference is present.

10:45

1aEA7. Vibration and airborne acoustic data fusion for diesel engine health management. Ruth Willet (Penn State Univ., 201 Old Main, University Park, PA 16802, raw5930@psu.edu) and Karl Reichard (Penn State Univ., State College, PA)

Prognostic health management (PHM) on ground vehicles is an important process to maximize vehicle availability and operational effectiveness. In doing so, maintainers can proactively address vehicle health concerns. Diesel engines are common to many platforms, so improving diesel engine health monitoring has wide applicability. Specifically, non-intrusive acoustic sensing techniques using microphones and accelerometers have been chosen for this application. This study presents initial research to fuse engine structural vibration and airborne acoustic signals for PHM. The research aims to understand how data from an accelerometer on an engine and multiple microphones placed at key locations around the engine can be integrated together for a better understanding of the engine health. By understanding the advantages and limitations of each sensor relative to the acoustic signals produced by different sources or processes on or within the engine, best practices can be developed for diesel engine acoustic data fusion for PHM.

11:00

1aEA8. Implementation of 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), Andrew Dittberner (GN Hearing, Bloomington, MN), 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 transducers 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. While theoretical work has been established for this process, all calculations have relied on far field equations leaving the near field not well understood. This was resolved by modeling the system in COMSOL Multiphysics with increasing complexity. Experimental implementation revealed additional flaws in the design including undesired vibrations and insufficient power from the key transducers. Vibrations in the window's crossbar were identified via accelerometers and suppressed by the addition of an aluminum brace. Experimental results will be presented.

11:15

1aEA9. 3D-printed acoustic head simulators that talk and move. Xinran Yue (Univ. of Illinois, Urbana- Champaign, Urbana, IL), Arya Nallanthighall (Univ. of Illinois, Urbana- Champaign, Champaign, IL), Manan Mittal (Univ. of Illinois, Urbana- Champaign, 1308 W Main St., #119, Urbana, IL 61801, manansm2@illinois.edu), Austin Lu (Univ. of Illinois, Urbana- Champaign, Champaign, IL), Kanad Sarkar (Univ. of Illinois, Urbana- Champaign, Urbana, IL), Ryan M. Corey (Discovery Partners Inst., Chicago, IL), Paris Smaragdis, and Andrew C. Singer (Univ. of Illinois, Urbana- Champaign, Urbana, IL)

Acoustic head simulators are used to collect realistic binaural audio with accurate interaural cues. Head simulators are difficult to acquire and often provide only listening and recording capabilities. For cocktail parties and conversation scenarios, it is necessary to have many head simulators that can simultaneously produce speech and capture sound. An added dimension of realism can be provided with motion. We rely on 3D printing to fabricate a large number of head simulators, each of which is equipped with a loudspeaker for mimicking speech production. We use silent multi-axis turrets to provide neck-like motion. Our actuated head simulator collection is integrated into the Mechatronic Acoustic Research System.

11:30

1aEA10. Numerical prediction nonlinear heat-driven acoustics behaviours in standing-wave thermoacoustic engines using stress-blended Eddy simulation method. Lixian Guan (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Dept. of Mechancial Eng., Private Bag 4800, Christchurch 8140, New Zealand, dan.zhao@canterbury.ac.nz)

The present work investigated a standing-wave heat-driven thermoacoustic engine (SWTAE) system by capturing its nonlinear thermoacoustic features with three-dimensional (3D) unsteady Reynolds-Averaged Navier-Stokes (URANS) and hybrid URANS/LES (large Eddy simulation) models such as detached-Eddy simulations (DES) and stress-blended Eddy simulation (SBES) models. The comparison studies show that the prediction of the acoustic power of SWTAE using URANS is about 21.0% lower than that using LES, while the results from SBES and DES are relatively in good agreement with LES. Comparative studies of nonlinear hydrodynamics in the flow fields show that the results from SBES are closer to LES than DES, which can be attributed to the SBES model providing a faster transition to an explicit LES model outside the wall boundary layer. Furthermore, the heat transfer characteristics are compared by analyzing heat leaks and transversal heat flux, and it is found that the URANS model over-estimates the heat transfer characteristics, while the results of the other three models are relatively smaller than those obtained by URANS. In conclusion, the SBES model has great potential to be applied in simulating thermoacoustic nonlinear, heat transfer and flow behaviors of heat-driven SWTAEs.

11:45

1aEA11. Experimental investigation on microfluidic devices for particle separation using standing surface acoustic waves. Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, dan.zhao@canterbury.ac.nz) and Justin Tay Jun Yang (College of Eng., Nanyang Technolog. Univ., Singapore, Singapore)

In this work, we designed and experimentally tested a microfluidic device for particle separation by using standing surface acoustic waves (SSAW). Emphasis is being placed on an effective microfluidic device being capable of continuous separation of particles in a microchannel design on its size. For this, we developed a miniaturized (e.g., lab-on-a-chip) system for biological and chemical analyses. For demonstration, we experimentally demonstrated that separation of dissimilarly-sized polystyrene particles suspended in water medium was successfully achieved. Numerical simulations were conducted by using COMOSL Multiphysics 5.0 to investigate the effects of sheath layer thickness and particle size on separation efficiency of the microfluidic device. The present numerical and experimental findings were then applied to propose an improved design and fabrication process of Standing surface acoustic waves devices, which has great potential for biological applications.

MONDAY MORNING, 8 MAY 2023 INDIANA/IOWA, 9:00 A.M. TO 11:15 A.M.

Session 1aMU

Musical Acoustics: Musical Acoustics General Topics I

Andrew A. Piacsek, Chair
Central Washington University, Ellensburg, WA 98926-7422

Contributed Papers

9:00

1aMU1. Circular breathing—An introspective study across all types of wind-instrument generators. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Circular breathing is an extended technique for wind instruments to play a continuous sound. To do so, the performer pushes out air temporarily stored in the cheeks while inhaling. In the second part of the cycle, the player exhales, refilling the air in the cheeks while driving the wind instrument. The two main challenges of this technique are maintaining a constant embouchure and air supply. This talk spans from the author's Hyperspecialization technique on the soprano saxophone that involves all major types of wind-instrument generators, including brass, flute, double-reed, single-reed, and free-reed generators. The history, difficulty, and outcome for each generator type will be discussed. The free-reed generator is the easiest to perform with circular breathing because the free-reed resonance frequency only depends on fingering and cannot be controlled by embouchure. The embouchure for the flute generator is very different from the other circular breathing techniques, and transferred knowledge and experience can hardly be applied. The type of wind-instrument generator itself is not always a good indicator of the easiness of performing circular breathing. For brass embouchures, for example, the didjeridu adapter was much easier to perform with circular breathing than the cornetto mouthpiece because of the latter's tight embouchure requirements. For double reeds, the duduk reed was more challenging than the bassoon reed, because it requires much more air to excite the instrument.

9:15

1aMU2. Bifurcation behavior and underblowing on the bass crumhorn. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh EH9 3FD, United Kingdom, d.m.campbell@ed.ac.uk), Sylvain Maugeais (Laboratoire Manceau de Mathématiques, Le Mans Université, Le Mans, France), and Alan Woolley (School of Phys. and Astronomy, Univ. of Edinburgh, Edinburgh, United Kingdom)

The crumhorn is a double reed musical wind instrument with a narrow cylindrical bore and a characteristic hook-shaped curve near its open end. It is played by blowing air through a narrow slit into a windcap which surrounds the reed. Crumhorns were made in several sizes, from soprano to contrabass, and in the sixteenth and early seventeenth centuries were frequently employed in wind ensembles in Germany, Italy and the Netherlands. Since no lip control of the reed is possible, the instruments cannot be overblown, and the playing range of a standard crumhorn is limited to a major ninth. On the bass crumhorn, however, a sixteenth century source mentions a technique for extending the lower range of the instrument by up to a perfect fifth by reducing the blowing pressure below its normal level. In 2000, Gibiat and Castellengo investigated these underblown notes experimentally and associated them with period doubling cascades. In the present work, the bifurcation behavior of an artificially blown bass crumhorn has been studied experimentally, and the results are compared with bifurcation diagrams calculated using a continuation method.

9:30

1aMU3. Experimental tests of free reed instrument simulations. Quinn Foti (Phys. Dept., University of North Carolina at Asheville, Asheville, NC 28804, tfoti@unca.edu), Max Sabin (Indiana Univ., Bloomington, IN), and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

Recent simulations using finite element software have been used to study the fluid dynamics, pressure acoustics, and structural mechanics couplings involved in self-sustained reed oscillation and reed-pipe coupling in free reed instruments. This paper reports on two experimental studies related to these simulations. One of these involves reeds of the type used in harmonicas and accordions. The results verify that the assumed dependence of sounding frequency on blowing pressure, which has long been used as a benchmark for a successful simulation is correct for blowing pressures close to those normally used in playing the instruments. At higher pressures, however, a variety of anomalous results can occur. The second study involves finite element models of pipes from the free-reed Asian khaen. Pipe simulations produce results for the input impedance of the pipe that agree very well with experimental measurements. A pressure acoustics model of the khaen pipe coupled to a free reed produces results that appear realistic for sound radiation but does not realistically capture the mechanism of the self-sustained oscillation of the reed driven by airflow. Measurements in the current study show that this model does provide approximate quantitative agreement with the predictions of the simulation.

9:45

1aMU4. End behavior of the acoustic standing wave in a flue organ pipe. Makayle S. Kellison (Dept. of Phys., Rollins College - Box 2743, Winter Park, FL 32789, mkellison@rollins.edu), Thomas R. Moore, Whitney L. Coyle, and Ashley E. Cannaday (Dept. of Phys., Rollins College, Winter Park, FL)

The acoustic standing wave in a transparent flue organ pipe has been imaged using high-speed transmission electronic speckle pattern interferometry. Spectral filtering of the interferograms at the acoustic resonance frequency of the gas inside the pipe results in an image showing the sinusoidal pressure dependence inside the pipe as well as the expected extension of the standing wave beyond the pipe end. However, the image also reveals anomalous and unexplained behavior as the standing wave transitions from inside to outside the pipe.

10:00–10:15 Break

10:15

1aMU5. A study of impedance of brass instruments and mouthpieces—Comparison of models and measurements. Miranda Jackson (CIRMMT, McGill Univ., Schulich School of Music, Montreal, QC H3A 1E3, Canada, miranda.jackson@mail.mcgill.ca)

The impedance of a brass instrument has an important influence on the frequencies of the notes that can be played and on the timbre of the sound. The shape of the mouthpiece has various features, such as the cup volume and shape, opening diameter, and length, that determine the characteristics of the overall impedance of the instrument-mouthpiece combination. Brass instruments, and especially mouthpieces, are designed for specific purposes, and many brass players own several different horns or mouthpieces, and choose which to use depending on their particular musical requirements at the time. In order to investigate the relationship between the physical parameters of instruments and mouthpieces and the resulting impedance, brass instruments and mouthpieces have been modeled with transfer matrix techniques, and the results are compared with impedance measurements of the instruments alone, the mouthpieces alone, and combination of instruments and mouthpieces. Trumpets, flugelhorns, horns, trombones, and the corresponding mouthpieces have been used for this study. The mouthpiece-instrument combination has been investigated in terms of intonation, playability, and timbre. The question of whether (and why) some mouthpieces are more suited to certain instruments and certain playing styles is investigated as well as the effect of varying the physical parameters of mouthpieces and instruments.

10:30

1aMU6. Amapiano music and a psychoacoustic approach to analysis through the Ski-hill graph. Andrea Calilhanna (Faculty of Arts, Elder Conservatorium of Music, The Univ. of Adelaide, New South Wales, Sydney, New South Wales 2126, Australia, a.calilhanna@gmail.com)

Local artists in African countries' use of computer technology to produce work is inevitable. The effect of popular commercial music on traditional musicians increases the genuine possibility of further extinction of local music, particularly in regional areas. The importance of preserving this music is underacknowledged as, like language, music is an essential form of communication and part of a person's identity. New developments can cast shadows over everyday music during modernisation, and young people seek new sounds. For others, especially those who have witnessed the slow disappearance of cultural practices unique to regions, the loss and damage are akin to whitewashing and theft. In many parts of the world, scholars and locals are scrambling to preserve music and transmit accurate cultural heritage accounts for future generations. Amapiano music, like so much music of the Nigerian diaspora, owes its traditional heritage to the ancient roots of the Igbo people of Nigeria. Through the psychoacoustic approach of mathematical music theory of the Ski-hill graph (Cohn, 2020), it is now possible to produce accurate, inclusive, and ethical representations of Amapiano and traditional Igbo music.

10:45

1aMU7. Evaluating the efficacy of music programs in hearing aids. Emily Sandgren (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN) and Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Hearing aid processing of music is much less thoroughly researched than hearing aid processing of speech. This study evaluates the efficacy of programs specifically designed for music listening in state-of-the-art hearing aids. Normal-hearing participants rated the sound quality of music samples recorded from seven hearing aid brands in programs designed for speech or music. Preliminary results confirm the significant degradation of music quality caused by hearing aids. However, the music programs for three of the seven manufacturers significantly improved music quality compared to their speech programs. An analysis of signal processing and acoustic features relating to efficacy will be presented.

11:00

1aMU8. Frequency characterization of laser microphones. Carson F. Cunningham (Phys., Brigham Young Univ., Provo, UT), Alexandra M. Hopps (Phys., Brigham Young Univ., Eyring Sci. Ctr., N-181, Provo, UT 84606, hoppsa@byu.edu), Wyatt Losee, Richard Sandberg, David Allred, and Tracianne Neilsen (Phys., Brigham Young Univ., Provo, UT)

Historically, lasers have been used for a variety of purposes such as spectroscopy or optics. Lasers have even been used as a surveillance device; however, minimal research has been done in this area. In this experiment, a laser microphone was designed and tested with several different colored lasers to determine the frequency response of the received signal due to wavelength differences from the emitted light. The laser microphone included two telescoping lenses and an interferometer setup with a flimsy mirror as one of the arms. This mirror created a vibrating diffraction pattern when sound waves were played in close proximity. The interferometer was set up in an anechoic chamber, and a transimpedance photodiode was used to collect data of a moving diffraction pattern. With this setup, different sounds were recorded with varying levels of clarity, and the spectral response of the laser microphone was analyzed. With further research, a choice of laser color could be chosen to optimize the signal's response.

Session 1aNS

Noise, Computational Acoustics, and Physical Acoustics: Rocket Noise Part I

Kent L. Gee, Cochair
Department of Physics and Astronomy, Brigham Young University, N281 ESC, Provo, UT 84602

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Chair's Introduction—8:00

Invited Papers

8:05

1aNS1. A brief rocket noise introduction for the interested ASA meeting attendee. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu), Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Caroline P. Lubert (Mathematics and Statistics, James Madison Univ., Harrisonburg, VA)

It can be difficult for ASA meeting attendees to find an entry point into topical special sessions; they may be interested in learning about a field outside their own but are unfamiliar with terminology, physical processes, and the challenges that motivate different research studies. For the topic of rocket noise, this presentation is intended to help attendees from different ASA technical areas feel more prepared to understand the significance and details of the other talks in this session. But because there is far more to discuss than can be addressed in a single presentation, attendees are also referred to a recent review article on launch vehicle acoustics [C. P. Lubert *et al*., J. Acoust. Soc. Am. **151**, 752-791 (2022)].

8:25

1aNS2. Rocket noise models for the US Department of Defense. Alan T. Wall (Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), J. M. Downing, Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC), and Reese D. Rasband (Ball Aerosp., Inc., Provo, UT)

Accurate rocket noise prediction models are critical to support growing space defense programs. Potential deficiencies in the NASA SP-8072 model developed in the 1970s have been identified, but their impacts on accurate predictions are currently unknown. Cross-validation comparisons to full-scale rocket measurements are needed to identify and quantify errors and to improve accuracy. This paper will review the DoD programs and policies that motivate rocket noise modeling requirements, summarize the current models in use, discuss some of the potential deficiencies of the models, and propose a roadmap for establishing confidence in models for official DoD use.

8:45

1aNS3. Updated frequency-dependent directivity indices for large solid rocket motors. Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., N357 ESC, Brigham Young University, Provo, UT 84602, grant_hart@byu.edu), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Mylan R. Cook (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

For many years, empirical models for rocket noise radiation relied on the directivity indices published in NASA SP-8072 (K. Eldred, 1971). Because these indices had known issues, NASA led a campaign to update the indices using a Space Shuttle reusable solid rocket motor (RSRM). The RSRM measurements, involving a polar arc at a radius of about 80 nozzle exit diameters, resulted in updated directivity indices (Haynes and Kenny, AIAA 2009–3160). However, because the arc origin was placed at the nozzle exit plane, James *et al.* [Proc. Mtgs. Acoust., 18, 040008 (2012)] corrected the low-frequency indices using an estimated dominant source position for each frequency and assumed spherical spreading. This paper revisits that correction effort by using near-field vector intensity measurements from a similar, but smaller diameter and lower thrust, GEM-60 motor to determine the frequency-dependent origin to adjust the apparent angles and distances of the measurements. Additionally, an effort is made to account for RSRM plume impingement downstream that likely resulted in lower high-frequency levels than would have been measured if the plume had been entirely free. This analysis results in updated, frequency-dependent directivity indices for a large solid rocket motor. Their applicability to other rockets will be discussed.

9:05

1aNS4. The effect of nozzle contour on the vibroacoustic loads that form from high area ratio rocket nozzles during sea level launch. Charles Tinney (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, charles.tinney@arlut.utexas.edu)

An accurate assessment of the vibro-acoustic loads that form during startup of large area ratio rocket nozzles is important for sea-level launch vehicle design and certification. These loads are driven principally by various flow and shock patterns that form inside the nozzle, which are unique to the nozzle contour. This presentation will review a number of laboratory-scale measurements of different nozzle contours and configurations reported by Baars and Tinney, Exp. Fluids, (2013), Donald *et al*. AIAA Journal (2014), Canchero *et al*. AIAA Journal (2016), and Rojo *et al*. AIAA Journal (2016) as it relates to launch platforms of current interest. In particular are the various sources of noise pertaining to transonic resonance, broadband shock associated noise, and the end-effects-regime (EER). The latter of these is unique to the thrust-optimized parabolic contour nozzle as is used on the current Space Launch System vehicle. This EER event occurs when the annular flow structure is in a partial restricted-shock separated (RSS) flow state and is categorized by an onset of relatively low frequency energy driven by intermittent buffeting between RSS flow and partial free shock separated flow at the nozzle lip.

9:25

1aNS5. Rocket noise predictions and scaling methods based on outdoor acoustic measurements. Peter Kerrian (ATA Eng., Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128, peter.kerrian@ata-e.com), John McShane (ATA Eng., Inc., San Diego, CA), Dan Larson (Ursa Major Technologies, Berthoud, CO), Parthiv Shah, David Cloutier, and Michael Yang (ATA Eng., Inc., San Diego, CA)

This paper will present follow-on work to the outdoor acoustic measurements that were presented at the May 2022 ASA conference for a static hot fire test of a rocket engine. The focus is on scaling the measured sound pressure levels to predict the acoustic performance of a different sized engine. Source models were developed based on the near field acoustic measurements and compared to far field measurements.

9:45–10:05 Break

10:05

1aNS6. Vector acoustic intensity analysis of noise from the Delta IV Heavy launch vehicle during liftoff. Mylan R. Cook (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com), Kent L. Gee, Carson F. Cunningham, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The noise sources within a turbulent rocket plume are not well understood, let alone the radiation from multiple rocket nozzles. Even less is known about the noise sources during launch vehicle liftoff. This paper seeks to simultaneously address these noise source analysis challenges using data collected during the NROL-82 Delta IV Heavy (DIVH) launch from Vandenberg Space Force Base. The three-core DIVH's liftoff noise was measured by a ground-based four-microphone array at a distance of 330 m. Hart *et al*. [Proc. Mtgs. Acoust. 45, 040003 (2022)] previously used this array with a cross-correlation technique to identify the overall noise source radiation locus as it switched from the flame trench exit to about 55 m downstream of the nozzle exit plane. Here, vector intensity is used to localize the frequency-dependent noise source axial distribution using a variant of the phase and amplitude gradient estimator method for acoustic intensity [Thomas *et al*., J. Acoust. Soc. Am. **137**, 3366–3376 (2015)]. The results help clarify a complex noise source generation process that is both time and frequency-dependent and involves both free and impinging jet noise phenomena.

Contributed Papers

10:25

1aNS7. Acoustic overview of the Delta-IV Heavy NROL-91 launch. Matthew G. Yancey (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, mgy22@student.byu.edu), Levi T. Moats, Mylan R. Cook (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (California State Univ., Bakersfield, CA), Mark C. Anderson, Grant W. Hart, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

As commercial space launches continue to become more common, there is increasing interest in the subject of noise. The United Launch Alliance Delta IV Heavy is particularly interesting because of its three-core design with three widely separated engines. Measurements made by Brigham Young University at the NROL-91 launch on September 24, 2022, feature several microphones placed in an arc surrounding the launch facilities of the NROL-91 launch, ranging in distance from 670 m to 1.8 km. In addition, two long-range measurement locations (13.4 and 19.5 km) were included for propagation comparisons with an earlier Delta IV Heavy launch measurement (NROL-82). This presentation focuses on a description of the measurement made and several results, including spectra, directivity, and the sound power level calculated for the launch vehicle, as well as comparisons with the NROL-82 data.

10:40

1aNS8. Assessing azimuthal asymmetry in the noise radiation from a three-core launch vehicle during liftoff. Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmoats359@gmail.com), Matthew G. Yancey, Grant W. Hart, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Creating accurate rocket noise models is important for assessing impacts on humans, the environment, and payloads. The United Launch Alliance Delta IV Heavy launch vehicle is unique because of the separation of the three cores and their associated RS-68A nozzles. This makes it a good candidate for determining how the asymmetry of nozzle configuration affects noise radiation, which can affect noise models. The NROL-82 and NROL-91 missions both launched from Vandenberg Spaceforce Base using Delta IV Heavy Vehicles. For both of these launches, acoustic data were recorded between $\sim$0.9 and $\sim$5.2 km from the vehicle at different azimuths to determine the extent of azimuthal asymmetry in noise radiation. Maximum overall sound pressure level, spectra, and overall power level were determined for each launch. Methods for comparing the datasets and results will be discussed.

10:55

1aNS9. An overview of acoustical measurements made of the Atlas V JPSS-2 rocket launch. Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, lmathew3@byu.edu), Carson D. Gardner, Bradley W. McLaughlin, Mark C. Anderson (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Brooke M. Hinds, Megan R. McCullah-Boozer, Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA), and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

On 10 November 2022, measurements were made of the Atlas V JPSS-2 rocket launch from SLC-3E at Vandenberg Space Force Base, California. Measurements were made at 11 stations from distances of 200 m to 7 km from the launch pad. Measurement locations were arranged at various azimuthal angles relative to the rocket to investigate possible noise asymmetry. This paper discusses preliminary results from this measurement including overall levels, temporal and spectral characteristics, evidence of nonlinear propagation, and potential azimuthal asymmetry effects.

11:10

1aNS10. Initial comparison of RUMBLE predictions with measurements for an Atlas V rocket launch. Mark C. Anderson (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, anderson.mark.az@gmail.com), Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Reese D. Rasband (Ball Aerosp., Beavercreek, OH), Alan T. Wall (Air Force Res. Lab., Wright Patterson Airforce Base, OH), and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

As commercial space launches continue to become more frequent, increasing attention will be given to the noise generated during these launches. Of particular concern are nearby communities, launch infrastructure, and wildlife. To understand the impacts of such launches, noise models have been developed. One commonly used software package is known as RUMBLE. This paper aims to calculate noise metrics from measurements of a United Launch Alliance Atlas V rocket and compare those to RUMBLE-predicted values. Locations were measured ranging in the hundreds of meters to several kilometers and the compared metrics will include the A-weighted and unweighted maximum sound level and the sound exposure level. This analysis is part of an overall goal to determine the prediction accuracy and determine how future models may be improved as the cadence of rocket launches continues to rise. [Research funded in part by the Oak Ridge Institute for Science and Education.]

Session 1aPA

Physical Acoustics: Acoustic Remote Sensing in Urban Environments

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

Sandra Collier, Cochair
U.S. Army Research Laboratory, Adelphi, MD 20783

Invited Papers

8:10

1aPA1. Compound variance gamma distribution for randomly scattered sound and noise at multiple sensors. D. Keith Wilson (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), James G. Ronan, and Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Gamma distributions for signal intensity (or, equivalently, Nakagami distributions for signal amplitude) are often used in acoustics and electromagnetics to describe the single-point statistics of randomly scattered signals and noise in urban and other complex environments. An extension of the gamma distribution, called the *compound gamma* (CG), has been furthermore shown [D. K. Wilson, M. J. Kamrath, C. E. Haedrich, D. J. Breton, and C. R. Hart, "Urban noise distributions and the influence of geometric spreading on skewness," *J. Acoust. Soc. Am.* **150**(2), 783–800 (2021)] to usefully generalize the gamma to scenarios in which the distribution becomes skewed due to random variations in the scattering strength or loudness of the noise sources. Here we present the *compound variance gamma* (CVG), which further generalizes the compound gamma to two-point statistics and is, therefore, useful for representing statistics on arrays of sensors or networks. The CVG distribution, which is expressed with a Gauss hypergeometric function, is shown to be in excellent agreement with realistic simulations of sound scattering in the atmosphere.

8:30

1aPA2. Spatial variation of exponentially modified Gaussian parameters in an urban setting. Matthew J. Kamrath (US Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, matthew.j.kamrath@erdc.dren.mil), D. Keith Wilson (US Army Engineer Res. and Development Ctr., Hanover, NH), Caitlin E. Haedrich (Ctr. for Geospatial Analytics, North Carolina State Univ., Raleigh, NC), and Carl R. Hart (US Army Engineer Res. and Development Ctr., Hanover, NH)

The distribution of urban acoustic noise levels varies spatially because of differences in traffic volume, distance to other sound sources (e.g., construction sites), urban canyon geometry, and several other factors. We want to connect these environmental variables to the parameters of a parametric probability density function of the acoustic levels. Previously, the exponentially modified Gaussian (EMG) distribution matched experimental observation more closely than other distributions for urban sound levels. Using the framework of common engineering methods (e.g., ISO 9613-2), the received pressure level is the source power minus several attenuation factors. The exponential distribution could be due to the attenuation from geometrical spreading because it is a good approximation for a source uniformly distributed on a line segment or within a circle. The Gaussian distribution could be due to multiple factors, including weak turbulent scattering, reflections and diffractions, multiple sources, and source power variability. We measured the sound levels at 37 locations in the North End of Boston (USA), plotted the spatial variation in the EMG parameters, and compared these maps to the environmental variables. The traffic volume, proximity to stationary sources, and scene geometry all seemed to impact the EMG parameters with the traffic volume being the most important.

8:50

1aPA3. Mobile sensing for the localization and tracking of acoustic sources in urban environments. Dorian Davis (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, dorian.davis1@udc.edu), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, the localization and tracking of acoustic sources in an urban environment is presented. Of particular interest is the performance of mobile acoustic sensors. In real world situations, sources are moving making it difficult to track their location especially in changing environments. The performance of mobile acoustic sensors is investigated. Beamforming and TDOA based techniques will be employed in various non-line of sight scenarios. The results and limitation of these techniques both numerically and experimentally will be discussed.

9:10

1aPA4. Acoustic sensing of wind generated vortices in urban environments. Samba Gaye (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, samba.gaye1@udc.edu), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, generated vortices within an urban microspaces are investigated numerically and experimentally. Of particular interest are vortices within an urban street canyon. Computational fluid dynamics simulations of wind vortices are compared to experimental observation of similar urban street geometries within the District of Columbia. Additionally, the vortex and ground interaction are investigated for acoustic propagation. Preliminary results are presented and discussed.

Contributed Papers

9:30

1aPA5. Wind-induced noise estimation of wind speed and direction within urban microspaces: Characterizing flow condition of urban environments. Lirane Mandjoupa (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, max.denis2@udc.edu), Samba Gaye, Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, the flow conditions within urban microspaces are predicted from wind-induced noise. Wind speed and direction are estimated by fitting the spatial coherence of the wind noise distribution among closely spaced microphones measurements to the Corcos model in the least squares sense. The estimated wind speed and direction are compared to measure results. The estimated flow conditions of the measured urban areas are discussed.

9:45

1aPA6. Model based spectral feature analysis in reverberant acoustic fields. Emi Aoki (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, 1 Univ. Ave. Lowell, MA 01854, emi_aoki@student.uml.edu), Pratik Gandhi, Flore Norceide, Kavitha Chandra, and Charles Thompson (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA)

The spectral features of a reverberant acoustic field that can lead to improved source localization methods are investigated. Of particular interest are identification of deterministic and probabilistic features that characterize the environment and models of the acoustic transfer function that generate such features. For example, models that predict Schroeder's invariant standard deviation in the pressure response at source-receiver distances where the direct sound approaches the reverberant field energy are proposed. Models examined include non-stationary Gaussian processes for the acoustic reverberant components and all-pass parametric models of the acoustic impulse response.

10:00–10:15 Break

10:15

1aPA7. Remote acoustic sensing for urban air quality assessment. Juan Estevez Hernandez (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, juan.estevezhernande@udc.edu), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, sound pressure levels and air quality are measured around near roadways in urban areas. The measured sound pressure level data are compared to the air quality index to investigate the relationship between roadway traffic levels and particulate matter concentration. Additionally, the effects of wind speed and direction are investigated. Of particular interest are the upwind and downwind conditions. The results of this work will be presented and discussed.

10:30

1aPA8. Collaborative acoustic sensing for autonomous vehicles in GPS denies areas: Application to urban environments. Herve Sandja Tchamba (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, hervecabrel.sandjatc@udc.edu), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, an acoustic sensing collabortive localization approach for autonomous vehicles is presented. Due to the autonomous nature of the vehicles, its environment and location must be known. Therefore, it is assumed that the autonmous vehicles share their movement information among each other. In addition to this information, we propose that acoustic self-localization measurements be coupled with a dead recokning in order to estimate the relative position and velocity of itself and other vehicles. Simulated and experimental results will be presented and discussed.

10:45

1aPA9. Gunshot detection from audio excerpts of urban sounds using transfers learning. Jamelia Ancel (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, jamelia.ancel@udc.edu), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, a pre-trained convolutional neural network is employed to detect gunfire from audio excerpts of urban sounds. The pretrained convolutional neural network is fined-tuned with transfer learned features to a new task using a smaller number of training signals. Two CNN methods are applied to the time-frequency representation of the audio signals. The first CNN method is based on classifying specific events in audio signals. The second CNN method is an image-based analysis method. The accuracy of the two CNN results will be compared and analyzed based on gunfire type and retrieved urban multipath conditions. A k-means clustering algorithm is employed to identify gunfire types and parametric modeling to retrieve the urban multipath conditions.

11:00

1aPA10. Characterization of urban microclimates from reconstructed wind velocity and temperature fields. Glacia Martin (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, glacia.martin@udc.edu), Lirane Mandjoupa, Samba Gaye, Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

This work aims to study microclimate changes within urban microspaces using atmospheric acoustic tomography (AAT). Of particular interest are the imaging of wind velocity and temperature field variations. To reduce the inaccuracy of using the conventional AAT straight ray model, a nonlinear reconstruction algorithm is employed. Simulated and measurement results are presented and discussed.

11:15

1aPA11. Cooperative verification of urban noise sources from generated acoustic and seismic waves. Telha Abdulbasit (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, abdulbasit.telha@udc.edu), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, co-operative verification of acoustic and seismic wave generating noise sources within an urban environment is presented. Sound and soil vibration produced from everyday urban noise sources including trains and vehicles are measured. Acoustic and seismic sensors can detect vehicles passively, independent of weather and daylight, at long ranges. Both signals contain contributions from the engine. Simulated and measured results will be presented and discussed

11:30

1aPA12. Classifying urban areas from learned acoustic and seismic data. Hans Matthew Baes (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, hansmatthew.baes@udc.edu), Herve Sandja Tchamba, Telha Abdulbasit, Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, a deep transfer learning algorithm is used to classify different urban areas from acoustic and seismic data. The deep transfer learning model combines Google's deep learning model AlexNet. Measurements of acoustic and seismic ambient noise were conducted in an urban environment. The urban acoustic and seismic measurements are heavily influenced by traffic noise but exhibit more variation with respect to urban location due to the influence of subsurface conditions. A K-means clustering analysis is employed on the acoustic and seismic spectrogram to classify the urban areas. The results demonstrate that seismic and acoustic data can have similar cluster centroid locations in the frequency band of 4 Hz–10 Hz. The transfer learning results will be presented and discussed.

11:45

1aPA13. Acoustic energy harvesting of ambient noise urban environment. Justin An (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, max.denis@udc.edu), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, the efficiency of engineered materials for converting ambient acoustic noise into electrical power is investigated. Of particular interest are 3D-printed acoustic metamaterials. Initial investigation will focus on sonic crystals acoustic metamaterials. Numerical simulations of the acoustic wave and sonic crystals interaction will be conducted. Of particular interests are the strong scattering interactions. 3D-printed samples will be measured and tested. The simulation and measurement results will be presented and discussed.

MONDAY MORNING, 8 MAY 2023 SALON I, 8:00 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session 1

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

All posters will be on display from 8:00 a.m. to 12:00 noon. Authors of odd numbered abstracts will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even numbered abstracts will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

1aPP1. Individual variability in speech recognition with compression and beamforming at positive signal-to-noise ratios. Varsha H. Rallapalli (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, varsha.rallapalli@northwestern.edu), Richard Freyman (Commun. Disord., Univ. of Massachusetts, Amherst, MA), and Pamela E. Souza (Northwestern Univ., Evanston, IL)

Individuals with hearing loss often struggle to communicate in noisy conditions. Aggressive hearing aid (HA) processing, such as fast-acting wide dynamic range compression (WDRC), may reduce speech recognition under some circumstances, and there is concern about disproportionate reductions in listeners with poor working memory. We investigated how speech recognition is affected when WDRC is combined with highly directional beamforming intended to reduce noise from lateral and rear locations. We also sought to determine how listener factors—working memory, age, and hearing loss—affect such interactions in typical listening conditions, including positive signal-to-noise ratios (SNR) and different spatial noise locations. To-date 22 listeners with mild to moderately severe sensorineural hearing loss repeated low-context sentences (0°) mixed with six-talker interference at $+3$ and $+8$ dB SNRs, distributed equally in two ($+/-90°$) or three ($+/-90°/180°$) locations. Wearable HAs presented four signal processing combinations, matched for audibility: microphone directionality (omnidirectional or beamformer) and WDRC (fast- or slow-acting). Data show higher speech recognition with beamforming (re omnidirectional), especially at $+3$ dB SNR, regardless of WDRC and spatial locations. Relative contributions of listener factors will be assessed in mixed models in the context of combined signal distortions from HA processing and interferers. [Work supported by NIH-K01DC018324.]

1aPP2. Comparing outcomes with the cloud-based and portable versions of an open-source hearing aid. Varsha H. Rallapalli (Commun. Sci. & Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, varsha.rallapalli@northwestern.edu), Martin Hunt (Nadi, LLC, West Lafeyette, IN), and Harinath Garudadri (Univ. of California San Diego, San Diego, CA)

The open-source speech processing platform (OSP) is an open-source hearing aid (HA), developed to overcome proprietary restrictions of commercial HAs and pave the way for innovative HA technology through rigorous and reproducible research. This study evaluates portable and cloud-based platforms of the OSP that will enable a wide range of lab and field applications. The two platforms share common software applications for HA signal processing and objective outcome assessments (HO2). The cloud-based platform is hosted on a web server whereas the portable platform is accessed on a browser via a WiFi-enabled portable communication device. Listeners with mild to moderately severe hearing loss or normal hearing completed a word identification task (Modified Rhyme Test) in a 6-alternate forced choice paradigm using the HO2 app. Words were mixed with different levels of speech-shaped noise; presented via calibrated headphones or behind-the-ear receivers-in-the-canal for cloud-based and portable platforms, respectively. Signals were processed with individualized gain and compression ratios based on a clinically validated prescriptive formula. Results-to-date from four listeners show comparable range of word identification scores between platforms. The full dataset will also compare phoneme-level scores and response time measures generated from the HO2 app. Clinical research applications will be discussed. [Work supported by NIH-R44DC020406.]

1aPP3. Laboratory measurements for objective evaluation of earplug fidelity. David A. Anderson (Elec. Eng., Univ. of Minnesota Duluth, 5230 Otsego St., Duluth, MN 55804, and10445@d.umn.edu)

Musicians and concertgoers can easily be exposed to dangerously high noise doses that lead to hearing loss, and earplug use is often encouraged during loud musical performances. Traditional earplugs for industrial use can distort musical signals reaching the ear to the extent that users find their use disorienting and uncomfortable. High fidelity earplugs are designed to preserve musical sounds better than traditional earplugs by presenting a flat spectral attenuation of sound reaching the ears. At present, it's not clear how to quantitatively compare different models in an analogous way to other devices in the high-fidelity sphere such as loudspeakers and amplifiers. In this presentation, a suite of laboratory measurements and derived metrics are proposed that lead to a comprehensive evaluation of earplug fidelity in terms of insertion loss levels, flatness, polar distortion maps, and temporal skewing. Initial results for a range of devices are presented, followed by a discussion of future directions.

1aPP4. Monte Carlo simulation of pure tone audiometry reveals threshold-dependent differences in error. Trevor T. Perry (VA National Ctr. for Rehabilitative Auditory Res., 4954 North Palmer Rd., Bldg. 19, Fl. 5, Rm 5607, Bethesda, MD 20889, trevortperry@gmail.com) and Douglas S. Brungart (National Military Audiol. and Speech Pathol. Ctr., Walter Reed, Rockville, MD)

This study used a Monte Carlo simulation to evaluate the reliability of pure tone audiometry while incorporating information about the population distribution of thresholds. Results confirm that audiometric error (outcome threshold minus modeled threshold) is symmetric when simulated data are binned by modeled threshold. However, when data are binned by outcome threshold, the error depends on the relative distribution of modeled thresholds. When a realistic distribution is used, outcome thresholds of 0 dB HL show the lowest error rate. Bins with outcome thresholds <0 dB HL show a negative median error (outcome threshold < modeled threshold) and bins >0 dB HL show a positive median error (outcome threshold $>$ modeled threshold). These results suggest that a substantial proportion of individuals who obtain a very good threshold during audiometric testing (-5 or -10 dB HL) are individuals who actually have higher thresholds, which are being underestimated. These findings are relevant for hearing conservation programs, which monitor for upward shifts from baseline thresholds. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

1aPP5. Influence of demographic factors on the agreement between audiometric hearing loss and Revised Hearing Handicap Inventory self-reported hearing difficulty. Lauren Dillard (Otolaryngology- Head and Neck Surgery, Medical Univ. of South Carolina, 39 Sabin St., Charleston, SC 29425, dillalau@musc.edu), Lois J. Matthews, and Judy R. Dubno (Otolaryngology- Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

This study evaluated the agreement between audiometric hearing loss and Revised Hearing Handicap Inventory (RHHI) self-reported hearing difficulty, and modification by demographic factors (age, sex, race, socioeconomic status [SES]) in adults across the lifespan. Baseline data from the Medical University of South Carolina Longitudinal Cohort Study of Age-Related Hearing Loss (ARHL), a community-based cohort study of ARHL (1988-current) were used. Audiometric hearing loss was defined as pure-tone average (0.5–4.0 kHz) >25 dB HL in the better ear. RHHI hearing difficulty was defined as a score ≥ 6 on the full and screening (RHHI-S) versions. Sensitivity, specificity, and accuracy evaluated ability of the RHHI(-S) to detect audiometric hearing loss. The number of participants was 1558 (56.9% female; mean age 63.7 [range 18-89 +] years). Prevalence of audiometric hearing loss, RHHI, and RHHI-S hearing difficulty was 38.2%, 48.8%, and 44.6%, respectively. Sensitivity, specificity, and accuracy were similar for RHHI and RHHI-S hearing difficulty (versus audiometric hearing loss). The RHHI(-S) was less sensitive in detecting audiometric hearing loss in older and non-White adults. Sensitivity was similar by sex and SES but specificity was lower in males and high/low (vs mid) SES. Agreement between audiometric hearing loss and RHHI(-S) hearing difficulty is modified by demographic factors.

1aPP6. The effects of background noise and reverberation on speech perception in cochlear implant users. Abbie A. Mollison (Boystown National Res. Hospital, 555 N 30th St., Omaha, NE 68131, abbie.mollison@boystown.org) and Z. Ellen Peng (Boys Town National Res. Hospital, Omaha, NE)

Indoor noise and reverberation interfere with speech understanding for listeners. Degradation in cochlear implant (CI) processing presents additional challenges for CI users to understand noisy and reverberant speech. In this study, we investigated the effect of adverse acoustics on speech perception on three tasks that changed in difficulty and increasing need for top-down processing: (1) word recall, (2) sentence recognition, and (3) conversation comprehension. Each task was tested in five conditions that varied in combinations of adverse acoustics from reverberation (i.e., anechoic, 0.4s RT, vs 1.2s RT) and noise (i.e., quiet versus Speech-shaped-noise at -3 dB). We compared performances from normal-hearing (NH) listeners and bilateral CI users between 15 and 25 years old. Preliminary results show that while NH listeners may have some tolerance for speech perception under adverse acoustics, CI users experienced more graded performance deterioration with increasing reverberation and noise. Moreover, we observed task-dependent performance change across adverse acoustic conditions in both NH and CI listeners: While some context (i.e., sentences) may help speech perception, comprehension based on pragmatic language processing (i.e., conversations) is vulnerable under difficult listening situations. These results provide insights on how NH and CI listeners differ in their access to top-down processing in real-world speech perception.

1aPP7. From sounds to scenes: Comprehension of environmental sounds by cochlear implant users. Emily A. Price (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 W. Paulina St., 1015 AAC, Chicago, IL 60614, emily_a_price@rush.edu), Louisa M. Forrest (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), Nathan Luzum, Michael S. Harris (Otolaryngol. & Commun. Sci., Medical College of Wisconsin, Milwaukee, WI), Laurie M. Heller (Psych., Carnegie Mellon Univ., Pittsburgh, PA), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chiago, IL)

Performance outcomes for cochlear implant (CI) users traditionally focus on measures of speech perception. However, existing research indicates that environmental sound identification tasks also remain challenging for adult CI users compared to normal-hearing (NH) or hearing-impaired (HI) peers. In contrast, anecdotal reports indicate that environmental sound perception improves post-implantation. Methodological choices may contribute to this discrepancy; CI users may be benefiting from more integrative higher-level processes that are not adequately measured by source identification tasks. This study employs two alternative tasks that are designed to assess perception of the semantic properties of environmental sounds. The first is a comprehension tasks, which requires listeners to make inferences about naturalistic sound scene recordings (e.g. which other activities might you expect to take place, or at which time of day does this scene most likely occur). The second task presents a triplet of isolated environmental sound recordings and requires participants to select the sound that does not belong. Preliminary data indicate that CI users are able to perform the tasks with varying levels of proficiency relative to NH and HI listeners. Comprehension tasks focused on context-dependent semantic processing may, thus, complement findings from more traditional single-sound identification tasks.

1aPP8. Results of the second "clarity" enhancement challenge for hearing devices. Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Hearing Sci. Bldg., University Park, Nottingham NG7 2RD, United Kingdom, michael.akeroyd@nottingham.ac.uk), Jennifer L. Firth, Graham Naylor (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom), Jon P. Barker (Dept. of Comput. Sci., Univ. of Sheffield, Sheffield, United Kingdom), John Culling (School of Psych., Cardiff Univ., Cardiff, United Kingdom), Trevor J. Cox (Acoust. Res. Ctr., Univ. of Salford, Salford, Greater Manchester, United Kingdom), Will Bailey (Dept. of Comput. Sci., Univ. of Sheffield, Sheffield, United Kingdom), Simone Graetzer (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), Rhoddy Viveros Muñoz (School of Psych., Cardiff Univ., Cardiff, United Kingdom), Eszter Porter, and Holly Griffiths (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom)

The clarity enhancement challenges (CECs) seek to facilitate development of novel processing techniques for improving the intelligibility of speech in noise for hearing-aid users through a series of signal-processing challenges. Each challenge provides entrants with a set of stimuli for development and testing of their algorithms. The performance of the algorithms is assessed using objective measures of speech intelligibility and subjective measures conducted with a panel of hearing-impaired listeners. CEC2 featured more complex listening environments than CEC1 with multiple interfering sound sources (speech, music, household appliance sounds) within a simulated living-room environment at signal-to-noise ratios (SNRs) from -12 to $+4$ dB. In addition, head rotation towards the target speech was introduced. Target speech came from a new dataset of 10,000 different English sentences spoken by 40 actors speaking 250 sentences each (Graetzer S *et al*., 2022 Data in Brief 41, 107961). The objective assessment was provided by HASPI (Kates & Arehart, 2021 Speech Comm 131, 35–46). All 18 entries achieved substantial improvements in HASPI, averaging 0.55 across all systems and SNRs. Improvements were greatest (averaging 0.61) for SNRs between -8 and 0 dB. The best-performing system achieved HASPI scores above 0.9 for all SNRs. Listening-test data will be reported.

1aPP9. Utilization of acoustic cues to identify emotional prosody: Results in adults with normal hearing or cochlear implants. Aditya M. Kulkarni (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, aditya.kulkarni@boystown.org), Jessica Combs (Univ. of South Florida, Tampa, FL), Denis Fitzpatrick, and Monita Chatterjee (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Emotional prosody is a dominant source of information about a talker's intended emotion. Multiple acoustic cues contrast different emotions. Voice pitch is a dominant acoustic cue for prosody, but individuals with cochlear implants do not have adequate access to this cue. Previous work has suggested that older individuals with normal hearing also have limited access to dynamic voice pitch contour cues. We are developing a cue-weighting method to estimate perceptual weighting of different prosodic cues in emotion identification by listeners with cochlear implants or normal hearing. Initial analyses indicate that in adult normally hearing listeners, increasing age reduces their weighting of voice pitch cues to emotion, but not their weighting of the secondary cue of duration. The perceptual weight for voice pitch is reduced when normally hearing listeners are attending to spectrally degraded (noise band vocoded) stimuli. Consistent with the latter observation, cochlear implant patients showed reduced weighting of voice pitch cues compared to normally hearing counterparts when both are attending to original (clean) stimuli. Taken together, these results suggest that increasing age and spectral degradation both reduce listeners' ability to utilize voice pitch cues to emotional contrasts.

1aPP10. Audio-visual scene analysis in conditions with head- and eye-steered beamformers in virtual reality. Axel Ahrens (Tech. Univ. of Denmark, Ørsteds Plads 352, Kgs. Lyngby 2800, Denmark, aahr@dtu.dk), Adam Westermann (WS Audiol., Lynge, Denmark), Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), Torsten Dau (Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), and Tobias Neher (Univ. of Southern Denmark, Odense, Denmark)

In crowded social settings, listeners often face the challenge of following a conversation in the presence of other conversations. Several factors influence the difficulty of this task, including the number of talkers, the amount of reverberation, and the hearing status. Beamformers in hearing aids have the potential to mitigate these factors by improving the signal-to-noise ratio, but their effectiveness in real-world settings has not yet been clearly demonstrated. Here, we used virtual reality to investigate the effect of head- and eye-steered beamformers on the ability of participants to analyze complex audio-visual scenes. The participants' task was to find and locate an ongoing story in a mixture of other stories in scenes differing in terms of the number of concurrent talkers and the amount of reverberation. The talkers were distributed in the frontal hemisphere between $\pm 105°$. The primary outcome measure was the time taken to identify the location of the target talker. Preliminary results show shorter response times with beamforming (in comparison to an omnidirectional setting), especially when more talkers were present. This framework provides a new means for examining the effects of hearing technologies on behavior in complex audio-visual scenes.

1aPP11. Real-time measurement of ambient noise levels during hearing screening in point-of-care settings. Evelyn Davies-Venn (SLHS, Univ. of Minnesota, 119C Shevlin Hall, 164 Pillsbury Dr SE, Minneapolis, MN 55455, venn@umn.edu), Odile Clavier (Creare LLC, Hanover, NH), and Kristi Oeding (Univ. of Minnesota - Twin Cities/Minnesota State Univ. - Mankato, Mankato, MN)

Access to hearing healthcare is a global issue that will necessitate several novel approaches to resolve. By evaluating the effectiveness of a hearing screening program in a non-traditional, acoustically hostile point-of-care setting, this study contributes to the global need for access to hearing healthcare. Specifically, this study measured the noise environment of a hearing screening conducted at the Minnesota State Fair, using a Sivantos Dosimeter and a custom Tabsint program, to measure ambient noise levels during hearing screenings using the Creare Wireless Automatic Hearing Testing System (WAHTS). The noise levels were measured and analyzed to determine the average and peak noise levels encountered during the hearing screenings. Normal hearing thresholds were obtained at ambient noise levels ranging from 40 to 65 dBSPL. These findings were compared to the recommended noise levels for hearing testing to determine whether current standards should be revised considering innovations aimed at improving access to hearing healthcare services in point-of-care settings.

1aPP12. A functional test for hearing aids outcomes assessment. Harinath Garudadri (Qualcomm Inst., UC San Diego, San Diego, CA, hgarudadri@eng.ucsd.edu), Martin Hunt (Nadi, Inc., West Lafeyette, IN), Aadyanjali Daita (11th Grade, Del Norte High School, San Diego, CA), Anusha Yellamsetty (Dept. of Audiol., San Jose State Univ., San Jose, CA), and Varsha H. Rallapalli (Commun. Sci. & Disord., Northwestern Univ., Evanston, IL)

We present a functional test for speech understanding called hearing aids (HAs) objective outcomes – HO_2 (pronounced as /hɑːt/) app. It uses natural speech stimuli and generates word level accuracies, phonetic, and broad-phonetic confusion matrices. In the current best practices, hearing loss (HL) is diagnosed using pure tone audiometry (PTA). While PTA is also a functional test, it characterizes only sound detection ability of discrete pure tones, but not overall speech intelligibility due to cochlear impairments leading to sensorineural hearing loss. HO_2 app stimuli are based on minimal contrast sets (MCS) of words in a given language. Each word within a subgroup of MCS words differ in a minimal set of (acoustic) phonetic features resulting in a different meaning—phonemics of the given language. The tool includes configuration parameters to include video and closed captions to minimize the role of higher level (cortical) language processes. Researchers can save HO_2 app experimental conditions in a configuration file and share with others for repeatability and reproducibility. We present the feasibility of normal and hearing-impaired subjects (mild and moderate HL) to self-administer speech comprehension tests in multiple listening conditions using HO_2 app. For the hearing-impaired subjects, results in unaided and aided conditions are presented.

1aPP13. Effects of spectral resolution and smearing on gated word recognition. Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., 326 Stevenson Ln., B8, Towson, MD 21204, cpatro@towson.edu), Ariana Bennaim, and Nirmal Kumar Srinivasan (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

In a word gating task, listeners are presented with increasing amounts of word-onset information (a series of increasingly longer temporal "gates"), and following each gate, they are asked to indicate what they think the target word is. Listeners with normal hearing (NH) typically recognize a target word well before they "hear" the entire word. Listeners with cochlear implants (CIs), however, need to hear almost the entire target word (require a greater amount of word-onset information) to recognize it. Here, we hypothesized that the poor spectral resolution, due to the limited number of channels and interactions between adjacent channels, may have a negative impact on gated word recognition performance. We manipulated spectral resolution by using: (1) a noise-band vocoder with a variable number of spectral channels; and (2) a vocoder with variable carrier filter slopes to simulate channel interaction. We determined the minimum amount of word-onset information required to recognize spoken words. Initial results suggest that the gated word recognition performance remained roughly unchanged as the amount of spectral degradation applied increased up to some extent, beyond which it deteriorated. With eight channels, which resemble the spectral resolution available to most CI users, even the most focused stimulation yielded poorer results.

1aPP14. An adaptive Bayesian algorithm for efficient auditory brainstem response threshold estimation: Numerical validation. Erik A. Petersen (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, erikp2@uw.edu) and Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

The auditory brainstem response (ABR) can be used to estimate the hearing threshold of animals or human subjects who are unable to respond to behavioral measures. However, ABR can be time-intensive and is vulnerable to human subjectivity while interpreting the waveforms. An adaptive procedure has been developed to efficiently and objectively estimate ABR threshold. The procedure iteratively fits a Gaussian process (GP) model to the ABR waveforms collected so far and optimizes the subsequent stimulus. The algorithm was validated through numerical simulation using pre-collected human ABR at 0.5, 1.0, 2.0, and 4.0 kHz with levels from below threshold to 90 dB in 5 dB increments from a cohort of normal-hearing listeners. This led to a full stimulus space of approximately 55 stimuli per test ear. For each test ear, the ABR threshold was estimated by human raters based on the entire stimulus space. The ABR threshold was also estimated using the adaptive procedure, which iteratively sampled a subset (∼35%) of the stimulus space. At the end of the adaptive procedure, the fitted GP model was able to capture the individual differences in waveform morphology between listeners. Furthermore, the threshold estimates from repeated runs of the adaptive procedure demonstrated adequate test-retest reliability.

1aPP15. Complementary measures for clinical audiological practice to better explain self-reported hearing difficulties: A study using an automated measurement tool in Mexico. E. S. Lelo de Larrea-Mancera (Cognit. and Clinical Neuropsychology, Instituto Nacional de Neurología y Neurocirugía Manuel Velasco Suárez, Ocaso 85, Insurgentes Cuicuilco, Coyoacán, Cdmx 04530, Mexico, elelo001@ucr.edu), Rodolfo Solís-Vivanco (Cognit. and Clinical Neuropsychology, Instituto Nacional de Neurología y Neurocirugía Manuel Velasco Suárez, Cdmx, Mexico), Yolanda Sánchez Jiménez (Oto-neurology, Instituto Nacional de Neurología y Neurocirugía, Mexico City, Cdmx, Mexico), Katia Padilla-Bustos (Cognit. and Clinical Neuropsychology, Instituto Nacional de Neurología y Neurocirugía Manuel Velasco Suárez, Mexico City, Mexico), Yadira Chimal-Gómez, M. Guadalupe Padilla-Canales (Oto-neurology, Instituto Nacional de Neurología y Neurocirugía, Mexico City, Mexico), Erick Correa-Medina (Cognit. and Clinical Neuropsychology, Instituto Nacional de Neurología y Neurocirugía Manuel Velasco Suárez, Mexico City, Mexico), David S. Clamage (Brain Game Ctr., Univ. of California Riverside, Riverside, CA), William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and Aaron Seitz (Brain Game Ctr., Univ. of California Riverside, Riverside, CA)

Accessibility issues, especially in developing countries, have prevented the evaluation of diverse hearing abilities beyond the pure tone audiogram in the general population. PART is a free-access automated auditory measurement tool that can address accessibility issues by providing a wide variety of auditory psychophysical measures in multiple languages on standard relatively low-cost commercial interfaces such as tablets and smart phones. Here, we show that PART can complement traditional audiological tests to better account for people's hearing complaints. Forty-four adults between 40 and 80 years old were tested in Mexico City with a traditional clinical battery (otoacoustic emissions and pure tone audiogram) and a battery of PART-based central auditory processing and speech-on-speech competition measures. Several measures from the PART battery correlated with self-reported hearing difficulties, and the spatial release from masking task was best able to capture this variance. These preliminary data support the inclusion of a speech-on-speech masking as a complement to traditional clinical measures to better account for the hearing complaints of the public. Furthermore, we show PART-based version of this test can be implemented in a clinical setting in Mexico and produce comparable results to those obtained in developed countries like the United States of America.

1aPP16. Auditory brainstem responses to peaky speech using zero- versus chirp-phases. Melissa J. Polonenko (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN, mpolonen@umn.edu) and Samantha Krocak (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Auditory brainstem responses (ABR) are used to identify hearing loss across multiple frequency bands in young children. ABRs typically use brief tonebursts or chirps, but we recently created a "peaky" speech method that uses stories to evoke ABRs—an engaging stimulus that may facilitate testing in infants and toddlers who cannot nap, sit still, or participate in behavioral testing. This study aimed to determine which phase profile (zero versus chirp) of multiband peaky speech evokes the largest ABRs in the fastest recording time. Fifteen adults with normal hearing listened to an hour each of zero- and chirp-phase multiband peaky speech while 2-channel ABRs were recorded. Overall, ABR wave V amplitudes were larger for chirp- than zero-phase multiband peaky speech, especially for lower frequency responses, and took 30 versus >60 minutes for all responses in the 5 bands (0.5–8 kHz) and both ears to reach a criterion signal-to-noise ratio. In summary, chirp-phase peaky speech provided larger—and thus, more effective and efficient—multiband ABRs than zero-phase peaky speech for ABR testing using continuous speech. Reducing testing time in half will be important for future application as an ABR hearing screener in toddlers using engaging stories. [Work funded by Hearing Health Foundation ERG.]

1aPP17. Optimal f0–phase parameters for auditory brainstem responses to multiband peaky speech. Melissa J. Polonenko (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN, mpolonen@umn.edu) and Benjamin Eisenreich (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Multiband peaky speech is a new method that uses narrated stories to evoke frequency-specific auditory brainstem responses (ABRs). However, testing times are not clinically feasible; thus, new efforts focus on maximizing response amplitude to facilitate faster testing. Recent work suggests that narrator f0, as well as chirp rather than zero phases, evoke larger ABRs. This study aimed to identify the optimal f0–phase combination for evoking large ABRs with multiband peaky speech. Using 11 stories with original f0s ranging from 114–230 Hz, the mean f0 over 15 minutes was shifted to f0s from 70–220 Hz in 10 Hz steps and then dichotic multiband peaky speech created using zero- and chirp-phases. A validated computational model of the auditory periphery was used to simulate ABRs using a flat 0 dB HL hearing configuration. Based on the simulated wave V amplitudes across all conditions, the optimal parameters for simulated ABRs to multiband peaky speech include stories with original f0s <170 Hz that have been shifted to a 100–110 Hz mean f0 and modified with a chirp-phase profile. Next we plan to confirm these parameters with measured ABR responses from adults with normal hearing. [Work funded by Hearing Health Foundation ERG.]

1aPP18. Psychophysical performance differences associated with auditory spatial attention in older listeners with and without hearing loss. Dana Cherri (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, dcherri@usf.edu) and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

In a busy listening environment with multiple speakers, even normal-hearing listeners may face difficulties in understanding speech (i.e., "the cocktail party problem"). Individual factors related to bottom-up and/or top-down processing difficulties may be associated with the listener's ability to attend to speech, a key component in navigating complex acoustic environments. The present study tested listeners on a battery of psychophysical measures and compared listener performance to cortical measures (EEG) of spatial-change tuning and auditory spatial attention. The behavioral tests measured both perceptual and cognitive performance and included: frequency modulation detection, spectro-temporal modulation sensitivity, spatial release from masking, temporal gap detection, auditory and non-auditory sustained attention, and visuo-spatial problem solving. The EEG paradigm tested the listener's ability to attend to moving speech in quiet or background babble in the free field. Older listeners with and without hearing loss (n = 20 per group) demonstrated wide individual performance for some measures and significant correlations between behavioral thresholds and cortical responses, consistent with a more refined view of age-related hearing loss and a need for a more holistic approach in diagnoses for individuals beyond standard clinical tests. [Work supported by NIH R21DC017832.]

1aPP19. Parametric voice difference benefit for speech-on-speech perception in adult cochlear implant users. Etienne Gaudrain (Lyon Neurosci. Res. Ctr., CNRS, Université Lyon 1, Ctr. Hospitalier Le Vinatier - Bâtiment 462 - Neurocampus, 95 bvd Pinel, Lyon 69000, France, etienne.gaudrain@cnrs.fr), Eleanor Harding (Dept. of Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Netherlands), Robert Harris (Prins Claus Conservatoire, Hanze Univ. of Appl. Sci., Groningen, Netherlands), Barbara Tillmann (Lyon Neurosci. Res. Ctr., CNRS, Université Lyon 1, Lyon, France), Bert Maat, Rolien Free, and Deniz Başkent (Dept. of Otorhinolaryngology, Univ. of Groningen, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

Speech-on-speech perception remains challenging for some cochlear implant (CI) users. In normal-hearing listeners, differences in voice characteristics such as F0 and vocal-tract length (VTL), which indicate age, sex, and size of the speaker, are known to support the segregation of competing voices. However, previous research indicated that postlingual adult CI users tend not to derive any perceptual benefit from voice differences, whereas prelingually deafened child CI users do. The current study tested speech-on-speech perception in adult CI users. In a coordinate response measure (CRM) paradigm, participants identified a number and color in a target speech stream competing with a gibberish speech masker. The masker voice was either identical to the female target voice, or the F0 and VTL were altered to create either an ambiguous fe/male voice, or a clearly male voice. The target-to-masker ratio (TMR) was also systematically manipulated to take the values: 0, + 6 and + 12 dB. While performance depended on TMR, only some of the CI users were able to benefit from voice differences. Results will be compared with F0 and VTL discrimination thresholds in the same participants, as well as with pediatric data in the same task.

1aPP20. Supra-threshold envelope-following responses in the ageing population: An early marker of sensorineural hearing damage. Sarah Verhulst (Ghent Univ., Technologiepark 126, Zwijnaarde 9052, Belgium, s.verhulst@ugent.be), Heleen Van Der Biest, Sarineh Keshishzadeh, Hannah Keppler, and Ingeborg Dhooge (Ghent Univ., Ghent, Belgium)

Envelope-following-responses (EFRs) evoked by supra-threshold amplitude-modulated sounds are promising markers of age-related or ototoxic-induced cochlear synaptopathy (CS) in research animals. A direct translation of these findings to EFR-based CS-quantification in humans is complicated by possible combinations of CS, inner-, outer-hair-cell damage or central deficits that can also affect EFR markers. To work towards a sensitive CS-marker for use in humans, we focus on an EFR stimulus that—in computational model simulations—is maximally sensitive to CS and investigate how this marker declines in an ageing population with or without OHC damage. 108 subjects participated in this study and were divided into two groups: (i) an ageing group with normal audiograms (i.e., 4-kHz-thresholds <20 dB HL, n = 89, 18–65 years) and (ii) older adults with impaired audiograms (n = 19, 45–75 years). We collected 120-Hz modulated EFRs, along with extended high-frequency audiograms, distortion-product otoacoustic emissions, and speech-reception thresholds (Flemish Matrix test). We calculated the rate of decline of each metric with age and conclude that the EFR marker predicts sensorineural hearing damage approximately 10 years earlier than the standard clinical audiogram. Additional relationships between audiometric, speech-intelligibility, and EFR markers are discussed. [Work supported by ERC 678120 RobSpear and 899858 CochSyn.]

1aPP21. Upward bias in audiometric thresholds is caused by stopping at the first opportunity. Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, 0100 LeFrak Hall, 7251 Preinkert Dr., College Park, MD 20742, ehoover@umd.edu) and Katherine N. Menon (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Tone-detection thresholds are the primary index of hearing. Although standard audiometry nominally estimates the stimulus level corresponding to 50% detection, thresholds are biased upward to an average of 98.1% at 500 Hz [Marshall, L. & Jesteadt, W. *J Speech Hear Res* **29**, 82–91. (1986)]. This has been explained by the large difference in the probability of detection between stimulus levels near threshold. However, bias is greater than predicted for measured psychometric functions and does not change proportionally with step size. In standard audiometry, the stopping rule is defined with flexibility for clinical judgement. Our hypothesis is that upward bias is caused by stopping at the first opportunity, because it makes specific ordered combinations of hits and misses impossible, like hit-hit-miss. Monte Carlo simulations were used to compare audiometric thresholds obtained using different stopping rules. Thresholds obtained when stopping at the first opportunity had greater upward bias compared to a fixed-trial stopping rule independent of the total number of trials presented, as predicted. When a baseline test is biased upward by stopping at the first opportunity, a follow-up test may inadvertently miss a significant change in hearing by continuing to present stimuli after the first opportunity to stop. [Work supported by R01DC015051.]

1aPP22. Preliminary findings from a study relating the effects of hearing loss and listening conditions on postural sway to fall risk in older adults. Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu), Shirley Rietdyk, Jeffrey Haddad (Health and Kinesiology, Purdue Univ., West Lafayette, IN), Melissa Newell (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN), Sharon Christ (Human Development and Family Sci., Purdue Univ., West Lafayette, IN), and Sarah Burgin (Otolaryngol. - Head & Neck Surgery, Indiana Univ. School of Medicine, Indianapolis, IN)

Hearing loss is associated with increased fall risk in older adults, but multiple mechanisms have been proposed to account for this. Hearing loss may reduce spatial awareness and/or increase cognitive load, both of which may increase fall risk but may be ameliorated by hearing aid use. Here, we present preliminary results from an in-progress study comparing fall incidence in daily life with postural sway measured under various listening conditions in older adults with and without hearing aids. Seventeen adults aged 65 to 81 years (7 using bilateral hearing aids, 10 without) stood with feet together and eyes closed, listening to noise vocoded speech (4, 8, 16 channels) and spatially distributed environmental sounds (silence, 1, 3 sources) for 1 min per condition while postural sway was recorded in the lab. Participants subsequently reported daily near-falls and falls for 4 months. Results suggest hearing aid users show less postural sway, potentially indicating greater rigidity which has been associated with greater fall risk. However, near-fall incidence was lower for hearing aid users. Further analyses showed higher near-fall rates associated with higher auditory thresholds, but hearing aid use may mitigate this trend. We are currently collecting more data with a wider variety of participants.

1aPP23. Localization and externalization of speech through hearing aids with adaptive features. Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., Commun. Sci. & Disord., 516 High St., MS 9171, Bellingham, WA 98225, anna.diedesch@wwu.edu), Virginia Best, Elin Roverud (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Hearing aids with advanced digital signal processing features that analyze acoustic scenes can amplify wanted sounds (e.g., speech) while utilizing complex noise reduction features to attenuate unwanted sounds (e.g., noise). These adaptive features can be programmed to be more or less aggressive in order to prioritize listening comfort or speech detail. Importantly, these adaptive features may alter the interaural cues of relevant target talkers located in front of the listener, causing unwanted effects on spatial perception. Here, we explored these effects. Spondees were presented from 11 different loudspeakers (located between $\pm 75^\circ$ azimuth separated by 15°) in an anechoic chamber. Binaural recordings were obtained from the ear canals of a Bruel & Kjaer Head and Torso Simulator, which was fitted with Oticon More hearing aids. The recordings were then presented over headphones to listeners, who indicated the perceived direction and degree of externalization of the spondees. Several conditions were tested, which differed in the acoustics of the room (anechoic and simulated reverberation), the background (quiet and noise), and the hearing-aid coupling (unaided, open-fit, and occluded). Behavioral data will be compared to measured interaural cues in the corresponding binaural recordings.

1aPP24. Externalization of speech through hearing aids differing in microphone position and dome type. Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave. Boston, MA 02215, ginbest@bu.edu) and Elin Roverud (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

An experiment was conducted to examine how different hearing aid styles affect the externalization of speech. Of particular interest were effects of microphone position (behind-the-ear versus in-the-canal) and dome type (closed versus open). Participants were young adults with and without hearing loss, who were fitted with hearing aids that allowed variations in the microphone position and the dome type. They were seated in a large sound-treated booth and presented with monosyllabic words from loudspeakers at a distance of 1.5 m. Their task was to rate the perceived externalization of each word using a rating scale that ranged from 10 (at the loudspeaker in front) to 0 (in-the-head) to -10 (behind the listener). We observed a large intersubject variability in ratings. On average, compared to unaided listening, hearing aids tended to reduce perceived distance and lead to more in-the-head responses. This was especially true for closed domes in combination with behind-the-ear microphones. We also observed a strong association between poor externalization and front-back confusions.

1aPP25. Bandwidth extension with air and body-conduction microphones for speech enhancement. Austin Lu (Univ. of Illinois Urbana-Champaign, 503 E Clark St., Champaign, IL 61820, austinl8@illinois.edu), Manan Mittal, Kanad Sarkar (Univ. of Illinois Urbana-Champaign, Urbana, IL), Ryan M. Corey (Univ. of Illinois Chicago, Chicago, IL), Paris Smaragdis, and Andrew C. Singer (Univ. of Illinois Urbana-Champaign, Urbana, IL)

Contact microphones are accelerometers placed on the skin to detect body-conducted speech, which is inherently robust to interfering sound sources. However, contact microphones are severely bandlimited. In contrast, standard air-conduction microphones capture all relevant frequencies from all sources. To partially recover lost frequency components of speech, we can use bandwidth extension, which uses formant information at low frequencies to reconstruct the spectral envelope and harmonic structure. We use a suite of bandwidth extension methods to combine high SNR, low bandwidth audio with low SNR, full bandwidth audio for speech enhancement.

1aPP26. Multiple-source dynamic range compression in hearing devices: What can we learn from music mixing? Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, 2021 Discovery Partners Inst., 200 S. Wacker Dr., 20th Fl., Chicago, IL 60606, corey1@uillinois.edu)

Dynamic range compression (DRC), a form of nonlinear gain control, is widely used in both hearing devices and music production, but in very different ways. In hearing devices, DRC boosts quiet sounds more than loud sounds to match the reduced dynamic range of listeners with hearing loss. Compression is applied to the mixture of sounds captured by the hearing device microphone, so it is known to cause distortion in noisy environments where the level of one sound can affect the gain applied to another sound. In music mixing, DRC can be applied to individual vocal or instrumental tracks or to the overall mixture. It can also be applied to combinations of signals to introduce deliberate distortion effects, for example, by adjusting the level of one track based on the level of another. In a hearing device that could process different sound sources independently, real-life sounds could be mixed like musical tracks, reducing unwanted distortion and enabling new nonlinear techniques to improve intelligibility. This talk will explore multiple-source DRC architectures that are commonly used for music but could also be advantageous for hearing devices.

1aPP27. Source separation using bandlimited external microphones and a microphone array. Manan Mittal (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1308 W Main St., #119, Urbana, IL 61801, manansm2@illinois.edu), Kanad Sarkar (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Austin Lu (Systems Eng., Univ. of Illinois, Urbana-Champaign, Champaign, IL), Ryan M. Corey (Discovery Partners Inst., Chicago, IL), and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Modern listening devices are equipped with air-conducted microphones and contact microphones. External microphones, like those on a listening device, have been used to estimate relative transfer functions (RTFs) at microphone arrays. With numerous active sound sources, the air-conducted microphones perform poorly while the contact microphones are robust to external noise. A drawback of contact microphones is that they are bandlimited. Past work has shown that the contact microphone and microphone array can be combined to estimate RTFs in the low frequencies. To overcome the limitations of the contact microphone, we propose a method that leverages the full-band signal at the microphone array to provide beamforming gains at higher frequencies. We demonstrate this method by separating three human talkers in a noisy environment.

1aPP28. The impact of hearing aid user's own voice on device signal processing. Robert A. Budinsky (Commun. Sci. & Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD 1017, Tampa, FL 33620, rbudinsky@usf.edu), Erol J. Ozmeral, and David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL)

Hearing aid processing is designed to improve audibility for sounds of interest, often by targeting external speech signals. During natural conversation, however, the hearing aid user is also the source of speech, potentially interacting with hearing aid function and leading to suboptimal processing. In this study, we investigated how the presence of own voice impacts the deployment of specific features designed to enhance audibility and intelligibility of conversational partners. We recorded real-time hearing aid feature engagement (directional microphones, noise reduction, and speech enhancement) during simulated conversations in quiet and in background noise. Simulations were performed using an acoustic manakin (GRAS 45BC-12) capable of producing speech via internal speaker, which was situated in the center of a 24-speaker spatial array. The results demonstrate how the presence of a hearing aid user's own voice disrupts the intended use of these adaptive features, and that there is a need for optimizing hearing enhancement devices to account for own voice in dynamic scenarios. Future studies will determine the impact of a hearing aid user's own voice on the device's ability to improve speech intelligibility and user satisfaction.

MONDAY MORNING, 8 MAY 2023 — CHICAGO H, 9:00 A.M. TO 11:05 A.M.

Session 1aSA

Structural Acoustics and Vibration, Engineering Acoustics, Noise, Signal Processing in Acoustics, and Physical Acoustics: Acoustic Excitation of Structures I

Brian E. Anderson, Cochair
Physics & Astronomy, Brigham Young University, Department of Physics & Astron., N245 ESC, Provo, UT 84602

Trevor Jerome, Cochair
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Ian C. Bacon, Cochair
Department of Physics & Astronomy, Brigham Young University, N283 ESC, Brigham Young University, Provo, UT 84602

Chair's Introduction—9:00

Invited Papers

9:05

1aSA1. The dynamics of partially encapsulated microbubbles subjected to ultrasound. Amit Dolev (Mech. Eng., Swiss Federal Inst. of Technol. Lausanne, EPFL STI IGM MICROBS, MED 3 2715 (Bâtiment MED) Station 9, Lausanne, Vaud 1015, Switzerland, amit.dolev@epfl.ch) and Selman Sakar (Mech. Eng., Swiss Federal Inst. of Technol. Lausanne, Lausanne, Vaud, Switzerland)

Acoustically excited microbubbles generate various nonlinear forces that can be leveraged in microscale systems for actuation and manipulation. To obtain optimal performances, bubbles should be characterized; however, so far, they were studied indirectly by measuring downstream phenomena. Here, we present a novel scheme to measure the vibrations of a bubble at the water-air interface using a laser vibrometer and the impinging pressure using a hydrophone. A custom-built optical setup couples the vibrometer to an inverted microscope. The microstructures encapsulating the bubbles are 3D nanoprinted on glass slides, which allows the realization of complex configurations with various polymers. The overall platform enables us to study the dynamics of bubbles with single or multiple interfaces, and their interactions. The measurements are also used to refine the analytical model of the multi-physics system and find optimal operating conditions. The conditions depend on the bubble geometry, boundaries, and excitation source. The measurements reveal that it is easier to excite certain vibration modes when the acoustic wavelength is much larger than the bubble. We demonstrate the controllable motion of 3D nanoprinted flexible structures by harnessing nonlinear forces that are produced by acoustically excited microbubbles. These structures will serve as building blocks for all-mechanical soft microrobots.

9:25

1aSA2. Intracochlear noise-induced vibrations. Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI, grosh@umich.edu), Wen Cai, and Vipin Agarwal (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

When designing an engineered electroacoustic sensor, a key question is "what is the lowest level sound that can be sensed?" To answer this, we design to achieve a desired input referred noise. In the cochlea, there are many sources of internal noise, such as Johnson noise arising from membrane conductances, clatter noise in channels, and thermoviscous damping that will conspire to cause vibrational responses in the absence of external stimulus. Noise can dramatically affect our ability to sense desired sounds. Experimental and theoretical cochlear mechanics has focused on determining the sensitivity of the cochlea. However, to our knowledge, there is only one set of published measurements of the displacement response of the cochlea in quiet (the levels are low, below 1 atto-meter^2 per Hz), and no simulations in a global cochlear model of the noise response. We introduce a method for predicting the global response to noise from electromotile outer hair cells in the cochlea for small fluctuations about equilibrium using our finite-element based numerical model. Furthermore, we show the relative contribution of different noise sources (in particular, channel noise versus conductance noise) and the spatial distribution of the response to these sources.

9:45

1aSA3. Abstract withdrawn.

10:05–10:20 Break

Contributed Papers

10:20

1aSA4. Vibro-acoustic characterization of the bone-implant system during insertion. Anne-Sophie Poudrel, Giuseppe Rosi, Vu-Hieu Nguyen (Multiscale Modeling and Simulation Lab., CNRS, Créteil, France), and Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr)

The femoral stem primary stability achieved by the impaction of a beam called "ancillary" attached to the implant during its insertion is an important factor of success in cementless surgery. However, surgeons rely on their proprioception, making the process subjective. The use of experimental modal analysis (EMA) without sensor nor probe fixation on the implant or on the bone is a promising non-destructive approach to determine implant stability. This study investigates whether EMA performed directly on a beam, which is temporary fixed to the implant during its insertion, could provide information on the boundary condition, which corresponds to the implant stability. A cementless femoral stem was inserted into 10 bone phantoms of human femurs and EMA was carried out on the ancillary using a dedicated impact hammer for each insertion step. Two bending modes were identified in the frequency range 400–8000 Hz for which the resonance frequency was shown to be sensitive to the insertion step and to the bone-implant interface properties. A significant correlation was obtained between the two modal frequencies and the implant insertion depth ($R^2 = 0.95 \pm 0.04$ and $R^2 = 0.94 \pm 0.06$). This study opens new paths towards the development of noninvasive vibration-based evaluation methods to monitor cementless implant insertion.

10:35

1aSA5. Vibrations of a thin, prestressed, fluid-loaded spherical shell forced by a point sound source. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

This paper investigates vibrations of and sound scattering by a thin elastic spherical shell when static pressures are different in the fluids inside and outside the shell. The work is motivated by stratospheric balloons being employed to carry acoustic sensors and by the proposed use of large, encapsulated gas bubbles for passive suppression of underwater sound [O. A. Godin and A. B. Baynes, J. Acoust. Soc. Am., **143**, EL67–EL73 (2018)]. Linearized equations of motion of thin, prestressed shells are derived from first principles. Differences in the fluid-loading terms from previously proposed *ad hoc* models are identified and their significance is analyzed. Analytic solutions are derived for vibrations of a spherical shell excited by an incident spherical sound wave and for acoustic fields in the internal and external fluids. The results are verified against exact solutions in several particular cases. The mathematical model of the shell vibrations is applied to characterize the influence of the shell's material properties on passive suppression of underwater noise and to quantify the effect of wave scattering by the balloon on performance of balloon-borne infrasound sensors in the atmosphere. [Work supported by ONR.]

10:50

1aSA6. Laser-acoustic detection of objects buried underwater. Guoqin Liu (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38677, gliu2@go.olemiss.edu), Vyacheslav Aranchuk (National Ctr. for Physical Acoust., Univ. of MS, University, MS), Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, MS), and Craig J. Hickey (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

An object buried underwater such as landmines can potentially be excited to vibrate by a sound source in air. The vibration then radiates a secondary wave in the water to excite the water surface vibration which can be detected by a laser sensor. This idea of laser-acoustic detection of buried objects is effective in detecting objects buried under ground where the object is mechanically excited and the ground surface vibration is scanned by the laser sensor. When applying this approach to detect objects buried underwater, the addition of the water layer has an impact on the flexibility of the detection. Numerical simulations and laboratory experiments are conducted to assess this flexibility. Vibrations of the object and the water surface are experimentally measured and numerically simulated for water layers of different depths. The results reveal the impact of the water layer and the effectiveness of the detection. [Work supported by the Office of Naval Research under Award No. N00014-21-1-2247.]

Session 1aSP

Signal Processing in Acoustics, Biomedical Acoustics, Architectural Acoustics, Physical Acoustics, and Underwater Acoustics: Acoustical Imaging, Reconstruction, and Localization I: General Applications I (Hybrid Session)

Yangfan Liu, Cochair
Ray W. Herrick Laboratories, Purdue Univ., 177, South Russell Street, West Lafayette, IN 47907-2099

Efren Fernandez Grande, Cochair
Technical University of Denmark, Lyngby DK 2800, Denmark

Chair's Introduction—8:30

Contributed Paper

8:35

1aSP1. Approximate extraction of late-time returns via morphological component analysis. Geoff Goehle (Penn State Univ., 225 Sci. Park Rd., State College, PA 16803, goehle@psu.edu), Benjamin Cowen, Thomas E. Blanford, J. Daniel Park, and Daniel C. Brown (Penn State Univ., State College, PA)

A fundamental challenge in acoustic data processing is to separate a measured time series into relevant phenomenological components. In the setting of sensing elastic objects using active sonar, we wish to separate the early-time returns (e.g., returns from the object's exterior geometry) from late-time returns caused by elastic or compressional wave coupling. Under the framework of morphological component analysis (MCA), we compare two separation models using the short-duration and long-duration responses as a proxy for early-time and late-time returns. Results are computed for broadside geometries using Stanton's elastic cylinder model as well as experimental data taken from an in-air circular synthetic aperture sonar (AirSAS) system, whose separated time series are formed into imagery. We find that MCA can be used to separate early and late-time responses in both the analytic and experimental cases without the use of time-gating. The separation process is demonstrated to be robust to noise and compatible with AirSAS image reconstruction. The best separation results are obtained with a flexible, but computationally intensive, frame based signal model, while a faster Fourier Transform based method is shown to have competitive performance.

Invited Papers

8:50

1aSP2. Spherical-sector harmonics domain processing for wideband source localization using spherical-sector array of directional microphones. Chibuzo J. Nnonyelu (Dept. of Electron. Design/Sensible Things that Communicate Res. Ctr. (STC), Mid Sweden Univ., Holmgatan 10, S241c, Sundsvall 85170, Sweden, chibuzojoseph.nnonyelu@miun.se), Meng Jiang, and Jan Lundgren (Dept. of Electron. Design/Sensible Things that Communicate Res. Ctr. (STC), Mid Sweden Univ., Sundsvall, Sweden)

The spherical microphone array can be uneconomical for applications where the sources arrive only from a known section of the sphere. For this reason, the spherical-sector harmonics was developed for processing spherical sector array. The orthonormal spherical sector harmonics (SSH) basis functions which accounts for the discontinuity arising from sectioning the sphere have been developed and shown to work for the array of omnidirectional microphones. In this work, the SSH basis functions are applied to far-field wideband sound source localization using spherical-sector array of first-order directional microphones (cardioid microphones). The array manifold interpolation method is used to produce the steered covariance matrix and the MUSIC algorithm applied for the direction of arrival estimation. The root-mean-square error performance of this spherical-sector array of the first-order cardioid microphones is compared against that of the omnidirectional microphones for different directions and signal-to-noise ratio.

9:10

1aSP3. Sensor array design for sparse sound field reconstruction. Yang Shen (Inst. of Sound and Vib. Res. of Hefei Univ. of Technol., 193 Tunxi Rd., Hefei 230009, China, 2019010012@mail.hfut.edu.cn), Chuan-Xing Bi, Xiao-Zheng Zhang, Yong-Bin Zhang (Inst. of Sound and Vib. Res. of Hefei Univ. of Technol., Hefei, China), and Rong Zhou (School of Mech. Eng. of Hefei Univ. of Technol., Hefei, China)

The compressive-equivalent source method can reconstruct the sound field of the sparsely distributed sound sources through promoting the sparsity of the equivalent source strengths. To reduce the mutual coherence of the sensing matrix, the spatially random sampling is performed. However, different samplings would make the reconstruction results instable, and a fixed sampling would not guarantee satisfied reconstruction results when reconstructing the sound field at different frequencies. To handle this issue, the present paper proposes a sensor array design method for sparse sound field reconstruction. After obtaining the matrix by the sum of the frequency-dependent sensing matrices, the optimization problem of the proposed method is formulated based on the mutual coherence minimization principle, and the solutions indicating the selected positions of the sensors are solved by the gradient projection. Numerical simulation and experimental results show that lower reconstruction errors can be obtained with the designed senor array compared to the spatially random sampling.

9:30

1aSP4. Frequency domain method for holographic localization of rotating sound sources. Jianxiong Feng (Purdue Univ., 1736 N 9th St. #30, Lafayette, IN 47904, jxfeng2016@gmail.com), Yangfan Liu, and Kai Ming Li (Purdue Univ., West Lafayette, IN)

Most of the previous studies employed beamforming techniques for the localization of rotating sources. These methods usually rely on the use of a virtual rotating array (VRA), in which the Doppler effects caused by source rotation can be eliminated. The VRA signal generation can be achieved in both frequency and time domains. In this presentation, compressive sensing (CS) near-field acoustic holography (NAH) has been extended to the reconstruction of rotating sources using the frequency domain (FD) VRA method. This combination can provide an improved resolution of spatial reconstructions at low frequencies compared to beamforming methods. In addition to the direct combination of the FD-VRA method and CS acoustical holography techniques, a more efficient algorithm for FD-VRA was developed to address the shortcomings of the traditional approach, such as higher computational cost and the requirement of a circular array with equally spaced microphones. The proposed FD-VRA method only involves a direct summation of the rotating sidebands of signals measured on a stationary array, which effectively reduces the computational load. Furthermore, an arbitrary array of microphones can be used. It will be shown that the proposed algorithm works for narrowband sources without overlapping frequency components in the rotating sidebands.

9:50

1aSP5. Advanced applications of the continuous-scan acoustic measurement method. John McShane (ATA Eng., Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128, john.mcshane@ata-e.com), Parthiv Shah, Peter Kerrian, Michael Yang, and Andrew White (ATA Eng., Inc., San Diego, CA)

This presentation will cover recent progress in the development of continuous-scan acoustic measurement (CSAM) methods for imaging, reconstruction, and localization of spatially complex sound fields. CSAM comprises an acoustic array of slowly moving and fixed sensors coupled with position tracking to enable high-spatial-resolution partial field measurements for visualization and source characterization. In previous work, ATA has developed and demonstrated the use of a rotating array for beamforming and acoustical holography. This presentation will focus on one or more advanced applications of CSAM. Examples may include (1) quantifying spatial coherence from sound fields generated by multiple uncorrelated sources, (2) combining multiple test runs to visualize larger 2D spatial apertures and/or 3D sound fields, (3) combining high-resolution measurements with vibroacoustic models to predict scattered sound fields, and (4) exploring alternate array configurations (e.g., a spherical-surface measurement aperture).

10:10–10:25 Break

10:25

1aSP6. A hybrid compressive sensing approach to noise source visualization: Application to a diesel engine. Tongyang Shi (Inst. of Acoust. Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, shitongyang@mail.ioa.ac.cn), J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN), and Frank Eberhardt (Cummins, Inc., Columbus, IN)

To identify sound source locations by using Fourier-based Near-field Acoustical Holography (NAH), a large number of microphone measurements is generally required to span the source region and ensure a sufficiently high spatial sampling rate. As a result, such measurements are costly, a fact which has discouraged the industrial application of NAH to identify sound source locations. However, recently, compressive sensing approaches have made it feasible to identify concentrated sound sources with a limited number of microphone measurements. In the present work, sound radiation from the front face of a diesel engine was measured by using one set of measurements from a 35-channel combo-array. The locations of significant noise sources were then identified by using three compressive sensing algorithms: Wideband Acoustical Holography (WBH), l_1-norm minimization, and a hybrid approach which combined WBH and l_1-norm minimization. The latter approach takes advantage of the l_1- norm's ability to locate spatially distinct sources, and WBH's ability to suppress "ghost" sources. It was found that the hybrid algorithm can localize and visualize the major noise sources over a broad range of frequencies, even though using a relatively small number of microphones. Finally, comments are made regarding sound field reconstruction differences between the algorithms.

Contributed Papers

10:45

1aSP7. An equivalent source method for holographic visualization of acoustic sources in an apertured enclosure based on exterior measurements. Reza Ramezanpour (Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, rramezan@purdue.edu) and Yangfan Liu (Purdue Univ., West Lafayette, IN)

Most acoustical holography techniques are developed to visualize sound sources or reconstruct sound field in a free-space environment, where no boundary reflected or scattered sound are present. However, in some engineering applications, a sound scatter is an inseparable part of source mechanism. For example, for a centrifugal fan, the aerodynamic noise sources are the turbulent pressure inside the fan shell, but the source distribution is significantly altered if the fan shell is removed. Thus, in order to visualize the sources inside the fan shell by holography methods based on exterior sound measurements, the measurements need to be performed with the presence of the shell (which is a sound scatter). This obviously violates the free-space assumption for most acoustical holography methods. In this work, traditional free-space equivalent source methods are extended to visualize acoustic sources situated in an enclosure with openings based on sound measurements exterior to the enclosure. The key step is to replace the free-space Green's function used in traditional equivalent source methods by a Green's function that satisfies the boundary condition of the apertured enclosure. Simulations are performed to prove the concept of the proposed acoustical holography method.

11:00

1aSP8. Improving the super resolution focusing above a two-dimensional Helmholtz resonator array using particle velocity decomposition of time reversal focusing. Andrew Basham (Dept. of Phys. and Astronomy, Brigham Young Univ. - Provo, 1305 N Canyon Rd., Apt 304, Provo, UT 84604, andrewtbasham@gmail.com), Adam D. Kingsley, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ. - Provo, Provo, UT)

Super resolution is possible using resonators placed in the near field of time reversal focusing. This presentation addresses the problem that a two-dimensional Helmholtz resonator array in a three-dimensional reverberant environment has limited ability to produce a high-resolution spatial focus in the time reversal focusing of audible sound. Acoustic waves propagating out-of-plane with the resonator array are not as strongly modified by the resonators, partially negating the effect of the resonators. The use of a two-dimensional waveguide limits the out-of-plane propagation, leading to improved resolution. It is also shown that post processing using an orthogonal particle velocity decomposition of a spatial scan of the focusing can limit out-of-plane particle motion in the near field of the array, which bypasses the effect of the unwanted third spatial dimension of propagation. Each of these techniques results in a pressure field focus reconstruction that has a higher spatial resolution.

11:15

1aSP9. A double zero MVDR beamformer that is universal over number of second-order notches. 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)

Adaptive beamformers, like the minimum variance distortionless response (MVDR) beamformer, suffer from interferer direction mismatch due to interferer motion. This mismatch reduces interferer suppression and degrades the beamformer's performance. Applying a flatter and broader notch near the interferer location effectively suppresses moving interferers. The double zero MVDR (DZ MVDR) adaptive beamformer [Tuladhar & Buck, JASA, 2015] produces second-order notches in the interferer's directions by squaring the array polynomial of a subarray SMI MVDR beamformer. The performance of the DZ MVDR beamformer depends on the chosen number of second-order notches by applying different subarray aperture sizes. If the number of interferers in a dynamic environment is known, then the ideal number of second-order notches can be implemented. In practice, the number of interferers is an unknown and often time-varying parameter. To address this challenge, we propose a universal DZ (UDZ) MVDR beamformer that is universal over the number of second-order notches. This universal beamformer requires fewer degrees of freedom than competing beamformers to suppress the interferers, thus providing a better white noise gain in the array output. We present simulation results illustrating the benefits of the UDZ MVDR beamformer in the presence of multiple moving interferers. [Work supported by ONR 321US.]

11:30

1aSP10. Difference-frequency MUSIC for beamforming. 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 Jeung-Hoon Lee (Changwon National Univ., Changwon, South Korea)

A beamforming method is derived to estimate direction-of-arrivals (DOAs) for high-frequency sources. The algorithm uses multiple frequencies and their difference frequencies (DFs). Due to the DFs below the aliasing frequency, the method mitigates spatial aliasing. DF-processing has been applied previously to the conventional beamforming [Abadi *et al*., J. Acoust. Soc. Am., **132**(5):3018–3029, 2012]. The proposed method uses a classical beamformer, MUSIC, which is a subspace method. The MUSIC shows sharp peaks in the beamforming spectrum and high-resolution. The DF-MUSIC integrates data samples into a sample covariance matrix and estimates DOAs. We propose three ways to treat multiple time and DF samples. The time-DF-MUSIC obtains MUSIC spectra at each DF and averages the spectra over time to get final DOAs. The frequency-DF-MUSIC obtains MUSIC spectra at each time snapshot and averages the spectra over frequency to get final DOAs. The time-frequency-DF-MUSIC considers all time and frequency jointly and improves the DOA estimation.

11:45

1aSP11. Acoustic source localization using Bayesian optimization with a Gaussian process surrogate model. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wjenkins@ucsd.edu), Peter Gerstoft, and Yongsung Park (Univ. of California, San Diego, La Jolla, CA)

Matched field processing is a computationally expensive approach to estimate acoustic source location, since it relies on a grid search through range and depth. We present an efficient alternative that samples the parameter space with a Bayesian approach using a Gaussian process as a surrogate model of the objective function. The objective function is defined as a Bartlett processor whose output measures the match between a received and replica pressure field on a vertical line array. Replica fields are obtained using a normal mode propagation model whose geoacoustic parameters are selected from the parameter search space. The surrogate model represents the posterior on the objective function and is updated with each model evaluation. Optimization is performed with sequential model evaluations, with an acquisition function guiding the next point in parameter space to be evaluated. We demonstrate our approach using both simulations and real data collected during the SWELLEX96 ocean acoustics experiment. Results indicate Bayesian optimization using the GP surrogate model converges on average within 700 evaluations of the objective function, far fewer than required in matched field processing.

Session 1aUW

Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics: Exploring Fine-Grained Sediments in the Variable Ocean I (Hybrid Session)

David P. Knobles, Cochair
Physics, Knobles Scientific and Analysis, 5416 Tortuga Trail, Austin, TX 78731

Tracianne B. Neilsen, Cochair
Physics and Astronomy, Brigham Young University, N251 ESC, Provo, UT 84602

Preston S. Wilson, Cochair
Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 East Dean Keeton Street, Mail Stop: C2200, Austin, TX 78712-0292

Chair's Introduction—10:00

Invited Papers

10:05

1aUW1. Estimation of bottom attenuation in shallow water with changing depth using intensity fluctuations due to moving nonlinear internal waves. Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), Valery Grigorev (Voronezh Univ., Voronezh, Russian Federation), and Yanyu Jiang (Marine Geosciences, Univ. of Haifa, Haifa, Israel)

The use of analysis of intensity fluctuations in the presence of non-linear internal waves (NIWs) for estimating the bottom attenuation is demonstrated in (JASA, v.140, p.3980, 2016) using data of Shallow Water 2006. Key objects of this analysis are the time dependence and spectrum of the total intensity (summed over the entire depth of the waveguide, in other words, over all hydrophones of the vertical line array—VLA). In this paper, we consider data on the measurement of field fluctuations (frequency 224 Hz) on an acoustic track $\sim$ 30 km long in the presence of NVWs, obtained in the ASIAEX 2001 experiment. The acoustic track consists of two parts: "deep," ($\sim$ 270 m) and "shallow" ($\sim$ 120 m). This feature leads, first, to variation of NIW parameters, propagating toward the coast, and second, it gives reason to assume different bottom properties for these sub-tracks. The paper proposes a technique for estimating the attenuation coefficients independently for two parts based on the study of spectrograms at the VLA and spectrum of total intensity, and the following fitting between experimental data and theoretical modeling. The obtained attenuation coefficients are compared with the data of other authors for the same area.

10:25

1aUW2. Effect of oceanography on broadband acoustic wave propagation in a range dependent shallow water environment. Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Rm. 140, Evans Hall Rm. 140, Newark, DE 19716, badiey@udel.edu), Jhon A. Castro-Correa, Christian D. Escobar-Amado, Lin Wan (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), and William Hodgkiss (Scripps Inst. of Oceanogr., San Diego, CA)

To investigate the effects of oceanographic variability on broadband acoustic wave propagation in shallow water environments, two experiments were carried out on the New England continental shelf region in 2017 and 2022, respectively. Towed sources were used to transmit chirp waveforms over the frequency ranges of 0.5—1.25 kHz and 1.5 – 4.0 kHz on the same acoustic tracks. The broadband acoustic transmissions were received by vertical acoustic arrays with multiple hydrophones. In 2017, the sound speed in the water column was isovelocity while in 2022 the presence of a duct at the depth of 20 m and upward refracting conditions reduced the interaction of acoustic transmissions with the seafloor compared to the experiment in 2017. Experimental observations show the importance of the water column on bottom interacting broadband waveforms causing challenges for the inversion algorithms. In this paper, we present acoustic and oceanographic results showing how the variability of sound speed could change the characteristics of the broadband transmission. Modeled results versus experimental data are discussed. [Work supported by ONR.]

Contributed Papers

10:45

1aUW3. Estimating seabed properties with a residual convolutional network. Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Michael C. Mortenson (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Amanda Schemel (Embry-Riddle Aeronautical Univ., Prescott, AZ), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), and William Hodgkiss (Scrips Inst. of Oceanogr., San Diego, CA)

Deep learning can assist in characterizing seabeds using sources of opportunity such as shipping noise. While previous work focused on seabed classification, this study uses a residual convolutional neural network to find individual seabed properties. The training data were labeled with sound speed, density, attenuation, and thickness of the layer values of the top sediment layer. A comparison was made between predictive capabilities of ResNet-18 networks when trained to learn a single parameter and those trained to simultaneously learn multiple parameters. For stiff parameters—those with high information content in the data—learning an individual parameter performed better. These single parameter predictions are fundamentally different from a geoacoustic inversion for one parameter. In geoacoustic inversion, all other parameters are held at a fixed value. In deep learning, variability in all other parameters is contained in the training data, but the network focuses on features in the data related to a single property. The trained networks are applied to ship noise measured during the 2017 Seabed Characterization Experiment. [Work supported by the Office of Naval Research and the National Science Foundation's REU program.]

11:00

1aUW4. Maximum entropy approach for sediment porosity and source strength of transiting ships. Jacob R. Nuttall (Phys. And Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, jnutt1@byu.edu), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

Remote sensing using passive sonar in the ocean is a challenging problem due to variations in the geoacoustic structure of the seabed and unknown source location and strength. One way to improve remote sensing is to perform an optimization for geoacoustic and source parameters. We use a Bayesian maximum entropy (BME) approach with a viscous-grain shearing parameterization for two sediment layers. The statistical optimization provides probability distributions for porosity and thickness of the sediment layers as well as ship speed, closest point of approach, and the source strength for the Wales-Heitmeyer empirical source level spectrum. We use this approach on spectrograms of transiting ships collected on a vertical line array during the 2017 seabed characterization experiment. We compare the resulting parameter distributions from distinct ships as well as previous estimates of geoacoustic values and source properties. This research shows that the BME approach obtains estimates for porosity and source strength that have narrow posterior probability distributions. [Work supported by the Office of Naval Research.]

Invited Papers

11:15

1aUW5. Joint trans-dimensional inversion for seabed and water-column properties in the seabed characterization experiments. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca) and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

This paper considers the information content to resolve both a geoacoustic model of the seabed and a model of the water-column sound-speed profile (SSP) in the inversion of modal-dispersion data. A Bayesian formulation of the inverse problem allows prior information representing varying levels of independent knowledge to be applied separately to the seabed and water-column models. Issues of interest include the extent to which knowledge (or lack thereof) of the SSP affects geoacoustic inversion, and the ability to remotely estimate the SSP from acoustic data in cases where the seabed is either well known or poorly known. The joint inversion for seabed and water-column properties is formulated in terms of a separate trans-dimensional model for each, providing an automated approach to quantitative model selection. The seabed model is formulated in terms of an unknown number of uniform sediment layers, while the SSP is formulated as an unknown number of depth/sound-speed nodes, with the reciprocal of sound speed squared varying linearly between nodes, as per normal-mode propagation models. The approach is applied to modal-dispersion data measured during the seabed characterization experiments at the New England Mud Patch. [Work supported by the Office of Naval Research]

11:35

1aUW6. Potential and kinetic energy of underwater noise measured at steep grazing angles over contrasting seafloors. Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98115-7834, dahl@apl.washington.edu) and David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA)

Observed near the seabed, broadband noise emissions from a source or vessel passing directly above the sensor exhibit frequency bands where potential acoustic energy is greater than kinetic energy while the opposite occurs in neighboring frequency bands. The condition where the dynamic and kinematic energy forms differ in this manner is characteristic to interference involving steep angles or near-normal incidence reflection from the seafloor. Recent experiments conducted on the New England Mud Patch (NEMP) and the neighboring (towards offshore) shelf break waters demonstrate the sensitivity to seabed structure, which evolves from the soft mud layers at the NEMP into the harder sediments that are exposed on the continental slope. Results are expressed as a ratio of kinetic to potential energy as a function of frequency, with resulting data series yielding information on seabed properties. Variance components as function of frequency are also isolated based on a singular value decomposition of a matrix constructed from the data series (known as singular spectrum analysis). The components are associated with different sub- bottom layering features. A model for kinetic and potential energy combined with candidate geoacoustic models is used to interpret both the data series and components.

Session 1pAA

Architectural Acoustics, Musical Acoustics, and Noise: Evaluation of Completed Performance Spaces: Goals and Methods

Jonah Sacks, Cochair
Acentech, 33 Moulton Street, Cambridge, MA 02138

Robin Glosemeyer Petrone, Cochair
Threshold Acoustics, 141 W Jackson Blvd., Suite 2080, Chicago, IL 60604

Invited Papers

1:00

1pAA1. Orchestra listening conditions—Some reflections on reflections. Paul H. Scarbrough (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, pscarbrough@akustiks.com) and Christopher Blair (Akustiks, LLC, Norwalk, CT)

Over many years, the authors have worked with orchestra musicians and conductors in efforts to improve their hearing conditions onstage. Experience gained from these efforts has led to some conclusions that vary strongly from common remedial assumptions, particularly with regard to early reflected energy. Anecdotal examples from several of our projects will be presented which indicate that careful reduction of early energy onstage may offer substantial improvements in musician (and audience) responses. The rooms discussed will include experiments conducted in the Sala São Paulo, Indianapolis' Hilbert Circle Theater, the Sala Minas Gerais, and the Ordway Concert Hall. Model studies that were conducted for the recently completed David Geffen Hall at Lincoln Center revealed interesting differences between predicted onstage EDT and that of the audience chamber, suggesting that a shorter EDT in the platform zone may be beneficial and produce greater satisfaction among musicians.

1:25

1pAA2. Round and round we go: Speech intelligibility and localization for in-the-round spaces. Eric Magloire (Charcoalblue, 330 7th Ave., Ste. 2002, New York, NY 10001, eric.magloire@charcoalblue.com), Bruno Cardenas (Charcoalblue, New York, NY), and Byron Harrison (Charcoalblue, London, United Kingdom)

The directional characteristics of the human voice are a key theatrical effect in spoken word drama. Maintaining intelligibility and believable localization as performers move and reorient on the stage is a formidable design challenge for in-the-round spaces. We provide an overview of the design strategies for this theatrical form using some notable precedents. Referencing Chicago's newest in-the-round theatre at Steppenwolf Theatre, we present the acoustic goals and the architecturally integrated design for speech acoustics. The presentation includes the key design parameters and the traditional and non-traditional intelligibility metrics used to evaluate the design options and the completed construction, along with subjective impressions of artists and audiences related to speech intelligibility and localization.

1:50

1pAA3. Using 3-dimensional rendering programs to visualize 2-dimensional impulse responses of the built environment. Robin Glosemeyer Petrone (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, robin@thresholdacoustics.com)

As the first notes from a performer are expressed on a newly constructed stage, we acousticians close our eyes and hold our breath. We do this not because our reputation is on the line but rather to focus all our attention on our auditory system. We listen for the impulse response, hearing the loudness, clarity, bass ratio, and profile of the decay. We open our eyes as the performer continues, pairing our visual and auditory senses to gain still more information. We look for those surfaces placed around the stage to support the direct sound. We follow the vibrations into the larger volume and map their path as they bounce around the room and rain down a shower of energy that envelops us in warm sonic embrace. Did we achieve the acoustic signature we designed? We will discuss our methods for mapping out the recording of two-dimensional time and energy in an impulse responses onto three-dimensional space with rendering software in order to analyze and tune our performances spaces.

2:15

1pAA4. The Acentech acoustical checkout: Measuring and listening to completed performance spaces. Kelsey Rogers (Acentech, 33 Moulton St., Cambridge, MA 02138, krogers@acentech.com), Jonah Sacks, and Khaleela Zaman (Acentech, Cambridge, MA)

Visiting a completed performance space is an invaluable learning opportunity. Our practice at Acentech has evolved to include approximately equal parts measurement and listening. We report relevant findings to the owner and design team; but often, much of the information that we collect is for our own internal research and development. This presentation will summarize our process for evaluating a completed performance space, including measurement, visualization, and analysis of spatial impulse responses, listening to and recording speech and music in the room, and listening to calibrated convolutions of measured spatial impulse responses with anechoic audio. Ideas for future analysis tools and related research efforts will also be discussed.

2:40–2:55 Break

Contributed Papers

2:55

1pAA5. Comparing measured sound strength to theory as a function of reverberation time and room volume. Joshua R. Dunham (Tenor Eng. Group, 811 1st Ave., Ste. 535, Seattle, WA 98104, joshua.d@tenor-eng.com)

ISO Standard 23591 demonstrates how the sound pressure level in a room may be estimated with a knowledge of (1) the sound power of sources in that room and (2) the sound strength G of that room as defined in ISO 3382-1. In this presentation, the theory connecting room volume V, reverberation time $T60$, and sound strength G is summarized, and a comparison of measured sound strength G to this theory is explored for a wide range of rooms. After an explanation of the mathematical model, a description of the calibration of measurement equipment using Odeon software is discussed, including measurements taken in a reverberation chamber. A brief review of the rooms in which sound strength was measured is provided. Both measured and theoretical sound strength results are presented for the rooms reviewed. The difference between measured G and theoretical G using this mathematical model is analyzed statistically both for data measured with this test setup and with concert hall data from Beranek's book. This model lends itself to designing rooms of appropriate reverberance and loudness for a given context, musical or otherwise.

3:10

1pAA6. Szilard-Wigner distributions completely define physics and perceptions of concert halls. James B. Lee (none, 6016 S. E. Mitchell St., Portland, OR 97206, cadwal@macforcego.com)

At the second joint meeting of the Acoustical Society of Japan and the Acoustical Society of America in 1988 Yamasaki and Itoh presented time-frequency measurements of several concert halls employing Fourier transforms on autocorrelations, the same device Szilard and Wigner had developed for quantum theory in the 1920s. In 1995, it was proved that resonant scattering in the body of good halls generates power spectra resolved in time and frequency, correctly presenting musical information emanating from the orchestra. In poor halls specular reflections from hard flat surfaces generate echoes encoding architectural information about size, shape, features of the hall, necessarily detracting from desired musical information. Yamasaki and Itoh also demonstrated that "reverberation-time" tells essentially nothing about properties, physical or perceptual, of concert halls; impulse responses, from which Szilard-Wigner distributions are computed, tell much more. Careful study of these matters explains why after the era of Wallace Sabine, when "reverberation-time" became the sine qua non of concert hall theory, design of fine concert halls became impossible.

3:25–3:55
Panel Discussion

Session 1pAB

Animal Bioacoustics, Acoustical Oceanography, Signal Processing in Acoustics, Underwater Acoustics, and Noise: Climate Change and Sound: How the Sound of the Planet Reflects the Health of the Planet II (Hybrid Session)

Laura Kloepper, Cochair
Department of Biological Sciences, University of New Hampshire, 230 Spaulding Hall, Durham, NH 03824

Megan Ballard, Cochair
University of Texas at Austin, 10000 Burnet Road, Austin, TX 78665

Edward J. Walsh, Cochair
VA Loma Linda Healthcare System, Loma Linda, CA 92357

Invited Papers

1:00

1pAB1. Coral reef soundscapes and noise. Sophie Nedelec (Biosciences, Univ. of Exeter, 38 Poltimore Dr., Pinhoe, Pinhoe, Exeter, Devon EX1 3DY, United Kingdom, s.nedelec@exeter.ac.uk), Andrew Radford (Univ. of Bristol, Bristol, United Kingdom), Peter Gatenby (James Cook Univ., Townsville, Queensland, Australia), Isla Keesje Davidson (Univ. of Bristol, Bristol, United Kingdom), Laura Velasquez Jimenez (James Cook Univ., Townsville, Queensland, Australia), Maggie Travis (Univ. of Puget Sound, Tacoma, WA), Katy Chapman, Kieran McCloskey (Biosciences, Univ. of Exeter, Exeter, United Kingdom), Timothy Lamont (Lancaster Univ., Lancaster, United Kingdom), Bjorn Illing (Thunen Inst. of Fisheries Ecology, Bremerhaven, Germany), Mark McCormick (Univ. of Waikato, Waikato, New Zealand), and Stephen Simpson (Univ. of Bristol, Bristol, United Kingdom)

Building back coral reefs requires limiting greenhouse gas emissions, limiting local threats, and active restoration. Vessel noise pollution is a widespread threat acting at a local level impacting a broad range of species from cetaceans to cephalopods. Ultimate consequences of noise pollution include death due to injury or predation, failure to develop and reduced offspring quality and survival. We tested the hypothesis that protecting coral reefs from motorboat noise can improve reproductive output in fish on the Great Barrier Reef in Australia. Across an entire breeding season, we limited motorboat traffic at three separate reefs with breeding spiny chromis. We compared these protected nests with nests on three further reefs where motorboats drove regularly nearby. We replicated the breeding experiment in a laboratory setting to isolate noise as the stimulus. Protecting coral reefs from traffic noise was beneficial for breeding fish and their offspring: egg fanning and offpsring growth were enhanced with lower energetic resource use, and reproductive success was enhanced. Limiting traffic noise at the local level presents a valuable opportunity for enhancing resilience in coral reefs by improving reproductive success in fish. This could speed up recovery following climate related mass mortality events caused by bleaching and cyclones.

1:20

1pAB2. Acoustic propagation and ambient noise in a thawing Arctic. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

Over the past 50 years, the extent and volume of sea ice coverage in the Atlantic Ocean has been receding at a nearly linear rate due to anthropogenic induced climate change. It is expected that some time within the next 10 to 30 years, there will be an ice free summer. For periods beyond this time, the Arctic Ocean will change from mostly covered by multi-year ice to mostly covered by first year ice. In this talk, the impact of the changes in ice morphology (roughness and thickness) and ice extent will be examined from both an under-ice propagation modeling and an ambient noise modeling perspective. The first year ice is expected to have less loss, leading to better sound propagation. The reduced ice coverage will lead to more open water exposed surfaces for wind generated noise, and more open water for ice-free propagaiton of shipping induced anthropogenic sound. Both of these features will drive up the background level of sound in the Arctic Ocean. Regional differences in ice cover and expected wind and shipping noise will be addressed.

1:40

1pAB3. Climate change drives frog call change in Puerto Rico: Predictions and implications. Peter M. Narins (Integrative Biology & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095, pnarins@ucla.edu)

Acoustic signal production and reception in ectotherms are acutely temperature-dependent. We recorded the advertisement calls of the Puerto Rican coqui frog (*Eleutherodactylus coqui*) along an altitudinal gradient and repeated the measurements twenty-three years later. We found that over this period, at any given elevation, calls exhibited both significant shortening of their duration and increases in

pitch. We also found that the observed differences are consistent with a movement to higher elevations for the population, a familiar response to rising ambient temperature. Using temperature data obtained from four local weather stations over the same time period, we confirmed a significant ambient temperature increase, the magnitude of which closely predicts the observed changes in the frogs' calls. Physiological responses to long-term temperature rises include reduction in individual body size and thus, population biomass. These changes can have alarming consequences, since coqui frogs form an essential component of the food web in the Puerto Rican rainforest. (Work supported by UCLA Academic Senate via Grant No. 3501.)

Contributed Papers

2:00

1pAB4. Ocean acidification: Future effects to ocean environments and biologic response. Grant Eastland (Test and Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

Ocean acidification has become one of many consequences of our modern civilization that potentially could cause major changes to Earth's undersea environments. There are several physical and chemical effects from acidification, one of which affects the undersea acoustic environment. There are various empirical models for acoustic absorption, but an empirical model by Francois and Garrison has a pH dependent contribution. As the ocean becomes more acidic, the pH decreases from its current value and the model predicts a reduction in the amount of acoustic absorption. The result of this absorption reduction changes the overall transmission loss of sound propagation and makes the underwater environment a "louder" place. The consequences of which would be minor complications to ocean exploration but will have a greater impact upon the undersea fauna. As the environment becomes louder, those animals that use acoustics to echolocate food, will find it more difficult to do so and others may find the environment to not be amenable to their survival. These changes are occurring on a time scale on the order of decades, so environmental acoustic monitoring of ambient conditions and large-scale migration can be utilized as an indicator of the changes to the water space.

2:15–2:30 Break

2:30

1pAB5. Blue whales track oceanographic variability across spatial and temporal scales. William Oestreich (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, woestreich.research@gmail.com), Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst. (MBARI), Moss Landing, CA), and John P. Ryan (Res., MBARI, Moss Landing, CA)

Animals' ability to track variable resources in space and time is critical to their survival in dynamic and changing ecosystems. Understanding populations' behavioral flexibility in response to natural and anthropogenic ecosystem variation requires long-term and detailed measurements of both animal behavior and ecosystem properties. In a series of recent studies, we leverage blue whales' widely-propagating songs to understand their capacity to track ecosystem variability across episodic foraging and seasonal-to-interannual migration scales. By integrating individual and population-level study of singing blue whales, we first identify an acoustic signature of their population-level transition from foraging to migration. Applying this acoustic signature to a six-year study period, we find that blue whales flexibly time their transition to migration to track interannual variability in the phenology of their foraging habitat. Within the foraging season, we further track blue whale behavior via a directional acoustic vector sensor. Using this approach, we find that blue whales maximize access to aggregated prey patches by dynamically tracking fine-scale wind-driven upwelling plumes in space and time. Combined, these findings display blue whales' ability to track oceanographic variability across spatial and temporal scales and suggest mechanisms by which these predators individually and collectively sense their dynamic foraging habitat.

2:45

1pAB6. Vertical migration of the ocean's deep scattering layer affects the health of the planet. Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst. (MBARI), 7700 Sandholdt Rd., Moss Landing, CA 95003, kbb@mbari.org)

The vertical transport of carbon, termed the biological pump, mitigates the effects of climate change by sequestering 25% of the carbon dioxide emitted by fossil fuel use each year. The nightly vertical movement of millions of zooplankton, fishes, squid, and shrimp that make up the deep scattering layer may account for as much as half of this carbon sequestration but this process remains poorly parameterized. We use a stationary echosounder deployed at $\sim$1000 m depth complemented by video imaging and environmental DNA sampling to describe the vertical migration of mesopelagic micronekton in Monterey Bay, California from the darkness of the deep sea to the bounty of food at the surface over 30 months. Vertical migration at this site showed as many patterns as have been described globally, with drastic changes in the proportion of animals migrating, the extent of migration, and the species involved. Patterns were consistent only over periods of a few days even as mobile sampling showed synchrony in patterns throughout the mesopelagic zone of the bay. These results provide critical information for incorporating the effects of this largest migration on Earth on predictions of future climate.

3:00

1pAB7. Effective medium modeling of acoustic propagation in a seagrass meadow. Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, 5705 Gloucester Ln., Unit A, Austin, TX 78723, thomas.jerome@arlut.utexas.edu), Megan Ballard, Kevin M. Lee, Colby W. Cushing (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kyle Capistrant-Fossa (Marine Sci. Inst., The Univ. of Texas at Austin, Port Aransas, 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 Kenneth H. Dunton (Marine Sci. Inst., The Univ. of Texas at Austin, Port Aransas, TX)

Seagrasses are foundation species in many coastal ecosystems, but these environments are declining globally due to climate change and other anthropogenic impacts. Ballard *et al*. [J. Acoust. Soc. Am. **147**, 2020] established the efficacy of acoustic remote sensing techniques for seagrass monitoring by exploiting acoustic sensitivity to gas bubbles produced by photosynthesis and gas channels within the seagrass leaves. However, the effects of seagrass on acoustic propagation are not understood with sufficient quantitative detail, and an improved model describing propagation through a mixture of seagrass leaves, free gas bubbles, and seawater is needed to aid in integrating acoustic methods into conservation efforts. This talk provides an overview of developments in the modeling of acoustic propagation through a *Thalassia testudinum* meadow using a homogeneous effective medium approach to represent the seagrass leaves and seawater. The model accounts for the complex microstructure of seagrass leaves including the encapsulated gas channels and the elastic properties of the seagrass tissue. The model is intended for use in geoacoustic inference algorithms for the overall goal of providing estimates of seagrass productivity and biomass. Candidate effective medium models include 2D cylindrical seagrass leaves and a micromechanical model of a seagrass leaf cross-section. [Work supported by NSF.]

3:15–3:35
Panel Discussion

Session 1pAO

Acoustical Oceanography: Topics in Acoustical Oceanography

Miad Al Mursaline, Cochair
Mechanical Engineering/ Applied Ocean Physics & Engineering, Massachusetts Institute of Technology/Woods Hole Oceanographic Institution, 70 Pacific Street, Cambridge, MA 02139

Brian T. Hefner, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

1:00

1pAO1. Impacts of seabed characteristics on short-range acoustic propagation in seismic surveys. Alexander S. Douglass (Oceanogr., Univ. of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu), Warren T. Wood, Benjamin J. Phrampus (Ocean Sci. Div., Naval Res. Labs., Hancock County, MS), and Shima Abadi (Oceanogr., Univ. of Washington, Seattle, WA)

An abundance of marine seismic reflection surveys with publicly available datasets have been collected over the last several decades to study the composition of the seabed up to multiple kilometers below the sea floor. In these surveys, a loud impulsive airgun source broadcasts sound into the seabed and the reflections from the seafloor and seabed layers are measured on a long (up to ∼15 km) towed hydrophone array. In these surveys, both the shallow seabed layers (several hundred meters below the seafloor) and deeper layers (multiple kilometers below the seafloor) can significantly impact the acoustic field in the water column at close ranges. Experimental data have shown instances of sound exposure levels (SEL) at ranges between ∼8 and 15 km that exceeded levels predicted by cylindrical spreading models by nearly 15 dB. This work aims to use the abundance of seismic data available to explore the relationship between seabed and sub-seabed characteristics and the acoustic field in the water column. Inversion results that provide seabed sound speed profiles and reflection characteristics will be used to understand the seabed characteristics and its potential effects on SELs or sound pressure levels evaluated along the towed array. [Work supported by ONR.]

1:15

1pAO2. Towards a cloud optimized data lake for archived water column sonar data. Carrie Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, 325 Broadway, Boulder, CO 80305, carrie.bell@colorado.edu), Rudy Klucik, Chris Slater, Charles Anderson, and Veronica Martinez (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, Boulder, CO)

Due to their value to the ocean science and fisheries management communities, NOAA National Centers for Environmental Information (NCEI), with NOAA Fisheries and University of Colorado Cooperative Institute for Research in Environmental Sciences, established a national archive for water column sonar data. There are currently 210 TB of data freely and publicly available, and that volume is growing rapidly as sonar technology advances. The spatially and temporally diverse archive is accessible through its dedicated data portal and Amazon Web Services. Throughout 2023, we will develop a cloud-optimized data lake of echosounder files representing a ∼100 TB subset of the archive holdings. The echosounder files will be translated from their complex, binary and proprietary file format into zarr files following the Earth Science Information Partners analysis-ready cloud-optimized standards. The resulting data lake will serve as the foundation for building analytical capabilities that can cost-effectively tap into the archive's sonar holdings, especially when coupled with compute power. The zarr stores will subsequently feed into EchoFish, the archive's AWS-hosted interactive data visualization platform to facilitate subsetting and prevent the data lake from becoming a data swamp. The progress and potential applications of this NOAA Center for Artificial Intelligence funded project will be presented.

1:30

1pAO3. Material properties and broadband backscatter measurements of individual euphausiids from the California coastal ecosystem. Joseph Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy., Southampton, NY 11968, joe.warren@stonybrook.edu) and Brandyn M. Lucca (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY)

During the 14–24 May 2019 leg of the NOAA juvenile rockfish survey, zooplankton and small nekton were sampled off the coast of Northern California using a midwater trawl. This talk will focus on three euphausiid species (*E. pacifica, T. spinifera, N. difficilis*) which were present in most catches. Krill morphology (size and shape) and animal density relative to seawater were measured for 175 live (or very recently expired) individuals. Roughly 150 individual krill were frozen and brought back to land for mass measurements. Soundspeed relative to seawater was measured for 15 mixed-species aggregations of krill. Broadband (38–73 and 130–210 kHz) and narrowband (38, 50, 70, 120, 150, and 200 kHz) backscatter measurements were made on 32 live (or very recently expired) individual krill. These empirical measurements were then compared to predictive backscatter measurements using scattering model parameters measured for those individuals (when available). Differences between predicted and measured backscatter were generally small, although there were individual krill where measurements at some frequencies varied significantly (i.e. more than 3 dB). Understanding the variability in scattering model inputs among and within these species, and the accuracy of scattering model predictions provides context (i.e. uncertainty estimates) for acoustic measures (or indices) of krill biomass.

1:45

1pAO4. Characterizing the deep sound scattering layer with a moored upward-looking broadband split-beam echosounder. Zhaozhong Zhuang (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS 16, Woods Hole, MA 02543, zhuangz@mit.edu) and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The deep scattering layer (DSL) is a ubiquitous feature of the global ocean. It consists of a large community of mesopelagic organisms which links the marine food web and has recently garnered much interest from

commercial fisheries. Such biological communities are inherently coupled with oceanic physical processes such as mesoscale eddies, internal waves and boundary currents. However, very little is known about the physical consequences and biological responses in this coupled system. To address these questions, an upward-looking broadband split-beam echosounder (36 kHz – 44 kHz) was moored at ∼580 m depth (bottom depth ∼2790 m) in the New England slope waters in July 2021. The time series analysis focuses on (1) volume scattering strength and spectrum shape and (2) target strength, layer density and compositions, and individual behavior in three dimensions. We demonstrate the seasonal variability in the biomass, distribution, and vertical migration patterns of the deep scattering layer, and quantify the temporal and spatial variability of the DSL under the influence of multiple oceanic physical processes, including warm core rings, semi-diurnal internal tides, and near-inertial waves.

2:00

1pAO5. Broadband attenuation-based measurements of bubble size distributions. Rachel E. Kahn (MIT-WHOI Joint Program, 86 Water St. MS 12, Woods Hole, MA 02543, rkahn@whoi.edu), Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA), Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, Seattle, NJ), and Robert Petitt (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

Oceanic bubbles generated by breaking waves are a key mechanism for air-sea gas exchange. Dense bubble plumes can also interfere with acoustic propagation in shallow water due to their high scattering and attenuation. Most studies of oceanic bubbles have used narrow-band acoustical and optical techniques to quantify bubble size distributions under breaking waves in open water. To study bubble distributions at estuarine tidal fronts, where localized downwelling and advection can generate large bubbles plumes, we developed a measurement system for estimating bubble size distributions via broadband excess attenuation. The system consists of two transducers (3–30 kHz and 30–110 kHz) and two hydrophones mounted on a towable frame. Testing was performed in a laboratory wave tank to measure bubble distributions below a field-scale (1 + meter) breaking wave. This presentation will discuss the bubble size distributions observed at different locations along the breaking wave crest in the direction of propagation and their associated time scales and sound speed anomalies. We also compare these distributions with those previously observed in open water and will present preliminary results from a field deployment of the system in the Connecticut River estuary.

2:15

1pAO6. Overview of distributed acoustic sensing technology and recently acquired data sets. Alexander S. Douglass (Oceanogr., Univ. of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu), John Ragland (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA), and Shima Abadi (Oceanogr., Univ. of Washington, Seattle, WA)

Fiber optic distributed acoustic sensing (DAS) is a recent innovation utilized primarily in the seismic community for measuring seismic acoustics signals at low frequencies (single digit Hz and below). The technique utilizes strain rates in a fiber optic cable, observed via the backscatter of light pulses, to measure the acoustic field. Recently, the capabilities of this technology to measure higher frequency acoustic fields (10s to 100s of Hz) have been explored. Low frequency marine mammals calls at ∼20 Hz and ship noises have been successfully recorded, and a recent experiment demonstrated the capability to record up to ∼500 Hz. This talk provides an overview of DAS technology and introduces two recent experiments for studying water column acoustics with DAS. A 4-day experiment conducted in November 2020 as part of the Ocean Observatories Initiative (OOI) provides data along two fiber optic cables extending west from the coast of Oregon by 65 km and 95 km, reaching depths of 590 m and 1575 m, respectively. DASCAL22, a recent experiment from October 2022, simultaneously recorded data using DAS at 2 kHz sampling rate on a cable extending 3.54 km at ∼100 m depth and multiple moored hydrophones placed close to the DAS cable, allowing direct comparison between a new and existing technology.

2:30

1pAO7. Comparing distributed acoustic sensing data with hydrophone recordings. Shima Abadi (Oceanogr., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, abadi@uw.edu), Alexander S. Douglass (Oceanogr., Univ. of Washington, Seattle, WA), and John Ragland (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA)

Distributed acoustic sensing (DAS) is a technology that transforms telecommunication fiber optic cables into dense sensor arrays by continuously transmitting pulses of light down the cable and measuring backscattering from natural inhomogeneities within the fiber cable. The technology can densely sample the acoustic field over long ranges (up to 100 km), providing a means for large scale passive acoustic monitoring. To evaluate the capabilities of DAS, it is necessary to benchmark and calibrate the technology relative to traditional hydrophone data. The DAS Calibration Experiment 2022 (DASCAL22) recorded 9 days of both DAS and hydrophone data in Puget Sound, WA in October 2022. The DAS data were recorded with a sample rate of 2 kHz, and the cable extended 3.5 km on the seafloor between two islands, reaching depths of 100 m, and the hydrophones were moored adjacent to the DAS cable at 5 m and 25 m from the seafloor. The recordings include impulses from an active source at 1 m, 5 m, and 10 m depths, and an abundance of passive acoustic data corresponding to ship traffic, wind, and rain. This work aims to draw comparisons between the hydrophone and DAS recordings to evaluate the capability of DAS at detecting sounds at frequencies as high as 1 kHz.

2:45

1pAO8. Using distributed acoustic sensing for ocean ambient sound analysis. John Ragland (Elec. and Comput. Eng., Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu), Alexander S. Douglass, and Shima Abadi (Oceanogr., Univ. of Washington, Seattle, WA)

Distributed acoustic sensing (DAS) is a technique that utilizes the back scattering in fiber optic cables to densely sample the strain rate in both space and time. This technique has been widely demonstrated as a powerful tool for seismic sensing, but the efficacy of submerged, under-sea cables for ocean acoustic sensing remains underexplored. The ocean observatories initiative (OOI) conducted a distributed acoustic sensing experiment in November of 2021, where two of the fiber optic cables continuously recorded the strain rate for four days. In this talk, the ambient sound field recorded by the OOI DAS experiment will be explored. A statistical comparison of hydrophone measurements and DAS measurements will be presented. Additionally, the possibility of using ocean ambient sound techniques, such as ambient noise interferometry will be explored and compared to hydrophone analysis. [Work supported by ONR.]

3:00

1pAO9. Optical detection of ensonified capillary-gravity waves using polarimetric imaging. Kaustubha Raghukumar (Integral Consulting, Inc., 200 Washington St., Ste. 201, Santa Cruz, CA 95060, kraghukumar@integral-corp.com), Lindsay Hogan, Christopher Zappa (Lamont Doherty Earth Observatory, Columbia Univ., Palisades, NY), Frank Spada, and Grace Chang (Integral Consulting, Inc., Santa Cruz, CA)

The optical detection of surface capillary-gravity waves induced by underwater sound has many potential applications that range from the detection of sound-generating underwater objects to airborne bathymetric surveys. While multiple lab-based efforts have measured acoustically generated surface capillary-gravity waves, we report on a recent field-based measurement using polarimetric imaging. A controlled acoustic source was placed 10 m below a lake surface and emitted sound in the 500 Hz to 10000 Hz frequency range. The lake surface was imaged using a polarimetric camera mounted 7 m above the lake surface. Measurable short-lived surface capillary-gravity waves (∼3 mm wavelength) were observed in the polarimetric camera images during ensonification of the lake surface. Changes were observed in both the omnidirectional and directional wave spectra. In the omni-directional wavenumber spectrum, enhanced capillary wave activity at high wavenumbers was observed for acoustic source frequencies in the 2–5 kHz range. Additionally, ensonification was observed to result in the amplitude and wavenumber modulation (enhancement/diminution) of existing wind-generated surface gravity-capillary waves. In the directional spectra, while ambient gravity-capillary waves showed a spreading function with stronger downwind versus upwind propagation, the acoustically generated gravity-capillary waves showed minimal impact on the directionality of the wave spectra.

3:15

1pAO10. Correlating shipping noise on multiple beams. Paul Hursky (Appl. Ocean Sci., 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

There has been much interest in using transiting surface ships as sources of opportunity for tomography and geo acoustic inversion. In tomography, it would be valuable to increase the ranges at which ship signatures can be processed by measuring times of arrival at the output of beamforming processes, which provide spatial processing gain. This additional gain may enable fainter multipath arrivals to be identified and exploited as additional paths to sample the ocean water column. This seems readily achievable when cross-correlating multiple vertical beams formed on a single vertical line array for example. However, cross-correlating beams from multiple arrays, vertical or horizontal, raises questions. In both cases, different beams may have different Doppler, which is manageable with a correlation process that includes Doppler compensation. But obeserving a ship from different vantage points may be problematic, if the ship signature is due to horizontally displaced noise sources distributed around the ship. We will present results of processing surface ships observed on single and multiple arrays and assess the use of such processing in ambient noise tomography.

MONDAY AFTERNOON, 8 MAY 2023 BELMONT, 1:00 P.M. TO 3:15 P.M.

Session 1pBAa

Biomedical Acoustics: Ultrasound Brain and Super-Resolution Imaging II

Chengzhi Shi, Chair
GWW School of Mechanical Engineering, Georgia Institute of Technology, 771 Ferst Dr NW, Atlanta, GA 30332-0001

Invited Papers

1:00

1pBAa1. Vortex ultrasound catheter for cerebral intravenous sonothrombolysis. Xiaoning JIang (NC State, 911 Oval Dr., Raleigh, NC 27695, xjiang5@ncsu.edu), Bohua Zhang (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Huaiyu Wu (NC State, Raleigh, NC), Raul Nogueira (Univ. of Pittsburgh, Pittsburgh, PA), and Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

One treatment challenge for cerebral venous sinus thrombosis (CVST) is to achieve recanalization within a short period of time (e.g., 30 min) for significant clinical outcomes. Ultrasound-mediated thrombolysis (sonothrombolysis) presents a promising treatment for venous embolism. In this paper, a miniature vortex ultrasound transducer with frequency of 1.8 MHz was developed for sonothrombolysis. A composite transducer with 2-by-2 sub-aperture piezoelectric elements was assembled into a 9-French catheter to generate a physical helical wavefront. The prototyped vortex ultrasound catheter was characterized by measuring the acoustic pressure amplitude and phases, followed by *in-vitro-* sonothrombolysis tests. It was found that a vortex ultrasound field can be successfully generated by the prototype. The vortex lytic rate was increased by more than 50% compared with the nonvortex lysis with the same acoustic power input. A long ($\sim$7.5 cm), completely occluded *in-vitro-* 3D model of acute CVST was fully recanalized within 8 min. The unprecedented sonothrombolysis rate was likely attributed to the vortex ultrasound-induced shear stress, which can effectively disrupt acute blood clots. Furthermore, no vessel wall damage over *ex-vivo* bovine veins was found after the vortex sonothrombolysis treatment.

1:20

1pBAa2. Microring resonator based disposable ultrasound-sensing chronic cranial window. Cheng Sun (Mech. Eng., Northwestern Univ., 2145 Sheridan Rd. B392, Northwestern University, Evanston, IL 60208-3111, c-sun@northwestern.edu)

Chronic cranial window (CCW) has been widely used to provide optical access to the brain cortex for longitudinal imaging, while preserving the physiological environments of the brain. Various optical imaging modalities have been demonstrated with CCW for longitudinal brain imaging, including single-photon and multi-photon fluorescence microscopy, optical coherence tomography, and photoacoustic microscopy (PAM). However, sophisticated surgeries and windows designs are still required for chronic imaging and direct cellular recording or manipulation because the surgical implantation of CCW precludes direct physical access to the brain other than optical access. Here, we report an active CCW with integrated ultrasound sensor based on a transparent microring resonator (MRR). The MRR is fabricated on a quartz substrate by using low-cost soft nanoimprinting lithography (sNIL) and then integrated on a glass window with optical fibers for disposable ultrasound-sensing CCW (usCCW). Encapsulating MRR inside an acoustic impedance matched protection layer further improves its reliability for *in vivo* applications over the observed period of a month. The functions of the active usCCW are experimentally validated through longitudinal (PAM) imaging of cortical vasculature in live mice over a 28-days period.

Contributed Papers

1:40

1pBAa3. Development of binder-jetting based skull phantoms for transcranial ultrasound research. Kazi Safowan Shahed (Penn State Univ., 227 Johnson Terrace, State College, PA 16803-2917, kxs6079@psu.edu), Hyeonu Heo (Penn State Univ., University Park, PA), Guha Manogharan (Penn State Univ., State College, PA), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Ultrasound imaging can be used in time sensitive and dynamic environments like trauma care due to its advantages, such as real-time, affordable, portable, noninvasive, and nonionizing. Despite these advantages, transcranial ultrasound is hindered for brain imaging by the severe phase aberration of the human skull. To study phase aberration, it is crucial to have a fundamental understanding of the relationship between the macro-/micro-structure of the skull and the speed of sound distribution in the skull. This is proven to be challenging with *ex vivo* skulls, because the properties of these skulls are not controllable. To address this issue, we used binder jetting (BJT) 3D printing, high fidelity additive manufacturing, to print skull phantoms using calcium hydroxyapatite (HA) powder because HA is one of the major constituent materials of the human bone. After printing phantoms, they were sintered to get to the final form. The phantoms were validated via ultrasound measurements.

1:55

1pBAa4. A model-based image reconstruction algorithm for near real-time transcranial photoacoustic imaging. Hyungjoo Park (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hyungjoo4408@gmail.com) and Yun Jing (Graduate Program in Acoust., Penn State Univ., State College, PA)

Photoacoustic imaging is a promising hybrid imaging modality that can visualize both microscopic and macroscopic structures and oxygenation changes in brain function. Phase aberration caused by the skull is a major barrier for high-quality photoacoustic imaging of the brain. Time-reversal methods have been used to address this issue, though they rely on solving the full-wave equation, and is therefore computationally heavy and slow. Herein, a near real-time model-based image reconstruction algorithm is proposed. The acoustic forward model is mathematically described as a model matrix and image reconstructions are performed by employing the pseudoinverse of the model matrix and calculating the initial pressure distribution. The pseudoinverse matrix only needs to be computed once for the same experimental setup and acoustic medium and is obtained offline prior to imaging. The proposed algorithm shows equivalent image quality but considerably faster reconstruction time (>40 times faster), compared to the time-reversal method and therefore could potentially enable near real-time photoacoustic imaging.

2:10–2:25 Break

2:25

1pBAa5. Acoustic plane wave detection of intracranial cavitation in a polyacrylamide human head model under blunt impact. Eric J. Galindo (Chemical Eng., New Mexico Inst. of Mining and Technol., PO Box 2404, 801 Leroy Pl, Socorro, NM 87801, Eric.Galindo@student.nmt.edu) and Michaelann Tartis (Chemical Eng., New Mexico Inst. of Mining and Technol., Socorro, NM)

Traumatic brain injuries (TBIs) occur in everyday life in the form of blunt impacts, while military personnel experience a combination of blast and blunt impacts from explosives. Damage mechanisms in TBIs, including cavitation, are poorly understood due to the inability to optically monitor brain deformation inside the skull during collisions. To overcome these limitations, high frame rate acoustic plane wave imaging in conjunction with high-speed optical imaging were combined to visualize intracranial cavitation during impact of a transparent brain phantom. The full-scale phantom was composed of 7% and 10% (w/v) polyacrylamide (PAA) and enclosed in a 3D printed skull with acrylic windows for optical image acquisition. This surrogate utilizes simplified geometry to represent key anatomical features such as the gyri, sulci, and ventricles while possessing rheological properties similar to the human brain. Optical and acoustic data correlate cavitation to high-contrast regions. Acoustic spectra show expected harmonic and broadband behavior during cavitation bubble growth and collapse. Further processing with superresolution techniques may enhance cavitation localization providing a stand-alone acoustic tool applicable in various TBI scenarios allowing observations at depth, where optical techniques are limited.

2:40

1pBAa6. Fast spectral approach for delay correction in heterogeneous media. Scott J. Schoen (Radiology, Harvard Med. School and Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, sschoenjr@mgh.harvard.edu) and Anthony E. Samir (Radiology, Harvard Med. School and Massachusetts General Hospital, Boston, MA)

Ultrasound imaging makes fundamental assumptions about the interrogated medium, including the propagation speed of the waves. For many clinical applications, assuming a constant speed of sound (SoS, typically 1540 m/s) is sufficiently accurate to yield diagnostically useful images. However, when the medium is appreciably heterogeneous (e.g., trans-skull, or through deep layers of subcutaneous fat), this approximation fails and the distortion of the wavefronts yields a degraded image (i.e., there is aberration). Correcting these distortions for a known SoS distribution is typically intensive computationally. Here, we demonstrate the utility of the heterogeneous angular spectrum method (HASM), a spectral method developed for passive cavitation imaging, to recover efficiently the appropriate apodization and delays in transmit and receive toward corrected images. Compared with a straight ray computation (i.e., no refraction) with mean 5% SoS variation, HASM was roughly tenfold faster, and computed delays matched to within (-84 ± 89) ns [compared with (-660 ± 312) ns error for 1540 m/s assumption]. Furthermore, transmit corrections for focused or plane-wave sequences may be obtained similarly with the method for a specified depth. HASM, thus, has promise toward real-time image enhancement and improved diagnostic utility. [Work Supported by the ASA FV Hunt Postdoctoral Fellowship.]

2:55–3:15
Panel Discussion

Session 1pBAb

Biomedical Acoustics: Ultrasound for Ocular Therapy

Maxime Lafond, Chair
LabTAU - INSERM U1032, 151, Cours Albert Thomas, Lyon, 69424, France

Chair's Introduction—1:00

Invited Paper

1:05

1pBAb1. Ultrasound enhanced ocular drug delivery. Vesna Zderic (The George Washington Univ., 800 22nd St. NW, Washington, DC 20015, zderic@gwu.edu)

Our objective has been to determine ultrasound parameters that can provide optimal delivery of different drugs into the eye via transcorneal and transscleral routes, study mechanisms of ultrasound action, and determine long-term safety of this approach. We showed previously that exposing cornea to therapeutic ultrasound can lead to up to 10 times more delivery of a drug-mimicking compound into the eye, with only minimal alterations in the corneal structure. Subsequently, we continued to work on drug delivery problems with clinical relevance, such as promoting delivery of antibiotics and steroids for treatment of eye inflammations. Our studies also included modeling of temperature increases in the eye during ultrasound application, effectiveness and safety of delivery of an anti-parasitic drug PHMB into the eye, and delivery of macromolecules such as Avastin via transscleral route for treatment of mascular degeneration. Our research work showed that ultrasound application can be effective and safe for delivery of drugs of different molecular sizes into the eye *in vitro* and *in vivo*. This work may eventually lead to development of an inexpensive, and non-invasive ultrasound method that can be applied in an outpatient clinic to allow targeted delivery of medications for treatment of different ocular diseases.

Contributed Papers

1:25

1pBAb2. Therapeutic ultrasound for enhanced transcorneal macromolecule delivery. Claire M. Allison (Dept. of Biomedical Eng., The George Washington Univ., Newport Beach, CA), Annette Jimenez-Benoit (Dept. of Biomedical Eng., The George Washington Univ., Washington, DC), Krishna Ramajayam (Dept. of Pediatrics, Medical Univ. of South Carolina, Washington, DC), Dieter Haemmerich (Dept. of Pediatrics, Medical Univ. of South Carolina, 800 22nd St. NW, Washington, DC 20052, haemmer@musc.edu), and Vesna Zderic (Dept. of Biomedical Eng., The George Washington Univ., Washington, DC)

Previous experiments demonstrated that ultrasound exposure at 400-600 kHz can increase transcorneal drug delivery of sodium-fluorescein. Our study aims to determine if these same methods can enhance the delivery of fluorescently labeled FITC-dextran macromolecules of similar molecular weights to clinically relevant drugs. Dissected corneas of adult rabbits were placed in a diffusion cell between a donor compartment filled with a solution of FITC-dextran macromolecules diluted with phosphate-buffered saline (PBS) to 1 mg/ml and a receiver compartment filled with PBS. Each cornea was exposed to the drug solution for 60 minutes, with the experimental group receiving 5 min of continuous ultrasound or 10 min of pulsed ultrasound at 50% duty cycle at the beginning of treatment. Unfocused circular ultrasound transducers were operated at 0.5 –1 W/cm^2 intensity and at 600 kHz frequency. Macromolecule delivery was quantified by the fluorescence intensity detected in the receiver compartment. The greatest increase in transcorneal drug delivery seen was 1.2 times ($p < 0.05$) with the application of pulsed ultrasound at 0.5 W/cm^2 and 600 kHz for 10 min with 40 kDa macromolecules. Gross observation of corneas after experiments demonstrated no significant damage. Ongoing microscopy and thermal-modeling studies aim to characterize any ultrasound-induced damage.

1:40

1pBAb3. Preliminary investigations on cavitation effects in the crystalline lens. Maxime Lafond (LabTAU - INSERM U1032, 151, Cours Albert Thomas, Lyon 69424, France, maxime.lafond@inserm.fr), Alice Ganeau (LabTAU - INSERM U1032, Lyon, France), Olfa Ben Moussa, Frédéric Mascarelli, Gilles Thuret (BiiO Lab., Jean Monnet Univ., Saint-Etienne, France), Stefan Catheline, and Cyril Lafon (LabTAU - INSERM U1032, Lyon, France)

Presbyopia is the age-related stiffening of the crystalline lens, reducing near vision. One of the main mechanisms of this lens stiffening is the formation of disulfide bonds. We propose ultrasonic cavitation to interact with the lens structure, disrupt disulfide bonds and restore flexibility. Here, we present early results on the feasibility of the technique. Cavitation was nucleated inside the porcine lenses using confocal transducers and signals emitted in the focal area were acquired using an imaging array and reconstructed using frequency-domain passive cavitation imaging. Peak negative pressure of 23 MPa for 20-μs pulses at 1.1 MHz provided a clear chance of nucleating cavitation with our system. Exposures were performed during 10 seconds of cavitation at a PRF of 250 Hz. Cavitation clouds were clearly visible in B-mode images for both 6-months old and 3-years old porcine eyes. To provide initial safety data, the absence of cataract was confirmed on transparency images. Histology revealed no damage to the lens structure. A gel phantom with a disulfide bond-dependent hardening is proposed to investigate the potential for cavitation to disrupt the disulfide bonds.

1:55

1pBAb4. Shear wave elastography (SWE) for the measurement of lens elasticity in the context of monitoring a presbyopia treatment by ultrasonic cavitation. Alice Ganeau (LabTAU - INSERM U1032, Lyon, France, alice.ganeau@inserm.fr), François LEGRAND, Gabrielle Laloy-Borgna, Cyril Lafon, Maxime Lafond, and Stefan Catheline (LabTAU - INSERM U1032, Lyon, France)

Presbyopia is a progressive loss of accommodation capacity due to a rigidification of the lens. In order to prove the efficiency of ultrasonic cavitation to soften the lens, curvilinear SWE has been developed. It is a method to measure stiffness based on the tracking of surface waves on the outer layer of lens. Experimental models have been implemented: flat phantoms and beads made of 10% and 15% of porcine gelatin and finally porcine lenses (young: 6 months and old: 1 to 6 years). A mechanical vibrator created vibrations (0.1–3.5kHz) and an ultrafast ultrasound imaging system was used to track the propagating waves. From the curvature of the frequency domain of the cross-correlated displacement field, the wavelength and velocity were computed. Experimental models enable to validated the method with dispersion curves substantially different between both concentrations. Results in lenses are similar: a dispersion curve highly dependent of the frequency, reflecting the complexity of elastic wave propagating in a multilayered medium surrounded by a membrane. To retrieve quantitative elasticity data, advanced numerical models have to be developed. This method will be applied on lenses treated by cavitation to assess its potential for treating presbyopia.

2:10

1pBAb5. Monitoring presbyopia treatment with ultrasonic cavitation using shear wave elastography (SWE): Experiments and simulations. François Legrand (LabTAU - INSERM U1032, 151 Cr Albert Thomas, Lyon 69003, France, francois.legrand@inserm.fr), Alice Ganeau, Gabrielle Laloy-Borgna, Cyril Lafon, Maxime Lafond, and Stefan Catheline (LabTAU - INSERM U1032, Lyon, France)

This numerical study aims at evaluating quantitative monitoring of elasticity changes induced by ultrasonic cavitation using shear waves elastography (SWE). SWE is a good method to safely evaluate the lens stiffness. Yet, the absence of backscattered signal inside the lens (regardless of the imaging technique) forbids elastography inside. Experimentally, we, thus, tracked surfaces waves on multiple models (gelatin beads, porcine lens) and evaluated dispersion curves of this waves. Finite difference time domain simulations are performed in order to evaluate elasticity with access to a few surface point, starting with a simple homogeneous inclusion towards complex inclusion closer to the lens configuration. In homogeneous inclusion, we noticed Scholte wave behavior on the dispersion. Complexifying the configuration by adding layers induced a gradient in the dispersion curves, the higher the frequency is the less the surface wave depends on the deepest layers. A similar behavior was observed experimentally: low frequencies with mostly resonances then a plateau matching Scholte wave expectation. Numerical studies indicate that the surface waves is sensitive to the shear elasticity inside the inclusion, yet this behavior is highly frequency dependent. This leads the way to investigate the frequency behavior of guided wave in experiments in lens in our team.

2:25

1pBAb6. Modulation of the blood-retina-barrier permeability by focused ultrasound: Computational and experimental approaches. Sam Bleker (Faculty of Eng. and Health, Univ. of Appl. Sci. and Arts, Von-Ossietzky-Str. 99, Göttingen 37085, Germany, sam.bleker1@hawk.de), Yuanlin Zhang (Molecular Biomarkers Nano-Imaging Lab., Brigham and Women’s Hospital, Boston, MA), John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Honolulu, HI), Ehsan Ranaei Pirmardan, Ali Hafezi-Moghadam (Molecular Biomarkers Nano-Imaging Lab., Brigham and Women’s Hospital, Boston, MA), and Christoph Russmann (Faculty of Eng. and Health, Univ. of Appl. Sci. and Arts, Göttingen, Germany)

The blood-retina-barrier (BRB) is a critical barrier that protects the retinal neurons. It selectively regulates the permeability of molecules, responds to pathologies, and modulates disease progression. While the BRB prevents harmful molecules from the blood from affecting the neurons and is thus essential for retinal health, it also poses a hurdle for therapeutic agents and impedes treatment. The goals are a non-invasive and well-defined transient increase of the BRB’s permeability by mechanical microscale-induced forces, to visualize the permeability change in real-time, and to computationally model the process for an improved understanding and control of the dynamics of larger molecules passage through the BRB. Our approach combines *in vivo* animal experiments with computational modeling in silico. The *in vivo* results are used to continuously and iteratively optimize the COMSOL model. In order to modulate BRB’s permeability, we developed a prototype acoustic lens assembly based on COMSOL simulation calculations for the delivery of ultrasonic pressure waves to targeted regions in the retina under optical guidance by light-based imaging. Our data indicate that lower energy focused ultrasound at a PRFP close to 0.8 MPa rapidly increases the permeability of the retinal vessels in a fashion that can be depicted in a real-time fluorescein angiography.

Invited Paper

2:40

1pBAb7. Ultrasound cycloplasty for refractory glaucoma treatment: The EyeTechCare Experience. Dietrich Wolf (EyeTechCare, EyeTechCare, Rillieux-la-Pape, France, DWOLF@eyetechcare.com), Maxime Lafond, and Cyril Lafon (LabTAU, Inserm U1032, Lyon, France)

In the 1980s, HIFU was successfully tested for treating refractory glaucoma (Lizzi *et al.*). A more recent device from EyeTechCare, made of six cylindrical transducers operating at 21MHz, can perform a conformal, partial, and fast thermal ablation of the ciliary bodies, a technique known as ultrasound cycloplasty (UCP). For the last 10 years, we investigated the safety and efficacy of UCP for reducing the production of aqueous humor and intraocular pressure (IOP). The primary outcomes that were validated (more than 55%) were therapeutic success (IOP reduction from baseline $\geq$20% and IOP $>$5 mm Hg without other surgical procedures) and vision-threatening complications. Secondary outcomes included mean IOP change from baseline at each follow-up visit, medication use, complications, and subsequent UCP and/or other postsurgical interventions. This technique was associated with a low complication rate and no cases of phthisis. In this presentation, we also highlight the remaining challenges in increasing the efficacy and reliability of UCP and its future developments. [Disclosure: Dietrich Wolf is CEO at EyeTechCare.]

Session 1pCA

Computational Acoustics, Structural Acoustics, and Vibration and Physical Acoustics: Computational Methods for Modeling Acoustic Damping

Shung H. Sung, Cochair
SHS Consulting, LLC, 4178 Drexel Dr, Troy, MI 48098

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center, U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290

Kuangcheng Wu, Cochair
Naval Surface Warfare Center - Carderock Division, 9500 MacArthur Blvd., West Bethesda, MD 20817

Invited Papers

1:00

1pCA1. Effects of material memory on acoustic absorption. Eric Abercrombie (Mech. Eng., Boston Univ., Dept. of Mech. Eng., Boston, MA 02215, abere@bu.edu) and James G. McDaniel (Mech. Eng., Boston Univ., Boston, MA)

Acoustic absorption may be expressed as a sum of energy dissipated in the fluid and energy dissipated in the material in contact with the fluid. This presentation explores the physical mechanisms of energy dissipation in the material, which is assumed here to be viscoelastic. The constitutive law of a viscoelastic material depends on the strain history of the material, thus giving the material a memory. The energy absorbed by the material is dependent on this strain history. Relationships between material memory and dissipation have been investigated [Harry H. Hilton and Sung Yi, Smart Materials and Structures, **1**(2), 113–122 (1992)], leading to 'viscoelastic designer materials' for vibration damping. Starting from this foundation, analysis and numerical simulations are used to investigate the relationships between material memory and acoustic absorption. This investigation establishes the influence of parameters in a generalized Maxwell model on the acoustic absorption. The results inform the creation of metamaterials with microstructures designed to produce specific memory characteristics. [Work supported by ONR.]

1:20

1pCA2. Radiation damping effect on the sound pressure response in a structural-acoustic cavity. Shung H. Sung (SHS Consulting, LLC, 4178 Drexel Dr., Troy, MI 48098, shsung1972@gmail.com) and Donald J. Nefske (DJN Consulting, LLC, Troy, MI)

A structural-acoustic system consisting of an enclosed cavity with flexible wall panels is considered to evaluate the effect of sound radiation-damping resulting from the panel vibration on the sound pressure level in the cavity. The interior cavity pressure can excite the flexible wall panels that will radiate noise to the exterior that in turn can reduce the interior noise due to the energy loss to the exterior surroundings. This energy loss is described as radiation damping of the interior cavity system. An enclosed rectangular cavity-plate system subjected to external random pressure excitation is used to evaluate the effect. The radiation damping is evaluated using a modal approach (FEM, BEM) to determine the radiation efficiency of the vibrating panels. The effect of the radiation damping for the example of a box-plate structure is evaluated using the CMA (classical modal analysis) method for the low-frequency range and the AMA (asymptotic modal analysis) method for the mid- and high-frequency ranges. The inclusion of radiation damping due to rigid body cavity mode excitation is shown to provide an improved prediction of the sound transmission loss especially in the low-frequency range.

1:40

1pCA3. Design and optimization of lightweight porous damping treatments. Yutong Xue (WanDong Medical, Midea Group, 177 S Russell St., West Lafayette, IN 47907, xyt@alumni.purdue.edu) and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Here, it is shown that properly-designed limp porous materials such as fibrous layers can provide damping equivalent to conventional viscoelastic dampers while providing advantages such as light weight and effective sound absorption. This then allows porous layers to be used as multi-functional noise and vibration control solutions in automotive and aerospace applications. It has also been found that the addition of bulk elasticity to the solid phase of the porous medium is beneficial since it improves damping performance compared to equivalent limp treatments. In this study, porous media, such as fibers and foams, were designed to serve as treatments for various vibrating structures to examine their damping effectiveness. Both analytical modeling and numerical simulation based on finite element methods were involved depending on the complexity of the structure. Specifically, a Fourier transform-based computational method was introduced as the key step to realize the accurate prediction of a panel's spatial response based on its wavenumber-frequency spectrum.

Then, parametric studies were conducted on a porous layer to identify the optimal bulk properties that would allow the layer to provide the largest possible damping within the target frequency region. Finally, design concepts for achieving the maximum damping potential of porous layers are summarized.

2:00

1pCA4. Topology optimization of wave-focusing waveguide acoustic black holes. Martin Berggren (Dept. of Computing Sci., Umeå Univ., Campustorget 5, Umeå 90187, Sweden, martin.berggren@cs.umu.se), Abbas Mousavi, Linus Hägg (Dept. of Computing Sci., Umeå Univ., Umeå, Sweden), and Eddie Wadbro (Dept. of Mathematics and Comput. Sci., Karlstad Univ., Karlstad, Sweden)

The acoustic black hole, as first proposed by Mironov for beams and plates, provides broadband wave focusing, exploitable for energy harvesting, for weak signal sensing, and as a strategy to minimize the amount of damping material needed for efficient, broadband damping. There have been a few suggestions aimed at providing an analogous device for acoustics in air. However, although these waveguide devices may work fine as broadband absorbers, their operational mechanism is decidedly different from their structural counterparts. Instead of providing wave focusing, all published devices seem to rely on the creation of resonances occurring increasingly closer to the waveguide entrance for increasing frequency. To explore a much larger class of possible designs than previously examined, we here apply a gradient-based material distribution topology optimization method. The optimization objective is a broadband maximization of power in a small region towards the end of the waveguide. We demonstrate that with this tool, it is indeed possible to design broadband true wave-focusing waveguides. These are geometrically much more complex than the previously proposed ones, a complexity that nevertheless efficiently can be handled by a recently developed gradient-based optimization method, able to accommodate also visco-thermal boundary-layer losses.

2:20

1pCA5. Sonic black hole duct. Kayla Petrover (70, Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., Bethesda, MD 20817, kayla.petrover@gmail.com)

The sonic black hole is a type of acoustic black hole that attenuates sound propagation via the acoustic black hole effect. This phenomenon is when an incident acoustic wave propagates through a power-law tapering geometry causing the wavelength to compress, the wave speed to decrease, and the time for the wave packet to reach the exit to extend infinitely. As a result, the wave never reaches the exit, so it cannot be reflected or transmitted. However, this outcome relies on the taper approaching exactly zero, which is impossible to manufacture. Therefore, the taper must be truncated leading to imperfect sound absorption. One method to increase the sound absorption is by applying perforations to the rings. This study explores the effect that varying the perforations on the sonic black hole's rings has on the absorption coefficient from 1 to 5 kHz.

Session 1pNS

Noise, Computational Acoustics, and Physical Acoustics: Rocket Noise Part II

Kent L. Gee, Cochair
Department of Physics and Astronomy, Brigham Young University, N281 ESC, Provo, UT 84602

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433

Contributed Papers

1:00

1pNS1. Initial observations and insights from a full-scale static rocket engine firing. Reese D. Rasband (Ball Aerosp., Provo, UT, r.rasband18@gmail.com) and Alan T. Wall (Sensory Systems, Air Force Res. Lab., Wright Patterson Airforce Base, OH)

With increased space travel focus and launch cadence, many commercial and public entities are looking to take advantage of this form of transit. One such use would be rapid delivery of supplies or troops across the globe. One question related to its viability is acoustic impacts on communities, structures, and personnel from both launch and landing maneuvers. In order to better predict these impacts, as well as improve measurement capabilities, AFRL recently measured a static firing of full-scale static rocket engine. This paper describes the measurement setup and initial results including directivity and decay. Relevance for future modeling and measurements is discussed.

1:15

1pNS2. Initial findings from Space Launch System liftoff measurements. Carson F. Cunningham (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, carsonfcunningham@gmail.com), Kent L. Gee, Grant W. Hart, Mark C. Anderson, Michael Bassett (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Jeffrey T. Durrant, Levi Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Whitney L. Coyle (Rollins College, Winter Park, FL), Makayle S. Kellison, and Margaret Kuffskie (Phys., Rollins College, Winter Park, FL)

This presentation documents initial findings from far-field noise measurements at NASA's Kennedy Space Center during liftoff of the Space Launch System's Artemis I mission, which occurred on November 16, 2022. The vehicle— the most powerful ever successfully launched into orbit—consists of four liquid-fueled RS-25 engines and two five-segment, solid-fuel rocket boosters (SRBs). Because this was the first launch, the noise radiation characteristics of this vehicle were previously unknown. Overall sound pressure levels, waveform characteristics, and spectra are described at distances ranging from 1.5 to 8.4 km. The levels due to the SRBs' ignition overpressure are particularly intense in the direction of the flame trench exit. The post-liftoff maximum one-third octave spectrum has a peak at 20 Hz, and maximum overall levels are greater than described in a pre-launch environmental assessment. These and other findings presently submitted to JASA Express Letters further understanding of super heavy-lift rocket acoustics.

1:30

1pNS3. Community-based noise measurements of the Artemis-I launch. Makayle S. Kellison (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, mkellison@rollins.edu), Whitney L. Coyle, Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL), Kent L. Gee, Grant W. Hart, Carson F. Cunningham, and Michael S. Bassett (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A group of students and professors from Rollins College, in collaboration with Brigham Young University's acoustics group, measured far-field noise from the Space Launch System's Artemis I mission liftoff. This presentation outlines the data collection process and measurements at locations 10 to 40 miles from Launch Complex 39B at Kennedy Space Center (KSC) in Cape Canaveral, Florida. Preliminary analysis of these measurements from outside the perimeter of KSC, including overall sound pressure levels, waveform characteristics, and frequency spectra, will be discussed.

1:45

1pNS4. Artemis I launch noise spectra: Effects of variable weather profiles over short- and long-range propagation paths. Whitney L. Coyle (Rollins College, 1000 Holt Ave - 2743, Winter Park, FL 32789, wcoyle@rollins.edu), Makayle S. Kellison (Rollins College, Winter Park, FL), Carson F. Cunningham, Kent L. Gee, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

NASA's Space Launch System (SLS) launched the Artemis I mission at 1:47 am local time on November 16, 2022. The timing of the night launch offered a multitude of interesting weather effects over the propagation paths to measurement locations ranging from 1.5 km to 60 km. This presentation discusses spectra at different distances from Launch Complex 39B, both inside and outside the Kennedy Space Center perimeter. Preliminary analysis of the SLS sound pressure levels and spectra suggests many propagation phenomena are at play. In addition to geometric spreading, atmospheric absorption, and nonlinear propagation, complex weather profiles and ground effects are discussed. Initial results of a single-frequency Greens-function parabolic equation model, with estimated weather profiles along the propagation paths, will be offered with approximate source and receiver height to compare levels at several locations.

2:00

1pNS5. Overview of BYU's acoustical measurements of the Falcon-9 SARah-1 launch, reentry sonic boom, and landing. J. T. Durrant (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, taggart.durrant@gmail.com), Mark C. Anderson, Michael S. Bassett (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Richard Batelaan (Phys. and Astronomy, Univ. of Nebraska-Lincoln, Lincoln, NE), David C. Lawrence (Culver Academies, Chicago, IL), Lucas K. Hall (California State Univ. Bakersfield, Bakersfield, CA), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

More rockets are launching than ever before, exposing structures, environments, and communities to intense sounds and vibrations. This increased launch cadence brings about a greater need to understand the noise radiated during these launches. This paper summarizes BYU's acoustical measurement and analysis of the Falcon 9 SARah-1 launch from Vandenberg Space Force Base in June 2022. In total, nine measurement stations were set up at locations between 400 m and 15 km from the launch pad. Additionally, this launch featured a booster landing back near the launch pad, resulting in a sonic boom about eight minutes after the launch. Each station successfully recorded the launch noise, sonic boom, and landing noise. Waveforms and spectra from the launch are discussed and compared across stations at different distances from the launch pad. The overall sound power level and other noise metrics are also discussed. [Work supported in part by NSF.]

2:15

1pNS6. Overall sound power levels from four launch vehicles. Richard Batelaan (Dept. of Phys. and Astronomy, Univ. of Nebraska-Lincoln, Lincoln, NE, richardhbatelaan@gmail.com), Mark C. Anderson (Phys. and Astronomy, Brigham Young Univ., Provo, UT), J. T. Durrant, Kent L. Gee, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

To better understand the radiated sound from launch vehicles, this paper describes an effort to quantify the overall sound power level (OAPWL) from different launches. The classic NASA SP-8072 report (K. Eldred, 1971) contains data showing OAPWL as a function of mechanical power but little has been published on actual OAPWL since. This paper uses noise data from four different vehicles launched from Vandenberg Space Force Base to examine the relationship between OAPWL and measurement distance from the launch pad. Because OAPWL values calculated from measurements lessen with increased distance, an empirical correction based on nozzle diameter is described. [Work supported in part by NSF.]

2:30

1pNS7. Assessing the influence of rocket launch and landing noise on threatened and endangered species at Vandenberg Space Force Base. Lucas K. Hall (Biology, California State Univ. BakersfieldSCI I, 2059001 Stockdale Hwy., Bakersfield, CA93311, lhall12@csub.edu), Megan R. McCullah-Boozer (CSU Bakersfield, Bakersfield, CA), Brooke M. Hinds (Biology, California State Univ. Bakersfield, Bakersfield, CA), Kent L. Gee, Grant W. Hart, Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), John P. LaBonte (ManTech SRS Technologies, Inc., Lompoc, CA), Darryl L. York, and Samantha O. Kaiseratt (30th Civil Engineer Squadron, US Space Force, Vandenberg SFB, CA)

Understanding how species of concern respond to anthropogenic activities is becoming increasingly important. While the ways in which anthropogenic activities affect species are many, our understanding the effects of anthropogenic noise on species is still developing. Vandenberg Space Force Base (VSFB) along the California Central Coast presents a unique opportunity to study the effects of anthropogenic noise on species of concern as (1) the rate of rocket launches/landings is predicted to significantly increase in the coming years compared to the last three decades and (2) there are multiple threatened and endangered species at VSFB that may be impacted by the increased launch cadence. Our interdisciplinary research team composed of wildlife biologists and acousticians will be studying the short and long-term behaviors of threatened and endangered shorebirds that nest relatively close to active VSFB launch complexes. Using previously collected and current bird counts and acoustic data from VSFB, models will be created that describe behavioral changes in species of concern. We will discuss the scope and overview of this project.

2:45

1pNS8. A comparative analysis of rocket noise recordings from wildlife acoustic monitoring devices. Megan R. McCullah-Boozer (CSU Bakersfield, SCI I, 205, 9001 Stockdale Hwy., Bakersfield, CA 93311, mmccullah@csub.edu), Brooke M. Hinds (CSU Bakersfield, Bakersfield, CA), Logan T. Mathews, Mark C. Anderson, Michael Bassett (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (CSU Bakersfield, Bakersfield, CA), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Remote acoustic recorders are used in wildlife passive acoustical monitoring (PAM) and species identification. Such devices are designed to be relatively low-cost, user-friendly, weather-robust, and operate for extended periods with low maintenance. Usually, the precision of these devices is limited because most applications do not require high-fidelity measurements for typical wildlife PAM applications. However, the relatively low cost and flexibility of deployment beg the question if they can accurately record other noise sources, such as rocket launches. Such a situation calls for an evaluation of accuracy and a complete understanding of their capabilities and limitations. As such documentation is limited, this study seeks to characterize the acoustic performance of two commercially available wildlife acoustic monitoring devices: the Wildlife Acoustics SM-4 and the Cornell Ornithology Laboratory SwiftOne. We present acoustical results from a laboratory experiment and a field measurement of an orbital rocket launch. Comparisons to industry standard devices are made in the time and frequency domains and show the spectral characteristics of these devices. Recommendations are made regarding the use of these devices based on their performance.

Session 1pPA

Physical Acoustics, Biomedical Acoustics, and Engineering Acoustics: Advances in Sonochemistry

James Kwan, Chair
Engineering Science, University of Oxford, Department of Engineering Science, University of Oxford, Parks Road, Oxford, OX1 3PJ, United Kingdom

Invited Paper

1:00

1pPA1. Ultrasonic cavitation processing of multi-functional materials. Iakovos Tzanakis (Oxford Brookes Univ., Wheatley Campus, Oxford OX3 0BP, United Kingdom, itzanakis@brookes.ac.uk), Abhinav Priyadarshi (Oxford Brookes Univ., Oxford, United Kingdom), Paul Prentice (Univ. of Glasgow, Glasgow, United Kingdom), Jiawei Mi (Univ. of Hull, Hull, United Kingdom), Koulis Pericleous (Univ. of Greenwich, London, United Kingdom), and Dmitry Eskin (Brunel Univ. London, London, United Kingdom)

Ultrasonic cavitation processing (USP) is a versatile technique that has gained a lot of momentum in the last decade as a sustainable, environmentally friendly, and cost-effective process. USP uses high-frequency sound to form bubbles that expand, contract, and eventually collapse, generating high-speed liquid jets, powerful shockwaves, and acoustic streaming effects. Despite its widespread application, use of USP remains mostly empirical. For processes involving the synthesis and production of materials, harnessing the power of cavitation requires an understanding of the fundamental mechanisms driving USP. In this presentation, we record recent studies to analyse, optimise, and control USP for applications related to grain refinement of aluminium alloys, exfoliation of 2D nanomaterials, and processing of composites. High-speed cameras and *in situ* synchrotron imaging were used to visualise cavitation dynamics, coupled with state-of-the-art hydrophones to detect acoustic waves and shockwave emissions. Results show that optimised USP with the help of advanced modelling significantly improves grain refinement of aluminium alloys in processes such as direct-chill (DC) casting, where shockwaves are primarily responsible for the fragmentation of intermetallic crystals/dendrites. Furthermore, shockwaves act as the main driving mechanism for the exfoliation of 2Ds while cavitation activity enhances fibre dispersion in highly viscous polymers, improving matrix stability and strength.

Contributed Papers

1:30

1pPA2. Improving sonochemical efficiency by pulsing cylindrically converging acoustic waves. Cherie Wong (Eng. Sci., Univ. of Oxford, Begbroke Directorate, Begbroke Hill, Woodstock Rd., Oxford OX5 1PF, United Kingdom, cherie.wong@eng.ox.ac.uk), Jason L. Raymond, Lillian N. Usadi, Zhiyuan Zong, Stephanie Walton (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Adam Sedgwick (Dept. of Chemistry, Univ. of Oxford, Oxford, United Kingdom), and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Sonochemistry is considered a green alternative for chemical synthesis, water treatment, and hydrogen production. There are several steps for sonochemical reactions. First, the nucleation of a bubble must occur through the application of an intense acoustic pressure field. After further ultrasonic irradiation, the bubble must grow to a diameter beyond a certain threshold so that it undergoes inertial collapse during the compressional phase of the acoustic field. Finally, this inertial collapse must achieve a quasi-adiabatic compression to cause pyrolysis of the gas or vapour molecules into radicals, which is the basis of sonochemical reactions. Conventional sonochemical reactors have sub-optimal acoustic fields and therefore require long reaction times to produce meaningful yields. We present a novel sonochemical reactor design that generates cylindrically converging waves into a reaction vessel. To determine the sonochemical efficiency (SE) of our design, we measured the hydroxyl radical yield under different operating conditions dosimetrically. We also compare the SE from an ultrasonic bath, immersed horn, and cup-horn reactor. These comparisons indicate that our reactor had a SE 200-fold, 3-fold, and 70-fold higher than the conventional reactors, respectively.

1:45

1pPA3. Optimization of radical polymerization in water through the characterization of the pressure field in nested cavities excited by a piezoelectric transducer. Tristan Nerson (LAUM, UMR-CNRS 6613, Le Mans Université, LAUM, Ave. Olivier Messiaen, Le Mans 72085, France, tristan.nerson.etu@univ-lemans.fr), Tharin Sensan (IMMM, UMR-CNRS 6283, Le Mans Université, Le Mans, France), Mathieu Gaborit (LAUM, UMR-CNRS 6613, Le Mans Université, Le Mans, France), Sandie Piogé (IMMM, UMR-CNRS 6283, Le Mans Université, Le Mans, France), Aroune Duclos (LAUM, UMR-CNRS 6613, Le Mans Université, Le Mans, France), Sagrario Pascual (IMMM, UMR-CNRS 6283, Le Mans Université, Le Mans, France), and Cyril Desjouy (LAUM, UMR-CNRS 6613, Le Mans Université, Le Mans, France)

The activation of a radical polymerization using ultrasound in water is an eco-friendly way to produce synthetic latexes, as it avoids using radical initiators and surfactants that negatively impact the environment. Such an experiment can be achieved by exciting a water bath (so-called generator) in which the reaction vessel is immersed with an ultrasonic transducer. Following data from the literature, several frequency bands may be used, but the present setup uses the 480-500 kHz range. Prior experiments by the same team and others have shown that such an excitation successfully fosters polymerization. The question now pertains to optimizing the setup geometry for a more efficient reaction. In the framework of linear acoustics, the experimental setup can be described as nested cavities and the pressure field in the different domains is modeled with modal theory. By describing the cavities with parameters related to their shape and position, this contribution aims at determining optimized sets of parameters to enhance the radicals generation rate and connect it to the stable or transient cavitation activity produced by sonication. The data from the linear model are compared with some experimental data at high frequencies and high intensity.

Invited Paper

2:00

1pPA4. High frequency ultrasound for destruction of per- and poly- fluoroalkyl substances. Timothy Sidnell (Chemical and Process Eng., Univ. of Surrey, Guildford, United Kingdom), Jake Hurst (Arcadis, Manchester, United Kingdom), Judy Lee (Chemical and Process Eng., Univ. of Surrey, Guildford, United Kingdom), and Madeleine Bussemaker (Chemical and Process Eng., Univ. of Surrey, M.S. J2, Guildford, Surrey GU2 7XH, United Kingdom, m.bussemaker@surrey.ac.uk)

Per- and poly-fluorinated substances (PFAS) are man-made chemicals, used for more than 50 years in plastics, waterproofing, non-stick pans, surfactants and aqueous firefighting foams. Due to their widespread use, and extreme persistence PFAS are now everywhere in the environment. However, PFAS are toxic, bioaccumulate in animals and plants and can lead to numerous ill-health effects. Many PFAS are now banned or heavily regulated and hence legacy and environmental PFAS require destruction. PFAS contain a hydrophobic fluorinated carbon chain and hydrophilic head group(s) giving them desirable chemical properties which also makes them extremely recalcitrant to typical degradation methods. This work will give an overview of how sonolytic degradation of PFAS is expected to be useful toward total remediation of environmental contaminations. We focus on high frequency ultrasound (1001,000 kHz) and examine parameters such as frequency, liquid height, power density, and reactor configuration. The approach will be discussed in reference to remediation samples of fire fighting foam and landfill leachate concentrate. Considerations of how high frequency ultrasound may be employed in an industrial setting will be discussed.

Contributed Papers

2:30

1pPA5. Cavitation noise characterization and classification using machine-learning techniques. Lillian N. Usadi (Eng. Sci., Univ. of Oxford, Parks Rd., Oxford, England OX1 3PJ, United Kingdom, lillian.usadi@magd.ox.ac.uk), Jason L. Raymond (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Cherie Wong (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Qiang Wu (Dept. of Eng. Sciense, Univ. of Oxford, Oxford, United Kingdom), Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom), Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, Oxfordshire, United Kingdom), and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Noise characterization of cavitation has use in biomedicine, sonochemistry, and waste degradation. Typically, spectral features of the cavitation noise signals obtained through Fourier analysis are specific to the experimental set-up and are analyzed to classify the signals in terms of presumed domains of bubble behaviors. The objective of this research was to develop the experimental capabilities and algorithm to monitor and classify cavitation a priori using machine learning (ML) in a precise, repeatable, and translatable fashion among various setups and applications. Initially, simultaneous high-speed videos and passive acoustics maps were acquired for a single oscillating bubble in a simple setup. Machine learning methods such as support vector machine and convolutional neural networks were used to classify the acoustic data and train a classification algorithm. The algorithm was then adapted for a novel sonochemical reactor that induces spontaneous cavitation nucleation and utilizes a passive cavitation detector to monitor cavitation noise. Further research will be done in correlating potassium iodide (KI) degradation with frequency spectra using ML classification in order to optimize the sonochemical efficiency.

2:45

1pPA6. Control of amyloidogenic protein aggregation by ultrasonic irradiation. Kichitaro Nakajima (Graduate School of Eng., Osaka Univ., Yamadaoka 2-1, Suita, Osaka 5650871, Japan, k.nakajima@prec.eng.osaka-u.ac.jp), Tomoki Ota, Keiichi Yamaguchi, Yuji Goto, and Hirotsugu Ogi (Graduate School of Eng., Osaka Univ., Suita, Japan)

Protein aggregation causes intractable diseases such as Alzheimer's disease. In these diseases, crystal-like protein aggregates, so-called amyloid fibrils, form *in vivo* and induce severe malfunction of biological tissues. Amyloid fibril formation comprises primary nucleation from a supersaturated solution of protein monomers and subsequent fibril growth. Within them, primary nucleation possesses an extremely high energy barrier, resulting in a long induction time for nucleation, typically over a decade *in vivo* and several days even in a test tube. Our previous study (1) revealed that ultrasonic irradiation to supersaturated protein solution drastically accelerates amyloid formation because of the effects of ultrasonic cavitation, which is applicable to clinical diagnosis (2). However, the degree of acceleration has not been controlled yet. Furthermore, the reproducibility of the induction time for nucleation highly depends on ultrasonic-irradiation conditions and the degree of supersaturation of initial protein solutions. Herein, we systematically investigate the optimized condition to accurately control amyloid formation using an originally developed sonoreactor (3) and discuss the underlying mechanism governing ultrasonic induction of amyloid formation. (1) K. Nakajima *et al*., Ultrason. Sonochem. 36, 206-211 (2017). (2) K. Nakajima *et al*., Nat. Commun. 13, 5689 (2022). (3) K. Nakajima *et al*., Ultrason. Sonochem. 73 105508 (2022).

Session 1pSA

Structural Acoustics and Vibration, Engineering Acoustics, Noise, Signal Processing in Acoustics, and Physical Acoustics: Acoustic Excitation of Structures II

Brian E. Anderson, Cochair
Physics & Astronomy, Brigham Young University, Department of Physics & Astron., N245 ESC, Provo, UT 84602

Trevor Jerome, Cochair
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Ian C. Bacon, Cochair
Department of Physics & Astronomy, Brigham Young University, N283 ESC, Brigham Young University, Provo, UT 84602

Chair's Introduction—1:00

Invited Papers

1:05

1pSA1. Excitation of structures by partially correlated pressures: A review of diffuse acoustic field and turbulent boundary layer models. Micah Shepherd (Brigham Young Univ., N249 ESC, Provo, UT 84602, mrs74@byu.edu)

Structures are sometimes excited by pressure distributions which exhibit complex spatial correlation. This differs from common acoustic excitations since the pressure at one location is only partially correlated with the pressure at another location due to inherent spatial randomness within the forcing function. Two forcing functions which exhibit partially-correlated pressures are the diffuse acoustic field (DAF) and turbulent boundary layer (TBL) flow. A basic model for representing the spatial correlation for these two forcing functions will be reviewed in both the spatial and wavenumber domains. Recent approaches for computing the vibration of structures excited by DAF or TBL flow will then be summarized. Interesting physical effects, such as intermodal coupling, will be highlighted to illustrate the importance of properly modeling partial correlations when they exist.

1:25

1pSA2. Acoustic resonators for detecting low frequency events. Paul Schmalenberg (Electronics Res. Dept., Toyota Res. Inst. of North America, Toyota Res. Inst. of North America, 1555 Woodridge Ave., Ann Arbor, MI 48105, paul.schmalenberg@toyota.com), Parveen Singh, Frederico Severgnini, and Ercan Dede (Electronics Res. Dept., Toyota Res. Inst. of North America, Ann Arbor, MI)

Acoustic resonators and related concepts have long been the subject of research for acoustic filtering in free space or energy harvesting applications. In this work, we instead apply acoustic resonators to low frequency (< 50 Hz) non-contact bio-signal sensing within a vehicle cabin. We describe our approach to the design of the resonator and method of simulation in this low frequency regime. Validation of select designs through experiments in both free space and using a coupled bodies method is discussed. We also demonstrate the bio-signal collection *in situ* and discuss challenges associated with a real-world, noisy, dynamic environment. Future work including signal processing of noisy data to recover the base bio-signal and ideas for miniaturization of the device are touched upon.

1:45

1pSA3. Multi-resolution non-contact laser-sensing damage detection technique for composite laminates. Lukasz Ambrozinski (Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Mickiewicza, 30, Krakow, N/A 30-059, Poland, ambrozin@agh.edu.pl), Jakub Spytek, Jakub Mrowka, and Lukasz J. Pieczonka (Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland)

Numerous ultrasonic (UT) modalities have been developed to perform inspections of engineered structures. However, increased precision often comes at the cost of longer inspection times. As an example, guided waves propagating in plate-like structures allow fast inspection of large areas due to their long propagation distances. Damage characterization accuracy is, however, limited in this case due to relatively large wavelengths. On the other hand, local inspection with high-frequency bulk waves provides much more accurate damage characterization thanks to shorter wavelengths. The drawback of local inspection is the long scanning time necessary for large components. Here, we discuss a two-step laser-interferometer-based non-contact approach for the inspection of carbon-fiber-reinforced structures. First, a full-field screening using guided waves is performed to determine the possible damage locations. The identified hot

spots are further evaluated in the second step with a high-frequency bulk wave system. A piezoceramic exciter is used for the guided wave setup, while a pulsed laser excitation source is used in the bulk wave setup. The proposed approach minimizes sample preparation time and is completely non-intrusive. The feasibility of the proposed framework is demonstrated in a complex shape, thin-walled composite structure with multiple defects.

2:05

1pSA4. Contactless modal analysis of elastic structures with acousto-elastic transmission matrix. Chloe Palerm (SafranTech, Institut Langevin, Paris, France), Claire Prada (Institut Langevin, CNRS, Paris, France), Benoit Gerardin (SafranTech, Paris, France), Arnaud Talon (Safran Helicopter Engines, Bordes, France), and Julien de Rosny (Institut Langevin, CNRS, Paris, France, julien.derosny@espci.fr)

Modal analysis is a major issue in the industry to identify resonances in mechanical parts. Indeed, resonances can induce high vibration levels that are potentially destructive. Active modal analysis methods require on the one hand to excite the controlled part and on the other hand to record the induced displacements/acceleration/stresses. Conventional methods, called SIMO, involve a mechanical excitation source associated with several receivers. More rarely, the analysis is performed with several sources and one receiver (MISO). Finally, for several years, MIMO techniques with several sources and several receivers are commonly implemented. However, for some parts, it may be difficult to implement an array of mechanical sources. That is why, here, we propose to carry out the modal analysis with a set of 8 loudspeakers as sources, and the measurements are performed at a large number of measurement points using a LASER vibrometer. The analysis of the singular value decomposition of the contactless transmission matrix allows to identify superposed modes. We will discuss the advantages and disadvantages of such a configuration. This method based on acousto-elastic coupling is successfully applied to the modal analysis of two axi-symmetric parts: a pinion and a rotating wheel. Finally, we will see that the same device can be used to control vibration fields.

Contributed Papers

2:25

1pSA5. Exploring ways to maximize the amplitude of steady state broadband noise signals using time reversal in order to excite structures acoustically. Rylee Russell (Phys. & Astronomy, Brigham Young Univ., BYU, Phys. & Astron., N283 ESC, Provo, UT 84602, rsrussell01@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Exciting structures with a broadband noise signal is a method used to assess a structure's health and how it responds to the noise. Past work has shown that time reversal (TR) acoustics has provided a robust method to focus impulses at high amplitudes. Maximization of the highest possible amplitude while maintaining the desired spectral shape can be challenging. This study explores the benefits of using TR with equalization methods to a steady state broadband noise signal to acoustically excite structures. Trade-offs exist between the possible amplitude of the focused noise and the duration of the noise and the amount of equalization employed. The spatial extent of the focusing will also be discussed. The use of TR to generate high amplitude broadband noise is compared to standard point application of noise equalization with the overall goal to use this high amplitude noise to excite structures under test.

2:40

1pSA6. On the coupled dynamics of near-field acoustic levitation bearing. Yaoke Wang (Mech. Eng., Northwestern Univ., 710 oakton St., Evanston, IL 60202, yaokewang2020@u.northwestern.edu)

Near field acoustic levitation (NFAL) bearings, as an alternative non-contact bearing solution, has been proposed and advanced in variable aspects in recent years. To further investigate the property of NFAL and fulfill its theoretical frame, a model based on the stiffness model to analysis the coupled dynamics of the NFAL bearing system, which takes the impact of dynamic stiffness on the levitated object into consideration is proposed. In this model, due to the impact of the driving surface vibration and the dynamic stiffness on the levitated object, a real-time movement of the levitated object is superimposed on the time-averaged movement. By inducing the real-time vibration of the levitated object into the model, it is found that the levitation force and the driving frequency will no longer follow an approximate hyperbolic tangent relationship. The model indicates that, when approaching a certain frequency, the stiffness-mass system of the bearing resonates, and then the phase of the two surfaces begins to increase rapidly, thereby increasing the amplitude of the actual air gap. It is verified with experiments that the object can only be levitated above the resonant frequency.

2:55

1pSA7. Improved fracture predictions using Acoustic Emissions data. Rahul Sheoran (Phys., Panjab Univ., 3, Type 4, CRPF Campus, Hallomajra, Chandigarh 160002, India, 27sheoran@gmail.com)

The mechanical health of any composite specimen can be monitored by recording the acoustic emissions (AE) from the surface and hence can be used to predict mechanical fracturing of the material. The non-destructive and non-invasive technique of AE is an established method to monitor the mechanical health of a structure. The AE captured from a specimen undergoing mechanical stress appears to display power law like distribution. While empirical data almost never follow power law behaviour completely, which, in turn makes the estimation of its exponent (b-value) to be highly erroneous. Literature shows that attempts have been made by the researchers to filter out the non-power law parts of the dataset, however, issues such as choice of upper and lower bounds, and fixed values of these bounds for different types of datasets remain a challenge for efficient and robust estimation of b-value. In the present work, the author has proposed a novel technique to rectify the inherent shortcomings of b-value and improved b-value (ib-value) estimation methodology. The proposed method, which is computationally light to implement, has been verified on synthetic, experimental datasets and reputed datasets from literature. The results show measurable quantitative improvements in the estimation of b-value parameter compared to previous reported methods in the literature.

Session 1pSC

Speech Communication: Speech Perception I (Poster Session)

Christine H. Shadle, Chair
Haskins Laboratories, 300 George St, Suite 900, New Haven, CT 06511

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 3:45 p.m.

Contributed Papers

1pSC1. Comparing pre-linguistic normalization models against US English listeners' vowel perception. Anna Persson (Swedish Lang. and Multilingualism, Stockholm Univ., Universitetsvägen 10D, Stockholm 10691, Sweden, anna.persson@su.se) and T. Florian Jaeger (Brain and Cognit. Sci. and Comput. Sci., Univ. of Rochester, Rochester, NY)

One of the central computational challenges for speech perception is that talkers differ in pronunciation–i.e., how they map linguistic categories and meanings onto the acoustic signal. Yet, listeners typically overcome these difficulties within minutes (Clarke & Garrett, 2004; Xie *et al.*, 2018). The mechanisms that underlie these adaptive abilities remain unclear. One influential hypothesis holds that listeners achieve robust speech perception across talkers through low-level pre-linguistic normalization. We investigate the role of normalization in the perception of L1-US English vowels. We train ideal observers (IOs) on unnormalized or normalized acoustic cues using a phonetic database of 8 /h-VOWEL-d/ words of US English (N = 1240 recordings from 16 talkers, Xie & Jaeger, 2020). All IOs had 0 DFs in predicting perception—i.e., their predictions are completely determined by pronunciation statistics. We compare the IOs' predictions against L1-US English listeners' 8-way categorization responses for /h-VOWEL-d/ words in a web-based experiment. We find that (1) pre-linguistic normalization substantially improves the fit to human responses from 74% to 90% of best-possible performance (chance = 12.5%); (2) the best-performing normalization accounts centered and/or scaled formants by talker; and (3) general purpose normalization (C-CuRE, McMurray & Jongman, 2011) performed as well as vowel-specific normalization.

1pSC2. Comparing pre-linguistic normalization models against US English listeners' perception of natural and resynthesized vowels. Anna Persson (Swedish Lang. and multilingualism, Stockholm Univ., Universitetsvägen 10D, Stockholm 10691, Sweden, anna.persson@su.se) and T. Florian Jaeger (Brain and Cognit. Sci. and Comput. Sci., Univ. of Rochester, Rochester, NY)

Talkers vary in their vowel pronunciation. One hypothesis holds that listeners achieve robust speech perception through pre-linguistic normalization. In recent work (also submitted to ASA), we modeled listeners' perception of naturally produced /h-VOWEL-d/ words. The best-performing normalization models accounted for ~90% of the explainable variance in listeners' responses. Here, we investigate whether the remaining 10% follow from (1) other mechanisms or whether (2) they reflect listeners' ability to use more cues than available to models. We constructed a new set of *synthesized* /h-VOWEL-d/ stimuli that varied *only* in F1 and F2. Unsurprisingly, listeners (N = 24) performed worse on these synthesized stimuli than on the natural stimuli (estimated as inter-listener agreement in categorization). Critically though, we find (1) that the same normalization accounts that best explained listeners' responses to natural stimuli also perform best explaining responses to synthesized stimuli; (2) the best performing model again accounted for ~90% of explainable variance. This suggests that the 'failure' of normalization accounts to fully explain listeners' categorization behavior is *not* due to restrictions in the ability to feed our models all available cues. Rather, normalization alone—while critical to perception—seems insufficient to fully explain listeners' ability to adapt based on recent input.

1pSC3. Studying Mandarin Tone sandhi in interaction. Eric Pelzl (Psych. and Linguist, The Penn State Univ., 111 Moore Bldg., Penn State University, University Park, PA 16802, ezp218@psu.edu), Anne J. Olmstead, and Navin Viswanathan (Commun. Sci. and Disord., The Penn State Univ., University Park, PA)

The Mandarin low-dipping tone (T3) undergoes an alteration, tone sandhi, when followed by another T3. The resulting F0 is superficially the same as that of the rising tone (T2). We investigated how Mandarin speakers adapt their speech production to overcome potential ambiguities induced by T3 sandhi during an interactive task. Ten pairs of Chinese participants completed an interactive phrase matching task. Participants were shown displays with Chinese phrases (surname + title, e.g., 1a-b). One participant read an indicated phrase which the other selected from their display. There were two conditions. In the no sandhi condition (1a-b), the title did not induce sandhi, and should result in distinct F0 patterns for the surnames. In the sandhi condition (2a-b), the title induced T3 sandhi, and should result in homophonous surnames. 1a 卢侦探 (lu2 tʃ̺ən1tan4 "Detective Lu") 2a 卢主任 (lu2 tʃ̺u3ɹ̺ən4 "Director Lu") 1b 鲁侦探 (lu3 tʃ̺ən1tan4 "Detective Lu") 2b 鲁主任 (lu3 tʃ̺u3ɹ̺ən4 "Director Lu") Task performance showed clear evidence of sandhi-induced ambiguity. Examination of accuracy and tone acoustics suggested that pairs deployed different strategies to attempt to overcome sandhi-induced ambiguity, including exaggerating F0 rise or T3 duration, or adding a pause to avoid applying sandhi processes.

1pSC4. Titrating effects of acoustic variability on context effects and psychometric function slopes in speech categorization. Caleb J. King (Psychol. and Brain Sci., Univ. of Louisville, 2301 S 3rd St., Louisville, KY 40292, cjking03@louisville.edu), Anya E. Shorey, and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

In temporal contrast effects (TCEs, also termed speaking rate normalization), a fast context sentence makes the onset of the subsequent target word sound longer, and vice versa. King, Sharpe, Shorey, and Stilp (2022 ASA) reported that variability in the talkers or lexical contents of context sentences each decreased TCE magnitudes. However, in that experiment, acoustic variability was not tightly controlled. Using the same stimuli as King *et al.*, listeners completed three blocks, hearing one talker speaking the same sentence (1 Talker/1 Sentence) or 200 different sentences (1/200), or 200 different talkers speaking 200 different sentences (200/200). On each trial, listeners heard one (fast/slow) context sentence, then identified the subsequent target word as "deer" or "tier." In one experiment, speaking rates were matched across 1/200 and 200/200 blocks; TCE magnitudes were comparable and psychometric slopes did not differ. In another experiment, all slow sentences were set to 2.67 syllables/sec and all fast sentences to 8 syllables/sec. Differences due to sentence variability (1/1 versus 1/200) were

extinguished; TCE magnitudes were smaller and psychometric slopes were shallower in the 200/200 block. Thus, multiple levels of variability must be considered in how they shape speaking rate normalization. [Work supported by NIH R01DC020303.]

1pSC5. Experimental evidence: Directionality preference of tone absorption. Ken Wing Hang Kwong (Linguist and Modern Lang., CUHK, G19, Leung Kau Kui Bldg., The Chinese University of Hong Kong, Hong Kong, kenwhkwong@link.cuhk.edu.hk)

Tone absorption is a very common tone alternation process in African and Asian tone languages, in which a part of the contour tone is absorbed by another tone, and this process can take place in two directions. Typologically, right-dominant tone absorption is much more attested than the left-dominant counterpart, although both are predicted possible. Such typological asymmetry hypothesizes the argument that there is a learnability difference between left- and right-dominant absorption. Specifically, given that the two patterns are equally complex, we hypothesized the difference is due to variations in phonetic groundedness. An artificial language learning paradigm was used for this study. We extracted possible tone absorption patterns from Tianjin Mandarin with reference to the Cantonese tonal inventory and designed an artificial language incorporating these absorption patterns. 56 native Cantonese participants took part in a word naming task, consisting of a pretest, a learning, and a test phase. They were taught a monosyllabic name of a color, and a monosyllabic name of a monster, and asked to combine them to form a disyllabic compound. Results showed that the right-dominant learners had significantly higher accuracy rates after learning. The analysis of reaction time (RT) revealed that right-dominant learners had significantly shorter RTs than the left-dominant group learners. Both findings showed that right-dominant tone absorption was better learned, suggesting that it is a universal directionality preference.

1pSC6. Positive implicit association to human babies' vocalization. Maria Fernanda Alonso Arteche (School of Commun. Sci. & Disord., McGill Univ., 1295 Rue des Carrieres, apt 402, Montreal, QC H2S 0E1, Canada, maria.alonsoarteche@mail.mcgill.ca), Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada), and Lucie Menard (Département de Linguistique, Université du Québec à Montréal (UQAM), Montréal, QC, Canada)

Cuteness in infants has mainly been associated with their morphological facial features. Infant cuteness promotes social interaction, empathy, and caregiving behaviors. It has been suggested that cuteness could go beyond the visual aspect to infants' auditory and olfactory characteristics. The current research examines whether there is a positive valence toward babies' neutral vowel vocalizations. Adult participants completed four Single Category Implicit Association Tasks (SC-IAT) using baby, adult, kitten, and cat vocalizations. Baby and adult vocalizations were vowels synthesized using VLAM. In this task, adults had to sort positive and negative written words and the vocalizations in the correct category; response times were measured. Preliminary results (response time) show that infant voices were associated with positive dimensions ($M = 0.27$), whereas adult ($M = 0.06$), cat ($M = 0.07$), and kitten ($M = 0.08$) showed neither a positive nor negative implicit association. A similar study testing facial features obtained comparable results (Senese *et al.*, 2013). These findings point to a positive implicit association to babies' vocalizations, suggesting that there is an auditory cuteness component specific to human infants. We will also discuss ongoing work focused on elucidating the acoustic characteristics that specify auditory cuteness, the role it plays in infant development, and the underlying neural mechanisms.

1pSC7. Why we talk about voices as we do. Jody Kreiman (Head and Neck Surgery and Linguist, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

The problem of how to characterize voice quality is an endless source of debate and frustration across disciplines. The richness of the vocabulary available to describe voice is overwhelming, but the density of the information conveyed by voice has led some scholars to conclude that language can never adequately specify what we hear. Others have argued that terminology derives from tradition and lacks an empirical basis, so that language-based scales are inadequate *a priori*. Finally, efforts to link terms to acoustic signal characteristics have had limited success. However, a reconsideration suggests that a few terms appear consistently across studies, disciplines, and eras. These terms align with dimensions that account for acoustic variance in voice across speakers, regardless of gender, language spoken, or the kind of speech sample, and correlate with physical size and arousal across many species. They, thus, may have an evolutionary basis. This suggests talk about voices rests on a bedrock of biology: We have evolved to perceive voices in terms of size/arousal, and these factors structure both voice acoustics and the language we use to describe voices. Such linkages could help integrate studies of physical signals and their meaning, producing a truly interdisciplinary approach to voice.

1pSC8. Tonal center of gravity predicts variation in the interpretation of rising and falling intonation in American English. Thomas Sostarics (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, tsostarics@gmail.com) and Jennifer Cole (Linguist, Northwestern Univ., Evanston, IL)

In English, the pitch trajectory at the end of a phrase distinguishes assertions from questions, through falling versus rising intonation, respectively. This study asks if, within these broad classes, variation in the shape or slope of the pitch excursion is meaningful. Recent work proposes a phonological distinction between two tonally distinct sub-categories of rising intonation: *shallow* rises favor an assertive interpretation, while *steep* rises favor a question interpretation (Jeong 2018). We test an alternative analysis, where the probabilistic association between intonation and assertive/inquisitive interpretation arises not from a discrete difference *between* shallow and steep rises, but from meaningful variation *within* the rising class. We present findings from two perception experiments where listeners hear short utterances and identify the speaker's intention to ASK or TELL. Pitch resynthesis (PSOLA) was used to create a stimulus continuum of phrase-final rising and falling F0 trajectories that vary in the slope and shape of monotonic F0 trajectories. Our results show a probabilistic relationship between interpretation (ASK/TELL) and rise variation, but do not support the previously proposed phonological distinction between shallow and steep rises. We propose a more parsimonious model of within-category variation grounded in the Tonal Center of Gravity acoustic measure (Barnes *et al*. 2012).

1pSC9. Hearing type influences preschoolers' phoneme-level phonological awareness. Erin Ingvalson (Dept. of Speech and Hearing Sci., Univ. of Washington, Dept. of Speech and Hearing Sci., University of Washington, Seattle, WA 98105, eingvals@uw.edu), Tina M. Grieco-Calub (Rush Univ. Medical Ctr., Skokie, IL), Lynn Perry (Dept. of Psych., Univ. of Miami, Coral Gables, FL), and Mark VanDam (Washington State Univ., Spokane, WA)

Children who are deaf/hard of hearing (D/HH) have lower performance on test of phonological awareness, particularly phoneme-level tasks, than children with normal hearing. These performance deficits have been seen both in children who use cochlear implants (CIs) and children who use hearing aids (HAs), but few studies have explicitly compared the effect of hearing type on phonological awareness. Forty-one preschoolers who were D/HH and 36 preschoolers with normal hearing participated. Participating children were between 3- and 4-years-old. Of the D/HH children, 14 CI were users and 27 were HA users. All children completed the phonological awareness test of the Test of Preschool Early Literacy, which assesses sound blending and elision, and the sound blending test of the Tests of Early Cognitive and Academic Development. Data are raw scores. There was a clear effect of hearing type. Children who used CIs had the lowest performance (TOPEL $M = 8.62$, ECAD $M = 2.62$), followed by children who use HAs (TOPEL $M = 10.76$, ECAD $M = 4.92$), with children with normal having the best performance (TOPEL $M = 15.56$, ECAD $M = 8.03$). Children who use CIs have more difficulty with phoneme-level tasks than children who use HAs, which likely has implications for their future language and literacy development.

1pSC10. Validating acoustic predictors of sentence intelligibility using a racially diverse corpus. Michael L. Smith (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, smit8854@umn.edu) and Benjamin Munson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

The goal of the present work is to validate a set of widely used acoustic measures that have been found to predict talker intelligibility using a new, racially diverse corpus of native speakers of American English. Previous work by Bradlow *et al* (1996, Speech Communication) predicted sentence intelligibility on a set of 20 demographically unspecified talkers from the Midwest, where they found changes in F0 and larger vowel spaces to be associated with higher sentence intelligibility. Here, we examine whether these same acoustic features can be generalized to a more racially diverse population of talkers, of which only 9 are white and non-Latine. The current corpus comprises 28 talkers, where each talker produced 60 sentences. Here, we focus on a core set of sentences, 10 Havard/IEEE and 10 BEL sentences, that all 28 talkers produced. Key acoustic features such as F0, speaking rate, and first and second formants of vowels, were extracted to associate with intelligibility measurements for these individuals. Intelligibility measures are taken from a previous study (Tripp *et al*., 2022, J. Acoust. Soc. Am.). Results from the present work are important in generalizing which acoustic features impact intelligibility in a racially diverse corpus.

1pSC11. The perceptual consequences of style shifting. Abby Walker (Virginia Tech, 181 Turner St NW, Blacksburg, VA 24061, ajwalker@vt.edu)

We focus on the perception of changes in production made by self-identified bidialectal Southern (SUSE) and Mainstream (MUSE) US English speakers, who were recorded reading words in both dialects. Acoustical analysis shows quantifiable changes across the two guises, but that speakers' MUSE guises still looks more Southern than monolingual speakers of MUSE-like dialects, consistent with claims that many bidialectals do not categorically shift between dialects (Hazen 2001). The stimuli were then used in three listening tasks – a lexical decision task, a transcription task and a speeded dialect classification task – and all show evidence that listeners ($N = 123$) were sensitive to the changes made by the speakers in their two guises. In fact, for the measures of communicative ease (accuracy in lexical decision, transcription accuracy), responses to the MUSE guise of the bidialectal speakers was indistinguishable from those to monodialectal MUSE speakers. For dialect classification, the MUSE guise of the bidialectal speakers is heard as significantly less Southern than their Southern guise, but more Southern than monodialectal MUSE speakers, and there is a significant effect of listener dialect, with Southern listeners more likely to hear the bidialectal MUSE guise as not Southern. The results sugget that self-identified bidialectal speakers likely use different criteria in assessing bidialectalism (intelligibility, in group perception) than linguists, who typically focus on analyzing speech production.

1pSC12. What we do (not) know about the mechanisms underlying adaptive speech perception. Xin Xie (Univ. of California Irvine, SSPB 2223, Irvine, CA 92617, xxie14@uci.edu), T. Florian Jaeger, and Chigusa Kurumada (Univ. of Rochester, Rochester, NY)

Recognition difficulties associated with unfamiliar talkers and accents can dissipate with exposure, sometimes within minutes. Although such adaptivity in response to unfamiliar input is now considered a fundamental property of speech perception, the underlying *mechanisms* remain largely unknown. Past work has hypothesized three, computationally heterogeneous, mechanisms: (1) low-level, pre-linguistic, signal normalization, (2) changes in/selection of linguistic representations, or (3) changes in post-perceptual decision-making. Direct comparisons of these hypotheses, or combinations thereof, have been lacking. We describe a general computational framework that—for the first time—implements all three mechanisms. We demonstrate how the framework can be used to predict adaptive changes of perception within a commonly used experimental paradigm (e.g., perceptual recalibration) under each of the three mechanisms. Using this approach, we find that—at the level of data analysis presently employed by most studies in the field— behavioral results that can be obtained from common experimental paradigms do not distinguish between the three mechanisms. This highlights the need for changes and refinements in research practices, so that future experiments provide data that can reliably distinguish between the three hypothesized mechanisms and/or test the influence from combinations of them. We provide specific recommendations towards this direction.

1pSC13. Examining the feasibility of integrating pupillometry measures of listening effort into clinical audiology assessments. Ian Phillips (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., America Bldg., Rm. 5400, 4954 North Palmer Rd., Bethesda, MD 20889, ian.phillips7.ctr@health.mil), Gregory M. Ellis (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Evanston, IL), Bridget McNamara (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Jacob Lefler (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Kristina DeRoy Milvae (Dept. of Communicative Disord. and Sci., Univ. at Buffalo, College Park, MD), Sandra Gordon-Salant (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Stefanie E. Kuchinsky, and Douglas S. Brungart (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Pure-tone thresholds are often a poor predictor of hearing difficulties experienced by patients in audiology clinics. Approximately 25 million Americans may experience hearing difficulties despite having audiometric thresholds within normal limits. Pupil diameter increases with increasing listening effort and could provide an objective measure for determining the presence of such subclinical hearing problems. However, the cost and technical requirements of research-grade eye trackers have limited their use in audiology clinics. This study aims to determine the feasibility of using a commercially-available head mounted display (HMD) with integrated eye tracking to collect pupillometry measures during audiological testing. Pupillary data were collected for $N = 46$ younger normal-hearing adults while they completed clinically-relevant tests including pure tone audiometry, gaps-in-noise (GIN), dichotic digits, and speech-in-noise tests. Across tasks, task-evoked pupillary responses were generally sensitive to differences in listening demands. Results suggest audiologic testing with commercial HMD eye trackers is feasible and provides additional objective information related to listening effort that may help clinicians better understand and address patient needs. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of Army/Navy/Air Force, Department of Defense, or U.S. Government.]

1pSC14. Children's perception of gender in other children's voices. Christopher E. Holt (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, holt.465@osu.edu), Quinn Baumgartner, Ewa Jacewicz, and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Studies investigating perception of speaker gender in children have mostly been conducted with adults as listeners in experimental tasks. Based on these adult judgments, results converge in showing that the gender of prepubescent children can be identified well above chance (even in the absence of reliable anatomical differences related to their vocal tract morphology) and that sentences provide more gender cues than do isolated syllables. Our previous work added that adults can utilize sociocultural cues in regional dialects in making their gender identification decisions (Holt *et al*., 2022). Here, we examine how children perceive gender in other children's voices, and whether their judgments differ appreciably from adults. Children ages 8-11 years listened to various utterances (isolated syllables, read sentences, and spontaneous speech) of age-matched peers from three dialect regions in the US. Our provisional findings are that, overall, children are less sensitive to gender cues than the adults, and their judgments are more dependent on the talker's age than on information in longer passages of speech in sentences and talks. Contrary to adults, regional dialect does not seem to influence children's decisions, suggesting that perceptual

attunement to a particular variety may not facilitate gender identification in other children.

1pSC15. Listener descrimination between age differences in two speakers. Lara Ozdogan (Linguist, Univ. of AB, 28 1428 Hodgson Way NW, Edmonton, AB T6R 3P8, Canada, ozdogan@ualberta.ca) and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

Often when listeners hear a speaker they have an impression of the age of the speaker. The present research seeks to determine how well listeners are able to identify a speaker's age when provided with two different recordings from one speaker. Previous research has asked participants to place a speaker's age on a continuum or estimate the age of the speaker. In the present study, we use an AX task which asks listeners to determine whether the X recording is younger or older, from speech recorded from two speakers spanning approximately 20 years. We also asked a few questions with regard to the listeners' language background, age, and experience with populations over the age of 40. We analyze the descrimination responses of our listeners, and find that listeners are not much better than chance in determining age differences in the recordings. We do find that as the number of years between recordings increases, participants are more accurate at determining the age difference between recordings. Listeners have fairly specific ideas about what acoustic features they are using to perform their descrimination, however these do not necessarily lead to more accurate descrimination.

1pSC16. Production and perception of novel consonant clusters. Matthew C. Kelley (Linguist, Univ. of Washington, GUG 407A, Seattle, WA 98195, mattck@uw.edu), Marina Oganyan, and Richard A. Wright (Linguist, Univ. of Washington, Seattle, WA)

The present study details a visual world paradigm eye tracking experiment on perception and production of novel consonant clusters. The clusters varied in difficulty based on sonority or perceptual salience scales. During the familiarization phase, listeners heard and watched a story on-screen and produced the names of novel creatures and objects. Each story focused on one cluster. Four creatures/objects were introduced corresponding to the (1) target cluster (e.g., [dlɑpɹ]), (2) epenthesized cluster competitor (e.g., [dəlɑkn]), (3) first cluster consonant (e.g., donkey), and (4) second cluster consonant (e.g., lollipop). During each trial, participants verbally identified one of the four creatures/objects, and their gaze was tracked. We analyzed the productions and gazes to investigate the relative difficulty of producing and perceiving the consonant clusters. We predicted participants would (1) correctly produce/perceive a cluster, (2) epenthesize a vowel/perceive vowel epenethesis, or (3) perceive/produce only one of the cluster sounds. We statistically analyzed the participants' propensity toward each behavior in both production and perception. As part of the analysis, we examined whether participant behavior was better predicted by analyzing the clusters with the sonority sequencing principle or in terms of acoustic salience and cue recoverability.

1pSC17. The effects of phonetics training on the production and perception of rhotic vowels in American English. Wil A. Rankinen (Commun. Sci. and Disord., Grand Valley State Univ., 500 Lafayette Ave. NE, 303D, Grand Rapids, MI 49503-3360, wil.rankinen@gvsu.edu) and Megan Andrzejewski (Commun. Sci. and Disord., Grand Valley State Univ., Sugar Grove, IL)

This talk examines how the American "r" phoneme affects the production and perception of American English front vowels in pre-rhotic positions. Rhotic vowels challenge listeners in accurately identifying the phonetic quality of vowels due to r-coloring affecting formants (Chung *et al.*, 2021; MacAllister *et al.*, 2017). For example, the vowel in "beer" is commonly perceived as [i] by listeners, yet can be produced as either a tense or lax high front vowel. This challenge is particularly concerning to aspiring clinicians learning phonetic transcription. This study investigates speakers' acoustic identity of pre-rhotic high and mid front vowels, and listeners' ability to accurately perceive pre-rhotic vowel identities. This study examines a 50-speaker sample of college-age participants separated into two groups (with and without phonetics training). All participants read a 100-word wordlist and completed a 200-word four-alternative forced-choice identification (AFCI) task. Furthermore, the AFCI task presents each word twice by either a male or a female voice. Listeners select from four options ("BEAT," "BIT," "BATE," "BET") and rate their confidence level for each choice. Preliminary data, demarcated in PRAAT, processed using FAVE-extract, and analyzed using R software, indicate the mapping between production and perception to be strongest for those with previous phonetic training.

1pSC18. Investigating the neural basis of the iambic/trochaic law. Fernando Llanos (Linguist, Univ. of Texas at Austin, Austin, TX, fllanos@utexas.edu), Timothy Stump, and Megan Crowhurst (Linguist, Univ. of Texas at Austin, Austin, TX)

The iambic/trochaic law predicts that listeners perceive loud sounds as rhythmic group onsets. While this law provides a solid framework to investigate the perception of foot inventories across languages, its neural underpinnings are currently unknown. Here, we test the hypothesis that the segmentation of rhythmic speech patterns is driven by neural oscillations that entrain in frequency and phase to the onset of rhythmic groups and units. We recorded EEGs evoked with a sequence of "ba" and "ga" units with their intensity manipulated to be perceived as a series of "BA ga" or "GA ba" groupings. We also manipulated the duration of rhythmic groups and units to achieve a rate of 3 units and 1.5 groups per second. Participants ($N = 12$) were instructed to segment each sequence ("BA ga BA ga …" and "ba GA ba GA …") multiple times while wearing an EEG cap with 32 scalp electrodes. Individual EEG spectra exhibited group and unit peaks at 1.5 Hz and 3 Hz, respectively ($ps < 0.001$). Additionally, individual EEGs entrained to each rhythmic group with a different phase ($ps < 0.01$). These results support the hypothesis that the segmentation of rhythmic speech patterns in the cortex is mediated by a temporal code driven by entrainment.

1pSC19. "Perceived" talker variability in high-variability perceptual training. Hanyong Park (Linguist, Univ. of Wisconsin-Milwaukee, Dept. of Linguist, University of Wisconsin - Milwaukee, Milwaukee, WI 53201-0413, park27@uwm.edu) and Jieun Lee (Linguist, Univ. of Kansas, Lawrence, KS)

It has been suggested that higher talker and acoustic variabilities in training materials are the driving factors impacting learning advantages, such as better generalization and longer retention of training effects. To investigate whether such advantages are diminished when trainees do not perceive talker variability in training stimuli, we gave three days of training on identifying Korean stop contrasts to three groups of English speakers: a group with three distinctive-voice stimuli (DV), a group with three similar-voice stimuli (SV), and a group with one-voice stimuli repeated three times (OV). We compared their performances twice: once right after the last training session and again one week later. Results showed the tendency that the DV group was better than the other two groups at generalizing their training effects to a novel talker, and this remained the same one week after their training. Although some participants in the SV group claimed they heard multiple voices during their training, their performance was similar to that of the OV group. Taken together, the results support the previously reported high-variability training benefits, and suggest that in training stimuli, the "perceived" talker variability, not just the talker and acoustic variability, is a driving factor for such benefits.

1pSC20. Judgements of a customer service agent's personality based on acoustic properties of their speech. Jieun Song (School of Digital Humanities and Computational Social Sci., Korea Adv. Inst. of Sci. and Technol., G08, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, jsong@kaist.ac.kr), Minjeong Kim (Ctr. for Digital Humanities and Computational Social Sci., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea), and Jaehan Park (KT Corp., Seoul, South Korea)

People frequently interact with customer service agents, whether they are a real person or an AI, and quickly form personality impressions from it. The present study aimed to investigate how acoustic characteristics of speech of human agents affect their personality impressions. To this end, spontaneous speech was elicited from 30 native Korean speakers while they played a role of an agent assisting a customer on the phone. We then conducted a personality rating experiment in which 30 native speakers of Korean rated the perceived degree of extraversion, agreeableness, conscientiousness, neuroticism, and openness (the “Big Five” personality traits) on the recordings on a 5-point scale. Various acoustic features were extracted from the stimuli, which were converted into two principal components using a Principal Component Analysis. The results of correlation analyses demonstrated that some of the acoustic variables such as measures of voice quality (H1-H2, Harmonic-to-Noise Ratio), the mean f0, and articulation rate were significantly correlated with the personality ratings, with the acoustic correlates different for male and female speakers. This helps discover characteristics of a more likable voice and speaking style for a customer service agent and has important implications for generating more likable synthetic speech for similar services.

1pSC21. Investigating the nature of speech representations by contrasting computational models against behavioral data. Maryann Tan (Ctr. for Res. on Bilingualism, Univ. of Stockholm, Universitetsvägen 10 D, Hus D, Plan 3–6, Södra huset, Stockholm 10691, Sweden, maryann.tan@biling.su.se) and T. Florian Jaeger (Brain and Cognit. Sci.; Comput. Sci., Univ. of Rochester, Rochester, NY)

How do listeners interpret speech input *when exposure does not contain information about the talker's category representations?* Competing accounts are rarely contrasted. We implement competing hypotheses within the same general computational framework (Bayesian inference). All models were trained on phonetic *productions* to predict *perception*, reflecting the hypothesis that listeners learn category representations from the speech input. We compare them against two experiments on the perception of L1-US English stop voicing ($N = 24$ and 122). Both experiments used minimal pairs (e.g., tin/din) varying in the primary cue (VOT) while keeping secondary cues (f0 and vowel duration) at expected correlations. VOT values occurred equally often and spanned the range observed in US English. We find that (1) models that integrated perceptual noise performed better in predicting categorization responses than those without; (2) models with multiple cues performed better than those with just VOT; (3) models trained on talker-normalized phonetic cues performed better than those trained on unnormalized cues; and, surprisingly, (4) models that also normalized the novel speech input during the experiment performed *worse* than those that did not. (3) and (4) suggest that listeners' long-term representations are based on talker-normalized cues but require *labelled* input—contrary to most normalization accounts.

1pSC22. What allows listeners to rapidly adapt to an unfamiliar talker? Evidence from a novel computationally-guided paradigm. Maryann Tan (Ctr. for Res. on Bilingualism, Univ. of Stockholm, Universitetsvägen 10 D, Hus D, plan 3–6, Södra huset, Stockholm 10691, Sweden, maryann.tan@biling.su.se), Rachel Sabatello, Iva Savic (Brain and Cognit. Sci., Univ. of Rochester, Rochester, NY), and T. Florian Jaeger (Brain and Cognit. Sci. and Comput. Sci., Univ. of Rochester, Rochester, NY)

Adaptivity in response to talkers with unexpected pronunciations is now understood to be central to speech perception. What mechanisms afford this ability, however, remains unclear. We present first results from a novel paradigm to investigate incremental changes in perception that occur with exposure to an unfamiliar talker. The paradigm is informed by a computational framework and provides high power to detect previously undocumented constraints on adaptive speech perception. L1-US English listeners ($N = 122$) were randomly assigned to one of three conditions, each corresponding to a different ‘accent’. Accents were identical except that the VOT distributions for /d/ and /t/ were shifted by either +0msec, +10msec or +40msec relative to typical talkers of US English. We find that (1) listeners' categorization functions shifted proportionally with the input; (2) shifts were rapid, leading to significant changes after 48 observations; but (3) adaptation plateaued after 96 observations at only 40% of the shift in categorization that would have been justified by the statistics of the input. Result (2) and (3) together require revision of standard normalization, distributional learning, and exemplar/episodic accounts. We use computational models to show our results follow if listeners adapt by ‘mixing’ talker representations learned from previously experienced input.

1pSC23. Hesitations in story retelling as a measure of listening effort. Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu), Lian J. Arzbecker, Geoff D. Green, and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Hesitation phenomena in natural speech (e.g., pauses, word repetitions, or phrase repairs) vary as a function of speaker characteristics, with non-native speakers typically hesitating more than native speakers (Gilquin, 2008). Here, we examine whether hesitations can also reflect listeners' mental effort required to comprehend native and non-native speech. We asked native English listeners to retell stories told by either a native or a non-native speaker and analyzed their hesitations using a 6-category taxonomy: filler vocalizations, filler words, filler nonspeech, self-monitoring, uptalk, and point of view switching. These story-retell behavioral data were obtained after recording their brain responses during listening to the stories using functional near-infrared spectroscopy (fNIRS) and speaker-listener interbrain synchrony technique (hyperscanning). We found a higher hesitation rate for retelling of stories produced by a non-native (11.2%) than a native speaker (7.6%), with significant increase in hesitation frequency for the former. The hesitations corroborated brain responses: Listeners had greater difficulty processing discourse in non-native English and the recruitment of additional executive resources delayed comprehension and memory processes. The analysis of hesitations is a promising approach in measuring speaker-listener communicative effort because hesitations increase as uncertainty of interpretation increases, suggesting greater demands on working memory during lexical and semantic operations.

1pSC24. Listeners experienced with backness harmony compensate for coarticulation in F2. Zuheyra Tokac (Linguist, Northwestern Univ., Evanston, IL 60208, zuheyratokac@u.northwestern.edu) and Jennifer Cole (Linguist, Northwestern Univ., Evanston, IL)

A natural consequence of articulation is the partial assimilation of neighboring vowels in vowel-to-vowel coarticulation (e.g., relative fronting of /ɑ/ in [ɑ$_i$Ci]). Previous research suggests that listeners are sensitive to coarticulatory variation and perceptually compensate for coarticulation, although the degree of compensation is impacted by how extensive coarticulation is in a language (e.g., English and Shona: Beddor *et al.*, 2002). This study partially replicates Beddor *et al.* with Turkish listeners ($N = 20$), who presumably have limited experience with coarticulation in F2 due to pervasive backness harmony in Turkish, on their perception of the Turkish front-back vowel pair /ɛ ɑ/. Participants completed a 4IAX discrimination task with coarticulated /ɛ ɑ/ vowels spliced into appropriate and inappropriate coarticulatory contexts, and vowel identification tasks on an /ɛ-ɑ/ continuum in [$_b$ɑb] and [$_{bib}$] contexts as well as in “no context.” Results suggest that Turkish listeners compensate for coarticulation, supporting the idea that compensation for coarticulation is automatic: Appropriate coarticulation impaired vowel discrimination, and there were boundary shifts in vowel identification across all contexts with significantly different front vowel response proportions (back>no context>front). Moreover, vowels presented with no adjacent coarticulation trigger were perceived more categorically, suggesting that coarticulatory contexts induce more continuous vowel perception.

1pSC25. Exploring speech characteristics for automatic pathological voice detection. Martyna Wloszczynska (Faculty of ETI, Gdansk Univ. of Technol., Gdansk, Poland) and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

This paper aims to explore speech characteristics for automatic pathological voice detection. First, the assumptions underlying the conducted experiments are presented, along with examples of databases containing pathological speech. Then, a selection process of features of the speech signal and algorithms used to distinguish between undisturbed and pathological speech are discussed. Two deep models are employed to perform binary classification of the speech signal. Their structure is presented along with the feature space chosen. The results of classifying undisturbed and pathological speech are compared with state-of-the-art literature sources. The accuracy values are comparable to those presented in the literature, i.e., they are within the range of 61%–71% for our research and 64%–98% for the

literature data, depending on the algorithms and databases used. An application is also built to illustrate the speech recognition process. The experiments are summarized with conclusions, and the direction of possible future development of the research performed is given.

1pSC26. The [s → ʃ] sound change in /str-/ contexts: Acoustic and perceptual results. Christine H. Shadle (Haskins Labs., 300 George St Ste. 900, New Haven, CT 06511, shadle@haskins.yale.edu), Laura Koenig (Haskins Labs., New Haven, NY), and Wei-Rong Chen (Haskins Labs., New Haven, CT)

Since 1995 [Shapiro, M. A case of distant assimilation: /str/ → /ʃ tr/. *American Speech, 70*, 101–107], many studies have noted a sound change occurring in some dialects of English with /s/ in the context /strV/ surfacing as [ʃ trV]. At first the phenomenon was restricted to certain contexts, though studies disagree on which; increasingly, those who produce the change do so in all contexts. Is the fricative is truly identical to / ʃ / as produced by the same speaker? Or is it acoustically on a continuum, though perhaps perceived as one category or the other? In a corpus of real words, speaker W1 was perceived as exhibiting the sound change, and her /s/ spectra in /str/ context were distinctly different from her /s/ spectra in other contexts. However, speaker M1, though not perceived as exhibiting the sound change, showed spectral differences similar to W1's. Here, acoustic measures that we have developed to characterize the [s → ʃ] difference will be applied to a more extensive corpus, and will also be related to results of formal perception tasks for multiple listeners. It is anticipated that a continuum will emerge, with perceptual classification into categories resulting in the reported sound change.

1pSC27. Perception of talker height in children's productions of sustained /ɑ/. Abbey L. Thomas (Dept. of Speech, Lang., & Hearing Sci., The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, abbey.thomas@utdallas.edu), Santiago Barreda (Linguist, UC Davis, Davis, CA), and Peter F. Assmann (Dept. of Psych., The Univ. of Texas at Dallas, Richardson, TX)

Children's voices reflect changes in physical size as they grow. At a previous ASA meeting, we showed that adult listeners can estimate the height of child talkers, ages 5–18 years, from /hVd/ syllables. The present study focuses on listener judgments of child talker height from a sustained /ɑ/ vowel, which lacks durational cues and transitions between phonological segments. The stimuli for this experiment were 1-s excerpts with stable amplitude and fundamental frequency (F0) from sustained /ɑ/ tokens by 174 talkers (including those from the /hVd/ experiment). 18 adult listeners judged talker sex and height for each token. Judgments of each talker's height were consistent across the two experiments. Perceived talker height was moderately correlated with actual height ($r = .66$). When talker sex was identified incorrectly, height was underestimated for taller females and overestimated for males. Consistent with our previous study, significant relationships were observed between listener judgments of talker height and acoustic measures (geometric mean of the lowest three formants, F0, and additional variables related to the voicing source). The results extend findings from the previous study, showing that the relationship between summary acoustic measures and perception of talker height is preserved when the stimuli comprise only sustained /ɑ/.

1pSC28. Comparing online versus in-person auditory experiments on talker adaptation. Lingyu Zi (Psych., Binghamton Univ., Binghamton, NY) and Sung-Joo Lim (Psych., Binghamton Univ., 4400 Vestal Parkway E, Psych. (Sci. 4), Binghamton, NY 13902, sungjoo@binghamton.edu)

Online experiments are increasingly used as an alternative to in-person lab-based experiments. While online experimental platforms offer convenience, conducting auditory experiments online also presents several challenges, including variations in latency and jitter in auditory stimulus presentation across participants' devices. Therefore, it is important to carefully examine whether response time data collected online are comparable to the reliability and precision of data collected in well-controlled laboratory settings. Here, we directly compared whether well-documented context- and talker-adaptation effects on response times are replicable between the laboratory and an online platform. Participants ($N = 58$) performed an established word identification task measuring response time in a 2×2 manipulation of context (isolated words versus continuous speech) and talker variability (single versus mixed talkers). The results from the laboratory session, but not the online platform, reliably replicated all main and interaction effects found in prior laboratory-based studies. While mean response times were comparable between environments, the variance of online data was significantly greater than in the lab, even after discarding outliers. These results suggest that careful considerations are needed when implementing online experiments that depend on precise auditory timing, particularly in considering how greater measurement error will impact power, sample sizes, and statistical inferences.

1pSC29. Effects of sleep on learning morphophonological alternations. Shiloh Drake (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, sdrake@uoregon.edu), Isabel Preligera, and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR)

In this experiment, we examine the contribution of sleep-based memory consolidation to the learning of morphophonological alternations in an artificial grammar. Periods of sleep after training seem to help participants to retain word formation patterns that they have previously learned, and also generalize learned word formation patterns to novel stimuli (Bryant *et al.*, 2020; Dumay & Gaskell, 2007, 2012; Gomez, 2017; Gómez & Edgin, 2015; Sandoval *et al.*, 2017; Simon *et al.*, 2017). As our previous work has shown, Arabiclike morphology is particularly difficult for English-speaking participants to learn and retain, despite L1 transfer (Drake, under review), varying the type of instructions provided (Drake *et al.*, 2022a), and multi-talker training (Drake *et al.*, 2022b). In this ongoing work, we, therefore, compare participants' accuracy rates after three periods of delay (no delay, delay without sleep, delay with sleep) to find out whether sleep aids in learning, generalizing, and retaining abstract morphological structure, and whether accuracy differs for the Englishlike or Arabiclike morphophonological alternations learned.

1pSC30. Predictive mechanisms in adaptive chunking during rapid extraction of meaning from spoken signals. Benjamin Pittman-Polletta (Boston Univ., Boston, MA) and Laura Dilley (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ldilley@msu.edu)

Speech understanding requires rapid of categorizations of auditory inputs. Neuronal rhythms play key roles in these computations; however, the principles by which timing, acoustics, and context are combined on-line to accomplish speech perception remain murky. Here, we outline a model of how temporal constraints sculpt meaning-making in speech processing according to an adaptively quasi-rhythmic process driven by fluctuations in certainty and predictability of timing and content. We propose that predictive mechanisms are leveraged to reduce possible identities of inputs based on prior inputs in which the smallest chunks of input can be rapidly compressed, recoded, and passed to higher representational linguistic levels. Evidence accumulation at each level of representation is used to assess likelihoods of candidate interpretations and their reliabilities at different timescales in a manner pegged to speech rate and situation-specific speed-accuracy tradeoffs. This synthesis illuminates mechanisms of human speech processing while making predictions for neuronal implementations and behavioral psychophysics.

Session 1pSP

Signal Processing in Acoustics, Biomedical Acoustics, Architectural Acoustics, Physical Acoustics, and Underwater Acoustics: Acoustical Imaging, Reconstruction, and Localization II: General Applications II (Hybrid Session)

Yangfan Liu, Cochair
Purdue Univ., Ray W. Herrick Laboratories, Purdue University, 177, South Russell Street, West Lafayette, IN 47907-2099

Efren Fernandez Grande, Cochair
Technical University of Denmark, Lyngby DK 2800, Denmark

Chair's Introduction—1:00

Contributed Papers

1:05

1pSP1. Direct acoustic velocity vector measurements for evaluation of multichannel sound reproduction systems with signals of varied partial coherence. Erin Driscoll (Elec. and Comput. Eng., Univ. of Rochester, 120 Trustee Rd., Comput. Studies Bldg., Rochester, NY 14620, edrisco5@ur.rochester.edu), Mark Bocko, and Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

In multichannel sound reproduction systems, the perceived virtual auditory scene is strongly influenced by inter-channel phase-shifts and the degree of mutual coherence among channel signals. Such degrees of freedom may be employed to effect an expanded virtual sound stage or a sense of spaciousness, but they also may lead to unwanted spectral coloration, diffuse or ambiguous virtual source locations, or startling and disorienting spatial image perceptions often described as "phaseyness." Such effects may be described and understood in terms of the acoustic velocity in the vicinity of the listener. In this work we present theoretical predictions of the acoustic velocity vector for a stereo reproduction system with channel signals of varying phase and levels of mutual coherence, and we report direct measurements of the acoustic velocity vector employing a 3D sound intensity probe for several specific cases. Acoustic velocity vector predictions and measurements also are correlated with listener perceptions. The presented measurements are also compared to subjective localization of sound sources. The direct measurement of the acoustic velocity vector in spatial sound reproduction systems is a useful tool to quantitatively evaluate the performance of multi-channel spatial audio reproduction systems, to predict listener perceptions of the acoustic field reproduced by such systems, and to investigate the impact of early reflections in a room on localization.

1:20

1pSP2. Experimental demonstration of magnitude-only bearing estimation. Christopher Gravelle (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, chriswgravelle@gmail.com) and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

Traditional bearing estimation systems exploit relative time delays across an array of sensors to localize a target. While highly effective, such systems are costly and cumbersome. Yovel *et al.* (Science, 2010) found that bats steer their echolocation beams off axis to maximize the Fisher information, implying that spectral magnitude cues improve target localization. This presentation shares experiments demonstrating a man-made system estimating a target's bearing from echo spectrum magnitude-information using a single directional sensor with frequency dependence over the bandwidth of the signal. The output of a 31-element microphone array steered to broadside serves as the frequency dependent directional sensor. Moving a source transmitting a linear FM chirp across a range of angles estimates the beampattern of the receiver. Maximum likelihood estimation using these recorded replicas gives the mean-squared error (MSE) as a function of angle. The MSE is compared with the Cramer-Rao lower bound. Moving away from broadside reduces the received SNR, but paradoxically the MSE exhibits local minima. These MSE local minima are consistent with the optimal angles observed in previous research simulating the impact of magnitude cues (Kloepper *et al.*, JASA-EL, 2018; Tidwell *et al.*, IEEE SSPD, 2019). [Work supported by ONR MURI Program.]

1:35

1pSP3. Direction of arrival estimation using two-dimensional microphone arrays. Curtis L. Garner (Brigham Young Univ., EB B120, Provo, UT 84602, curtislgarner@gmail.com), Jonathan D. Blotter, and Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT)

Many signal processing applications require knowledge of where sources are located relative to the receiver. In cases where the source locations are not known in advance, methods must be implemented to estimate the direction of arrival (DOA) of the incoming signals. This paper proposes a method for estimating the DOA of acoustic signals in real time using a two-dimensional array of microphones. While many existing methods focus solely on obtaining an accurate estimate of one or more sound sources, the proposed method aims to provide a more complete picture of the sound field measured by the microphone array. This method can be easily adapted to identify and track any number of independent sources without a significant increase in either complexity or computational expense, making it suitable for applications where very little about the sound field can be known in advance.

Invited Papers

1:50

1pSP4. Methods for high-performance jet aircraft noise source imaging and sound field reconstruction. Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, lmathew3@byu.edu), Tyce Olaveson (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Jon P. Johnson (Phys., Brigham Young Univ. Idaho, Rexburg, ID), and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Source characterization and sound field reconstruction are vital to understanding noise radiation from turbulent, heated, supersonic jets produced by high-performance engines. This paper describes the application of two acoustic imaging methods to full-scale tactical aircraft noise analyses: statistically optimized near-field acoustical holography and hybrid beamforming. These methods constitute equivalent wave and equivalent source models, respectively. A review of these methods is given and their respective features are discussed.

2:10

1pSP5. Super resolution focusing of complex sources above a two-dimensional array of resonators in a three-dimensional environment. Adam D. Kingsley (Dept. of Phys. and Astronomy, Brigham Young Univ. - Provo, Brigham Young University, Provo, UT 84602, adamkingsley@gmail.com), Andrew Basham, and Brian E. Anderson (Dept. of Phys. and Astronomy, Brigham Young Univ. - Provo, Provo, UT)

Time reversal focusing above an array of resonators creates subwavelength features when compared to waves in free space. Previous work has shown the ability to focus acoustic waves near the resonators with and without time reversal with an array placed coplanar with acoustic sources. In this work, a two-dimensional array of resonators is studied with a full three-dimensional aperture of waves in a reverberation chamber. The full impulse response is recorded (including reverberation), and the spatial inverse filter is used to produce a focus among the resonators. Additionally, complex images are produced by extending the spatial inverse filter to create focal images such as dipoles and quadrupoles. Although waves at oblique angles would be expected to degrade the focal quality, it is shown that complex focal images can still be achieved.

2:30–2:40 Break

Contributed Papers

2:40

1pSP6. Tracking array deformations using segmented least squares. Kanad Sarkar (Univ. of Illinois at Urbana Champaign, B10 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801-2447, kanads2@illinois.edu), Jae Won Choi (Univ. of Illinois Urbana-Champaign, Urbana, IL), Manan Mittal (Elec. and Comput. Eng., Univ. of Illinois, Urbana-Champaign, Urbana, IL), Ryan M. Corey (Discovery Partners Inst., Chicago, IL), and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Recording on moving devices leads to constant deformations in microphone array structure, which interferes with spatial audio processing. Previously, we have established that there is a locally linear mapping between the manifolds of relative transfer functions (RTF) and array geometries. This mapping also applies to trajectories of array deformations and their corresponding RTFs. We run segmented least squares (SLS) on the RTF trajectories to discern potential states of motion in an unsupervised manner. We then examine an implementation of SLS for adaptive beamforming in a deformable setting.

2:55

1pSP7. Spatial sigma-delta modulation for coarsely quantized linear, circular, and spherical arrays. Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, 2021 Discovery Partners Inst., 200 S. Wacker Dr., 20th Fl., Chicago, IL 60606, corey1@uillinois.edu) and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Quantization noise shaping, also known as sigma-delta modulation, is widely used in analog-to-digital converters. Such converters oversample a signal in time, quantize it to one or a few bits of precision, and then use analog feedback to shape quantization noise outside the band of interest, producing a high-precision output. Similarly, if a sensor array is spatially oversampled so that its elements are much less than one-half wavelength apart, then quantization noise can be propagated between channels to shape its spatial distribution. A beamformer acts as a spatial filter, removing shaped quantization noise to produce a high-precision output even with coarsely quantized sensors. Spatial sigma-delta modulation has been explored primarily for radio-frequency applications using linear arrays. In this presentation, we apply spatial noise shaping to dense acoustic sensor arrays, including circular and spherical arrays. For immersive audio applications, we demonstrate how noise-shaping feedback can be used to push quantization noise into higher-order Ambisonic modes.

3:10

1pSP8. Uncertainty quantification of deep learning based direction-of-arrival estimation with conformal prediction. Ishan D. Khurjekar (Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, La Jolla, CA 92037, ikhurjekar@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Direction-of-arrival (DOA) estimation is an important task in array signal processing with applications in underwater acoustics, biomedicine, geophysics, and robotics. The signals recorded on a spatially distributed array along with an analytical model of wave propagation, are used to estimate the unknown DOA. Recently, end-to-end deep learning-based methods have been proposed for improving DOA estimation performance. Yet, real-time DOA estimation has a number of challenges such as sensor noise, reverberant surroundings, uncertainty in sensor locations and environment. In this work, we propose to use conformal prediction for uncertainty quantification in DOA estimation. Conformal prediction is a statistically rigorous method to provide confidence intervals for an estimated quantity without making distributional assumptions. With conformal prediction, confidence intervals are computed via quantiles of user-defined score values. This easy-to-use method can be applied to any trained classification/regression model as long as an appropriate score function is chosen. The proposed approach shows potential to enhance the real-time applicability of deep learning methods for

DOA estimation. We illustrate the advantages of conformal prediction for different deep-learning methods for DOA estimation. To the best of our knowledge, this is the first analysis of using conformal prediction for uncertainty quantification of deep learning methods for acoustic estimation.

3:25

1pSP9. Spatial location from running binaural signals using cepstral analysis and logarithmic filtering. Jeramey Tyler (Cognit. Sci., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, tylerj2@rpi.edu), Mei Si (Cognit. Sci., Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

This paper proposes an acoustic model for predicting the acoustical room characteristics from a running binaural signal. This is accomplished via training a convolutional neural network on a precedence effect model to extract the spatial locations of the direct sound source and its early reflections. The precedence effect model extends and modifies the BICAM algorithm with cepstral analysis [Tyler, J., Si, M., \& Braasch, J. Acoust. Soc. Am. 151] and a logarithmic filter. The logarithmic filter takes human perception into account and provides better separation at higher frequencies. A synthetic dataset of binaural signals was generated using anechoic orchestral recordings with added reflections and reverberations. The binaural model generates binaural activity maps from binaural input signals, which are then used to train a convolutional neural network. The ability to predict the traits of a direct sound source and its reflections has applications in academic areas like perceptual modeling and room acoustical analysis. It can also be applied to industrial areas such as television and movies, video games, and augmented and virtual reality, to name a few. [Work supported by the National Science Foundation: HCC-1909229.]

MONDAY AFTERNOON, 8 MAY 2023 MICHIGAN/MICHIGAN STATE, 1:00 P.M. TO 3:05 P.M.

Session 1pUW

Underwater Acoustics, Acoustical Oceanography, and Physical Acoustics: Exploring Fine-Grained Sediments in the Variable Ocean II (Hybrid Session)

David P. Knobles, Cochair
Physics, Knobles Scientific and Analysis, 5416 Tortuga Trail, Austin, TX 78731

Tracianne B. Neilsen, Cochair
Physics and Astronomy, Brigham Young University, N251 ESC, Provo, UT 84602

Preston S. Wilson, Cochair
Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 East Dean Keeton Street, Mail Stop: C2200, Austin, TX 78712-0292

Invited Papers

1:00

1pUW1. Physics-based acoustic inversion of sound velocity and attenuation in low-velocity marine sediments. Ji-Xun Zhou (School of Mech. Eng., Georgia Inst. of Technol., 3186 Amesbury Way, Duluth, GA 30096, jixun.zhou@me.gatech.edu), Zhenglin Li (School of Ocean Eng. and Technol., Sun Yat-sen Univ., Zhuhai, China), xuezhen zhang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Jixing Qin (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Using data from the Yellow Sea, arrival times of the direct wave and surface/bottom reflections from explosive sources to a vertical hydrophone array are used to precisely determine each explosive source's location, the source energy spectral density (SESD), and the water depth. Long-range propagation waveforms reveal modal dispersion: the ground wave, water wave, Airy phase, etc. There are two high-frequency (HF) groups of water waves. One propagates with the sound speed in the water below the thermocline, the other with a speed close to the sound speed in the water above the thermocline. The HF group arrival times offer a time reference for dispersion analyses, including the ground wave speed at the cutoff frequency and the group velocity at Airy frequency. Associated with a top layer of low-velocity sediments (LVS), seafloor reflections have two pulses: one from the water-sediment interface, one from the sediment-basement interface; Long-range transmission loss (TL) exhibits abnormal peaks at selected frequencies. Above-mentioned physics characteristics and the SESD-normalized TL(r) and TL(f) in 50–5000 Hz range up to 27.6 km are used for observationally driven inferences of seafloor geo-acoustic parameters, such as the sound velocity and attenuation in the LVS layer, its thickness, the sound velocity in the basement, etc.

1:20

1pUW2. Investigation of surficial sediment acoustic, physical, and biological properties in the New England Mud Patch. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan Ballard, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), W. Cyrus Clemo, Madeline R. Frey, Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Measurements of surficial sediment properties at five locations in the New England Mud Patch (NEMP) were conducted during a coring survey in April 2022. At each location the Acoustic-Multi-Corer (AMC) sampled the upper few decimeters of sediment near the water-seabed interface. The AMC inserted acoustic probes into the sediment to collect *in situ* measurements of compressional waves (30–140 kHz) and shear waves (0.8–1.2 kHz) at near ambient conditions, after which cores were collected from the inter-probe propagation paths for *ex situ* analysis of the sediment properties. Onboard the ship, the cores were scanned with a Core and Resonance Logger (CARL), which provided high-frequency sound speed and attenuation measurements (100–1000 kHz) along the lengths of the cores, as well as resonance-based measurements of sound speed at 12–16 kHz. A subset of cores was sieved for infauna onboard the vessel, and the remaining cores were later analyzed in an onshore laboratory for porosity, bulk density, mineral and organic matter composition, and grain size. The frequency dependence of the surficial sediment acoustic properties will be discussed in the context of the mud's physical and biological properties as well as sediment acoustics models. [Work sponsored by ONR.]

Contributed Papers

1:40

1pUW3. Effects of weak shear rigidity of the seabed on sound propagation in a range-dependent ocean. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

The shear wave speed is often small compared to the compressional wave speed in unconsolidated marine sediments. Sediment stratification and especially mass density variation in the seabed enhance the effects of weak shear on acoustic normal modes in range-independent ocean and produce significant shear wave contributions to mode attenuation in shallow water [O. A. Godin, J. Acoust. Soc. Am., **149**, 3586–3598 (2021)]. A distinctive feature of the shear-induced perturbations in the mode phase speed and attenuation is their rapid variation with sound frequency due to shear wave interference within a sediment layer. This paper investigates theoretically the shear wave-induced perturbations in normal mode phase, travel time, and attenuation when water depth and/or sediment layer thickness vary gradually with range. Frequency dependencies of the adiabatic normal mode travel time and attenuation are found to be highly sensitive to range dependence of the seabed layering. Gradual changes in the sediment layer thickness have an effect similar to frequency averaging or artificially increased shear wave attenuation. Implications of these findings for geoacoustic inversions will be discussed. Contributions of shear rigidity and cross-range gradients of the sediment layer thickness to horizontal refraction of normal modes will be also examined. [Work supported by ONR.]

1:55

1pUW4. Mid-frequency sound propagation over a mud seabed with a compressional sound speed depth gradient. David P. Knobles (Knobles Sci. and Anal., 5416 Tortuga Trail, Austin, TX 78731, dpknobles@kphysics.org), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

During Seabed Characterization Experiment 2017 in the New England Mudpatch under near isospeed conditions of the water column, observations were made of well-defined intensity striations in time-frequency spectrograms in the 1.5–4.0 kHz band. These observations provided the basis for the hypothesis that the compressional sound speed profile for a homogenous fine-grained sediment possessed a gradient in the upper portions of the seabed. This hypothesis was tested with a statistical inference method that extracts probability density functions for parameter values representing an empirical-based nonlinear profile whose origins go back to Hamilton. Inferred was a surface sound speed that is less than the sound speed of the water at the seafloor and a surface sound speed gradient of about 6 1/s. In 2022 the experiment was repeated during non-isospeed conditions of the water column. The results for 2017 and 2022 are compared with the intent of understanding the response of the geoacoustic properties of the upper portions of a fine-grained sediment to changes in the bottom water temperature and salinity of the water column. [Work supported by Office of Naval Research.]

Invited Papers

2:10

1pUW5. *In situ* measurements of compressional and shear wave speed from the New England Mud Patch and Shelf Break using the Acoustic Coring System. Dante D. Garcia (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dantegarcia@utexas.edu), Megan Ballard, Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), Edward F. Braithwaite, and Samuel Griffith (US Naval Res. Lab., Washington, DC)

In situ measurements of geoacoustic properties provide direct characterization of the seabed at near ambient conditions. The Acoustic Coring System (ACS) is a gravity corer equipped with acoustic probes that obtain *in-situ* compressional wave (30–200 kHz) and shear wave (400–1200 Hz) measurements as the corer penetrates the seabed. During the April 2022 R/V Endeavor coring survey, the ACS was deployed at 36 locations within the New England Mud Patch and New England Shelf Break areas. Data from these measurements will be presented to characterize the depth-dependent structure of the geoacoustic seabed properties as well as their spatial variability. The *in-situ* measurements will be interpreted in the context of stratigraphic layering measured by a seismic survey.

Depth-dependent profiles of compressional speed from a subset of these deployments in the NEMP will be compared to profiles previously collected at nearby locations in 2016. *In situ* compressional wave records from both areas will be compared with *ex-situ* sediment core measurements, including data collected from core loggers and laboratory analyses. Finally, preliminary shear speed results will be discussed. [Work sponsored by ONR.]

2:30

1pUW6. Detailed composition and physical properties analyses of surficial and sub-surface fined-grained sediments from the Continental Shelf and Upper Slope Sediments Offshore Southern New England, USA. Jason Chaytor (U.S. Geological Survey, Woods Hole Coastal andMarine Sci. Ctr., Woods Hole, MA 02532, jchaytor@usgs.gov), Megan Ballard (Univ. of Texas at Austin, Austin, TX), Brian Buczkowski (U.S. Geological Survey, Woods Hole, MA), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (ARL:UT, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

In recent years, detailed geological characterization of the seafloor and shallow sub-surface on the northern U.S. Atlantic margin have been an important component of multi-disciplinary ocean acoustics and environmental sensing projects to support experiment planning and inform geoacoustic models of sound propagation. Comprehensive surficial sediment and core sampling across shallow continental shelf, shelf-edge, and upper continental slope sites offshore southern New England provide the opportunity to evaluate the importance of many sediment characteristics that impact sound propagation through the seabed. Beginning in the mid-shelf New England Mud Patch and continuing into deeper water, we have begun to develop a detailed understanding of the composition and physical properties of the shallow sediment column record using a combination of surficial and sub-surface sediment sampling tools, coupled with an extensive suite of discrete laboratory analyses. Quantitative data on sediment density, porosity, mineral and biogenic composition, and grain size distribution provides both valuable input parameters for geoacoustic models and helps establish a framework for interpreting independently derived measurements and modeling results. The physical, biological, and other environmental sediment characteristics can be integrated with existing geological assessments of the region and used to investigate ocean and sub-surface acoustic propagation. [Work supported by the Office of Naval Research.]

Contributed Paper

2:50

1pUW7. A granular physics-based model for sediment dispersion. Abe Clark (Phys., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, abe.clark@nps.edu), Derek Olson (Oceanogr., Naval Postgrad. School, Monterey, CA), Andrew Swartz (Phys., Naval Postgrad. School, Monterey, CA), and Sobing Phua (Oceanogr., Naval Postgrad. School, Monterey, CA)

Current models of dispersion in marine sediments include losses due to fluid viscosity and shearing at grain-grain contacts. An additional loss mechanism, which is well established in granular physics but is not included in existing models, is the inelasticity of normal compression at grain-grain contacts. Additionally, force-bearing contact networks, often called "force chains," involve primarily normal compressive forces. These "force chains" are known to play a dominant role in force transmission in granular materials, including wave propagation. Using theoretical analysis of a 1D model "force chain" as well as DEM simulations in higher spatial dimension, we show that this granular mechanics perspective, where forces are transmitted along lossy force chains, may be able to explain salient features of the acoustic properties of marine sediments. In particular, the low and high-frequency behavior of the attenuation coefficient match a large collection of experimental data. [This was supported by the Office of Naval Research.]

Session 1eID

Interdisciplinary: Keynote Lecture

Barbara G. Shinn-Cunningham, Chair
Neuroscience Inst., Carnegie Mellon Univ., Pittsburgh PA 15213

Chair's Introduction—4:00

Invited Paper

4:05

1eID. Clinical research in acoustics: Reflections on a journey from bench to bedside and back again. Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., 3181 Sam Jackson Park Rd., Portland, OR 97239, gallunf@ohsu.edu)

Thirty years ago, the author entered the field of acoustical research, curious as to how the human mind processes music. That search quickly led to the field of psychoacoustics and explorations of how basic sound elements are processed, which was presented at an ASA meeting in 1996. In 1999, the author had a benign tumor removed from his auditory nerve, leaving him deaf in his right ear. Over the next few years this led to his increasing interest in how people with altered hearing process sound. This presentation will describe the journey into clinical research that resulted, the amazing collaborative community that exists in hearing research, both on the clinical and basic sides, and how important it is to form connections among researchers and clinicians in order to fully explore the questions of how humans understand sound.

Exhibit

An instrument and equipment exhibition will be located in Chicago Ballroom D/E on the 5th floor.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Monday, 8 May, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including a complimentary beverage.

Tuesday, 9 May, 9:00 a.m. to 5:00 p.m.: Exhibit Open Hours including a.m. and p.m. breaks serving coffee and soft drinks.

Wednesday, 10 May, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. coffee break.

Session 2aAA

Architectural Acoustics, Noise, Structural Acoustics and Vibration, and ASA Committee on Standards: Classroom Acoustics I (Hybrid Session)

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

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

Lily M. Wang, Cochair
*Durham School of Architectural Engineering and Construction,
University of Nebraska - Lincoln, Omaha, NE 68182*

Chair's Introduction—8:00

Invited Papers

8:10

2aAA1. Classroom acoustics: Bridging the gap between standards and application. Coralie van Reenen (The Council for Sci. and Industrial Res., Meiring Naudé Rd., Pretoria, South Africa, CvReenen@csir.co.za) and David Manley (DLR Group, Omaha, NE)

Many countries have developed standards to regulate the acoustic conditions in classrooms. Most standards agree that the ideal ambient noise level should be between 30 and 40 dB, a reverberation time of 0.4 to 0.7 s, depending on the age and specific needs of the students and the type of activities taking place. Some standards include noise transmission levels to reduce unwanted noise transfer between adjacent spaces. Despite these standards, studies continue to show that many classrooms do not have acoustic qualities that are conducive to good learning outcomes. In most counties, the acoustic standards are not compulsory standards and good classroom acoustics is treated as a "nice to have" rather than as a key to good learning outcomes. This paper provides an overview of the classroom acoustics standards in several countries across the globe, considering how widely the standards are applied and identifying key concerns or gaps in implementation. The objective is to seek examples of successful implementation of standards, whether compulsory or voluntary, and to recommend mechanisms or interventions that could support the uptake and mainstreaming of good acoustic design for classrooms.

8:30

2aAA2. Classroom design, the architect, and teachers perspectives. Vanessa Schutte (DLR Group, 6457 Frances St., Ste. 200, Omaha, NE 68106, vschutte@dlrgroup.com) and Dr. Marilyn Denison (DLR Group, Dallas, TX)

How do we design classrooms? The use and needs of modern learning environments have changed dramatically compared to the classrooms of 40 years ago. The fundamental reconsideration of the classroom environment is a direct result of teachers, school administrators, and architects coming to the realization that we all learn differently. The modern classroom is not just about four walls, it is about bringing diverse groups together. The ability to adapt to a variety of activities that support diverse learning styles is critical to today's learning environments. Research proves that the design of a learning environment can have a negative or a positive impact on a student's learning progress. We must understand how to design not only a simple classroom, but a suite of spaces that serve all student needs throughout the day. This presentations reviews the modern process and approach to classroom design from a broad-use perspective.

Contributed Papers

8:50

2aAA3. Classroom mock-up meeting Standard S12.60. Stephen J. Lind (LindAcoustics LLC, 1108 Valley Vue Dr., Onalaska, WI 54650, stephen.j.lind.ut88@gmail.com)

The Acoustical Society of America (ASA) developed a standard entitled Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (ASA/ANSI S12.60-2010) that provides a minimum set of requirements to help school planners and designers "provide good acoustical characteristics for classrooms and other learning spaces in which speech communication is an important part of the learning process." In this revision, several details were revised for incorporation into building codes. Despite the obvious benefits of quieter classrooms, incorporating the standard into building codes met with resistance. A common perception was that in order to meet the standard's requirements, specialized HVAC equipment and installation methods would be necessary, resulting in an installed cost beyond what most schools could afford. Acoustical prediction tools indicated that the requirements could be met using standard HVAC equipment and installation methods. To verify the predictions, a simulated classroom was built and several common types of equipment including a single-zone air handling unit, a high-efficiency water-source heat pump, and a packaged rooftop unit were tested. The selected products were standard catalog offerings, without any special attenuation features, that were operated at typical airflow and static pressures. This paper describes those tests, the conclusion, and the resulting recommendations.

9:05

2aAA4. Sound Transmission Guidelines in Mass Timber Schools; Meeting ASA/ANSI S12.60-2010. Aedan Callaghan (Pliteq, Inc., 4211 Yonge St., Ste. 400, Ste. 404, Toronto, ON M2P 2A9, Canada, acallaghan@pliteq.com)

In recent years, the education sector has begun to turn to mass timber for new construction and expansions of schools. There are several reasons that make mass timber well suited for the education sector with seismic resistance and faster construction timelines being two major drivers. ANSI/ ASA S12.6 Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools is often used as the standard for school design and adequately addressing sound transmission between learning spaces. This analysis looks at mass timber assemblies that meet the ANSI/ ASA S12.60 guidelines. ASTM E336 and ASTM E1007 testing post construction was completed to evaluate *in situ* performance for comparison to the design guidelines.

Invited Papers

9:20

2aAA5. In-Progress Updates to ANSI S12.60 Part 1 Pertaining to Building 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. The current progress of the working group's actions regarding modifications to the standard's building envelope sound isolation language will be discussed.

9:40

2aAA6. The Italian standard on classroom acoustics UNI 11532-2:2020 explained through case studies. Dario D'Orazio (Univ. of Bologna, Viale Risorgimento, 2, Bologna 40126, Italy, dario.dorazio@unibo.it)

The UNI 11532-2:2020 is the Italian standard for acoustic environments in schools, comprised of classrooms and ancillary locals like libraries, gyms and canteens. Two years after its release, this talk is aimed to present some early results based on case studies, both on design simulations and *in situ* measurements. Indeed, on the one hand, the UNI 11532-2:2020 takes care of active classrooms asking consultants to consider the acoustic absorption of occupancy to reach the target range of reverberation time in a wide frequency range. It was shown how mixing material typologies becomes crucial to reach these targets. On the other hand, adequate intelligibility is needed in unoccupied conditions too. It showed how the requirements could be reached in small and large environments. A selection of case studies—a primary classroom, a library, and a university lecture hall—aims to show some design solutions to solve issues needed by the UNI 11532-2:2020.

10:00–10:15 Break

Contributed Papers

10:15

2aAA7. The benefits and challenges of the collaborative for high performing schools acoustical performance requirements in Colorado K–12 schools. Matthew Whitney (K2, 4900 Nautilus Ct N, Boulder, CO 80301, matt@k2avt.com) and Kristin Hanna (K2, Boulder, CO)

The Collaborative for High Performing Schools (CHPS) is a national organization that provides technical resources for school design, construction, operations, and maintenance standards. It also provides a verification program, which includes an acoustical performance prerequisite and enhanced acoustical performance credits. This presentation will explore the benefits and challenges of the various acoustical prerequisite and credit requirements, including reverberation time (RT) targets, background noise criteria, and interior and exterior sound isolation, through the design and construction of several projects in Colorado. By comparing the theoretical benefits to the practical applications of these requirements, improvements to constructibility are proposed. Additionally, the financial impact of pursuing a CHPS certification is examined, especially for school districts with limited financial resources.

10:30

2aAA8. Quantifying teachers' response to acoustic environments: a review of several recent studies. Eric J. Hunter (Com Sci. & Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu)

School teachers have an elevated risk of voice problems due to the vocal demands in the workplace. This presentation summarizes the results of several studies investigating voice use in real and simulated classroom settings. The first study used 57 teachers, who were monitored all day for 2 weeks in teaching and nonteaching environments. In the second study of short-term responses to room conditions, the speech of 20 talkers was analyzed in order to evaluate how voice production and effort were affected by speaking style, room acoustics (added background noise, presence of early reflections), and short-term vocal fatigue in a single room. A third study investigated the effects of voice production and perception due to visually concealed acoustic changes (both noise and reverberation) while speaking in a room. In all studies, talkers significantly adjusted their voice production to both dramatic and seemingly small acoustic changes. Both teaching experience and biological sex were factors in participant response differences.

Invited Papers

10:45

2aAA9. Listening in complex acoustic scenarios: Ongoing advances on methodologies and approaches in relation to classrooms. Giuseppina E. Puglisi (Dept. of Energy, Politecnico di Torino, corso DC degli Abruzzi, 24, Torino 10129, Italy, giuseppina.puglisi@polito.it), Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Torino, Italy), and Anna Warzybok (Medizinische Physik, Universität Oldenburg, Oldenburg, Germany)

The acoustic quality of the built environment where we spend our days strongly affects our social life and indirectly our health. Particularly, people with a hearing impairment (usually related to ageing), children with or without learning disorders and non-native speakers, require better room acoustics to enhance speech communication and be included in the challenging multicultural processes. In the specific framework of children in their everyday learning environments, i.e., in classrooms, enhanced speech intelligibility (SI) is necessary. Noise and reverberation in classrooms degrade SI, but the combined effect of reverberation, informational noise and position of target, listener and noise source on SI still needs insights. This work provides an overview on the recent research carried out in the field of SI under complex acoustic scenarios. Results from in-field measurements as well as from advanced predictive models will be presented and discussed. Particularly, measurements were integrated with listening tests based on the Matrix material, since it has been optimized for the Italian language and for many other foreign languages. Concerning the predictions' results, the Binaural Speech Intelligibility Model (BSIM) implemented at the University of Oldenburg was used to validate an accurate method to predict the listeners' ability at the classroom's design stage.

11:05

2aAA10. Sound propagation and strength in small classrooms. Arianna Astolfi (Energy, Politecnico di Torino, Corso DC degli Abruzzi, 24, Torino 10129, Italy, arianna.astolfi@polito.it), Giuseppina E. Puglisi (Dept. of Energy, Politecnico di Torino, Torino, Italy), and Greta Minelli (Energy, Politecnico di Torino, Torino, Italy)

The study involved 29 occupied primary-school classrooms in Italy where speech level (L_S) and Sound Strength total (G_{tot}), early (G_{50}) and late (G_{late}), have been acquired at 1 m from the source and in other three positions along the main axis of the room. The background noise level during the measurements was less than 56 dB(A) and the reverberation time (RT) was between 0.5 s and 1.4 s. A cut-off value of RT was assumed equal to 0.8 s for subdividing the classrooms in two groups of similar size, with lower or equal and higher RT values, respectively. The slopes per double distance of the speech level along the axis, mL_S, is -2 dB/dd for both the groups. The same slope per double distance is obtained for G_{50} of -2.7 dB/dd, while G_{late} shows a slope of -0.6 dB/dd and -0.9 dB/dd for the two groups, respectively. The slope per double distance of G_{tot} is -2.0 dB/dd as in the case of the speech level, mL_s, as expected. Optimal reverberation time range, between 0.6 s and 0.9 s, in occupied conditions, has been obtained from the difference G_{late}-G_{50} at the rear positions between 6 and 3 dB, respectively.

11:25

2aAA11. Age effect on students' intelligibility listening to a teacher with a dysphonic voice. Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu), Silvia Murgia, and Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

The purpose of this project is to assess the acoustical conditions in which optimal intelligibility and low listening effort can be achieved in real classrooms for elementary students, taking into consideration the effects of dysphonic voice and typical classroom noise. Speech intelligibility tests were performed in classrooms with 80 normal-hearing students (7–11 years old). The speech material was produced by a female actor using a normal voice quality and simulating a dysphonic voice. The stimuli were played by a Head and Torso Simulator. Child babble noise and classrooms with different reverberation times were used to obtain a Speech Transmission Index range from 0.2 to 0.7. The results showed a significant decrease in intelligibility when the speaker was dysphonic, in STI higher than 0.33. The rating of listening difficulty showed a significantly greater effort in perceiving the dysphonic voice. Younger children showed poorer performance and greater listening difficulty compared to older children when listening to the normal voice quality. Both groups were equally impacted when the voice was dysphonic. The results suggested that better acoustic conditions are needed for children to reach a good level of intelligibility and to reduce listening effort if the teacher is suffering from voice problems.

Session 2aAB

Animal Bioacoustics, Psychological and Physiological Acoustics and Signal Processing in Acoustics: Session in Honor of James A. Simmons I (Hybrid Session)

Laura Kloepper, Cochair
Department of Biological Sciences, University of New Hampshire, 230 Spaulding Hall, Durham, NH 03824

Alyssa W. Accomando, Cochair
*Biologic and Bioacoustic Research, National Marine Mammal Foundation,
2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106*

Chair's Introduction—8:00

Invited Papers

8:05

2aAB1. Cognitive perception of biosonar echo delay, phase, and spectrum. James A. Simmons (Neurosci., Brown Univ., 195 Meeting St., Providence, RI 02912-9067, James_Simmons@brown.edu) and Andrea M. Simmons (Cognit., Linguistic, & Psychol. Sci., Brown Univ., Providence, RI)

FM echolocating big brown bats combine the acoustic delay, phase, and spectrum of echoes into a unitary cognitive attribute of perceived delay. In this talk, we will present psychophysical, neurophysiological, and modeling data showing how this might be accomplished. Delay accuracy measured psychophysically approximates coherent matched-filter accuracy. Psychophysical curves of echo-delay resemble pulse-echo crosscorrelation functions, with phase manifested directly within 20 μs. Computational modeling shows that low-pass time-frequency bandpass filtering at 10 kHz enables the contribution from phase. Neural responses from the auditory midbrain are phase-sensitive for tone-burst offsets up to15 kHz. Echo spectrum (spectral nulls and ripples) is transformed into psychophysical percepts of delay. The external-ear spectral null creates an elevation-dependent peak at about 35 μs for vertical target tracking. Off-axis clutter masking is suppressed by a Gestalt-like normalization process that operates within delay percepts. The organization of perceived delay and the appearance of phase shifts is facilitated by delay processing that begins at the low-frequency tail end of FM sweeps and the linking of 1st and 2nd harmonic frequencies. Surprisingly, this result requires the entire echo to be received before any delay is determined. The implications of this result will be discussed. [Work supported by ONR.]

8:25

2aAB2. "Let's see what happens!"—His words were and still are guiding me. Shizuko Hiryu (Doshisha Univ., Ishinkan IN505N, Tataramiyakodani 1–3, Kyotanabe 610-0394, Japan, shiryu@mail.doshisha.ac.jp)

The first paper I read when I started studying bats as a student was Jim's paper. At the time, I never dreamed that a few years later we would be able to conduct bat experiments together in Japan and the United States. Although it was my first and last study abroad experience, and it was only for a month, the experience of working with Jim every day guided my confidence and career path as a researcher. I will never forget the excitement when we conducted a flight experiment using a telemetry microphone together and found that the big brown bats were fluctuating the pulse frequency up and down in a cluttered environment. This experience sparked my interest in the jamming avoidance behavior of bats, which has become a very important research topic for me. I have been very fortunate to have had the opportunity to see up close the ideas that come from his vast experience and knowledge and to witness the respectful attitude he has toward bats and nature as a pure researcher. Jim is a great researcher whom I genuinely admire and who taught me to enjoy bats, nature, and research indeed.

8:45

2aAB3. Specializations of bat auditory cortex for processing echoes arriving at different delays. Michael Smotherman (Biology, Texas A&M Univ., 3258 TAMU, College Station, TX 77843-3258, smotherman@tamu.edu) and Silvio Macias (Neurosci., Virginia Tech Univ., Blacksburg, VA)

The bat auditory system follows the standard mammalian plan, but many specializations have been uncovered that appear uniquely tailored to support echolocation. However, it still remains uncertain which, if any, neurophysiological specializations are truly unique to echolocation. The question is important because it defines the extent to which bats may serve as general models of auditory processing, and it points towards which elements are critical for biosonar processing. One of their most unique neurophysiological features is the presence of neurons sensitive to the time delay between the outgoing pulse and returning echo. These delay-tuned neurons are found in the auditory cortex of all bats, but their neuroanatomical distribution is distinctly different in FM-type bats compared to CF-type and

neotropical fruit bats. In FM bats, delay-tuned neurons are sprinkled throughout primary auditory cortex rather than comprising their own auditory subfield, suggesting they might do more than just target ranging. We hypothesized that delay-tuned neurons might dynamically modulate the neural substrate to match changing pulse acoustics for targets at different distances. Results supported the hypothesis by showing that activation of delay-tuned neurons was directly correlated with concomitant changes in the frequency tuning properties of nearby neighboring neurons.

9:05

2aAB4. Dynamic control of the acoustic scene in echolocating bats. Lasse Jakobsen (Biology, Univ. of Southern Denmark, Campusvej 55, Odense M 5230, Denmark, lasse@biology.sdu.dk), Felix T. Häfele, and Danuta M. Wisniewska (Biology, Univ. of Southern Denmark, Odense M, Denmark)

Echolocating bats can detect, localize, classify, pursue, and capture night-flying insects in less than a second. During pursuit, aerial hawking bats dynamically change their emitted calls as they approach prey, increasing call rate, reducing call duration, increasing bandwidth, and reducing source level. Additionally, many bat species dynamically control the shape of the emitted echolocation beam by broadening the beam in close proximity to prey. However, the acoustic scene, as experienced by the bats, depends as much on the characteristics of the emitted call as on the shape and orientation of the outer ears. We show that aerial-hawking bats also dynamically modify the shape and orientation of the outer ear in synchrony with changes to the emitted beam during prey pursuit. We believe that it is the combination of all these dynamic changes, both to the emitted call and to the receiving morphology, that underlie the effectiveness with which bats navigate and forage in the night sky.

9:25

2aAB5. Bats to belugas: Functional anatomy of air versus water biosonar. Darlene R. Ketten (Biology, WHOI, Woods Hole Oceanographic Inst., Neurosciences Brown University, Boston, MA 02543, dketten@whoi.edu), Andrea M. Simmons (Cognit., Linguistic, & Psychol. Sci., Brown Univ., Providence, RI), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Acoustic task similarities of odontocetes (toothed whales) and microchiropterans (insectivorous bats) suggest they have common biosonar mechanisms. However, media differences; e.g., sound speeds in air versus water, may have driven variations in auditory system adaptations. We examined the peripheral auditory system of an FM bat (big brown bat *Eptesicus fuscus*) and two toothed whale species (*harbor porpoise Phocoena phocoena; bottlenose dolphin Tursiops truncatu*) using ultra high resolution (11 μm) isotropic voxel microCT. Significant differences were found among species for oval window location, cochlear length, basilar membrane gradients, cochlear spiral morphometry, cochlear curvature, and basilar membrane stiffness. High and low frequency hearing range cut-offs correlate with basilar membrane thickness/width ratios and cochlear radii of curvature. These features predict species specific high and low frequency hearing limits. Harbor porpoises, the highest frequency echolocator in the study, have large basal basilar membrane ratios and a "foveal" region with a constant membrane ratio similar to those reported for some bat species, consistent with a "stretched" frequency region. Furthermore, both the bats and harbor porpoises examined displayed unusual stapedial input locations and tightly coiled cochleae, specializations that may enhance ultrasonic frequency signal resolution and diminish low frequency cochlear propagation.

9:45

2aAB6. Behavioral hearing thresholds and response latencies in killer whales (*Orcinus orca*) depend on sound duration. Alyssa W. Accomando (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, alyssa.accomando@nmmpfoundation.org), Kayla Nease, Kaitlin Van Alstyne (National Marine Mammal Foundation, San Diego, CA), Todd Robeck (SeaWorld Parks and Entertainment, Orlando, FL), and Brian K. Branstetter (National Marine Mammal Foundation, San Diego, CA)

Large odontocetes are considered at risk of auditory and behavioral impacts from sonar, and it is necessary to predict the effects of tonal sonar signals that have widely varying durations. Here, the effect of sound duration on hearing threshold and reaction time was investigated in two killer whales (*Orcinus orca*) using behavioral methods. Hearing tests were conducted using sounds between 50 μs and 2 s in duration for frequencies across the hearing range. Hearing thresholds within each frequency were dependent on sound duration, with threshold increasing with duration out to a plateau, which was used to calculate auditory integration time. Vocal response reaction time data showed decreasing latency with increasing signal level and length. These data were consistent with other marine mammal species and demonstrate that signal detection and behavior are affected by the duration of tonal sounds in killer whales.

10:05–10:20 Break

Contributed Papers

10:20

2aAB7. Adaptive behavior in sonar prey tracking tasks: Empirical data and modeling. Michael G. T. Wilkinson (Mech. Eng., Johns Hopkins Univ., 3101 Wyman Park Dr., Baltimore, MD 21218, mwilki23@jhu.edu), XingYao Wang (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Noah J. Cowan (Mech. Eng., Johns Hopkins Univ., Baltimore, MD), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Echolocating bats can discriminate small differences in target range (Simmons, 1973) and show a clutter interference zone along the range axis of 8–9 cm (Simmons *et al*., 1988). Because bats use an active sensing system, they can employ various strategies to mitigate the effects of clutter. We studied clutter rejection behaviors of the big brown bat, *Eptesicus fuscus*, as it tracked a moving target under controlled conditions. We also created data-driven models using control theory and system identification techniques to evaluate oscillating target motion tracking strategies. We trained bats to rest on a stationary platform and track a moving tethered mealworm in two conditions (1) approaching and passing by stationary clutter objects at varying distances and angular offsets, and (2) moving in sinusoidal patterns in an open room. Data from the first experimental condition revealed adaptive echolocation call features and head movements that enabled target tracking in the presence of clutter. Under the oscillating target condition, we analyzed adaptive echolocation behavior to create a data-

driven model that maps changes in target motion with sonar call rate. The results of this study demonstrate the bat's use of active motor control that separates targets from clutter and to track complex motion trajectories.

10:35

2aAB8. Studying dolphin biosonar with the jittered-echo paradigm. James Finneran (US Navy Marine Mammal Program, Naval Information Warfare Ctr. Pacific, 53560 Hull St., San Diego, CA 92152, james.j.finneran.civ@us.navy.mil), Jason Mulsow, and Dorian Houser (National Marine Mammal Foundation, San Diego, CA)

In his 1979 paper "Perception of echo phase information in bat sonar" [*Science* **204**, 1336–1338], Jim Simmons introduced the "jittered-echo" paradigm. In this method, bats discriminated between electronic echoes with fixed delay (i.e., simulating fixed range) and those with delays that alternated ("jittered") on successive echoes. The jittered-echo paradigm was developed to minimize the interfering effects of head movement, under the assumption that head movement between successive pulse emissions is negligible. Echo delay thresholds obtained with the jitter technique are small, with multiple studies reporting sub-microsecond echo delay thresholds in the big brown bat. The jitter method has also been controversial, because of the extremely low echo delay resolution (10 ns) and sensitivity to echo phase (fine structure) reported by Simmons. Recently, the jitter delay paradigm has been adapted for underwater use with dolphins. Results with dolphins show qualitative similarities to those from bats: echo delay thresholds below 1 μs and sensitivity to echo fine structure. This talk will briefly review Simmons' and other's biosonar jitter experiments with bats, then present in detail experiments with bottlenose dolphins featuring jittered echo delay and phase.

10:50

2aAB9. Auditory brainstem response recovery from forward masking in insectivorous and frugivorous bats. Grace Capshaw (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 North Charles St., Ames Hall, Rm 235, Baltimore, MD 21218, gcapshaw@jhu.edu), Clarice A. Diebold, Susanne Sterbing (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Amanda M. Lauer (Dept. of Otolaryngology-Head and Neck Surgery, Johns Hopkins Univ. School of Medicine, Balitmore, MD), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Echolocating bats rely on precise temporal processing of sound, as echolocation calls may be emitted at rates as high as 150–200 sounds per second in the terminal "buzz" phase that precedes prey capture by insectivorous species. High call repetition rates could introduce forward masking effects that interfere with echo detection; however, echolocating bats may have evolved auditory specializations to prevent repetition suppression of auditory responses and facilitate detection of sounds separated by very brief time intervals. We assessed the time course of post-stimulus recovery of the auditory brainstem response (ABR) in two bat species that differ in the temporal characteristics of their sonar behaviors: the insectivorous *Eptesicus fuscus*, which uses high sonar call rates to capture prey and the frugivorous *Carollia perspicillata*, which uses lower call rates to forage. We recorded forward-masking ABRs to paired clicks at varied inter-stimulus intervals and measured recovery from prior stimulation as the ratio of response amplitudes and latencies evoked by the second stimulus relative to the first. We observed significant species-specific effects of forward masking on ABR responses in which *E. fuscus* maintained comparable responses to ISIs less than 4 ms compared to longer post-stimulus response recovery times (>6 ms) in *C. perspicillata*.

11:05

2aAB10. Linear time-invariant (LTI) modeling for aerial and underwater acoustics. Jason E. Gaudette (Adv. Technol., Raytheon Technologies, 1847 West Main Rd., Portsmouth, RI 02871, jason.gaudette@rtx.com) and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Most newcomers to acoustic signal processing understand that linear time-invariant (LTI) filters can remove out-of-band noise from time series signals. What many acoustics researchers may not realize is that LTI models can be applied much more broadly, including to non-linear and time-variant systems. This presentation covers an overview of the autoregressive (AR), moving-average (MA), and autoregressive moving-average (ARMA) family of LTI models and their many useful applications in acoustics. Examples include analytic time-frequency processing of multi-component echolocation signals, fractional-delay filtering for acoustic time series simulations, broadband acoustic array beamforming, adaptive filtering for noise cancelation, and system identification for acoustic equalizers (i.e., flattening the frequency response of a source-receiver pair). This talk serves as a brief tutorial and inspiration for researchers who want to expand their use of signal processing, especially those in the fields of animal bioacoustics, aerial acoustics, and underwater acoustics.

11:20

2aAB11. Midbrain inactivation produces sonar navigation deficits in the echolocating bat, *Eptesicus fuscus*. Clarice A. Diebold (Psychol. and Brain Sci., Johns Hopkins Univ., 2812 N Howard St., Baltimore, MD 21218, clarice.diebold@gmail.com), Kathryne Allen, Jennifer Lawlor, and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

The contribution of different brain structures to auditory-guided behaviors in echolocating bats is not fully understood. Historically, area-specific inactivation studies have provided insights into the function of brain regions; however, lesioning is permanent and can cause incidental damage to non-target areas, which can compromise the interpretation of behavioral effects. We have established the use of Designer Receptors Exclusively Activated by Designer Drugs (DREADDs) to reversibly inactivate excitatory neurons in the midbrain of the insectivorous bat, *Eptesicus fuscus*. We targeted the inferior and superior colliculi, two structures implicated in processing returning echoes and coordinating motor responses, including the production of sonar vocalizations. After bilateral infusions in either the IC or SC, bats were trained to fly down a corridor and navigate through an opening in a partition to receive a food reward. Midbrain inactivation dramatically altered performance in the behavioral task. Bats with IC inactivation showed reduced precision in steering through the partition opening, as well as changes in echolocation calls. SC inactivation produced a profound deterioration of the bat's behavior, with the bat unable to fly down the corridor. Our preliminary results open the door to many potential applications of DREADDs to uncover the neural mechanisms supporting navigation through echolocation.

11:35

2aAB12. Echolocating bats dynamically regulate hearing sensitivity: Behavioral and physiological evidence. Huan Ye (School of Life Sci., Central China Normal Univ., 152 Luoyu Rd., Hongshan District, WuHan, HuBei 430079, China, yehuan@mails.ccnu.edu.cn) and Jinhong Luo (School of Life Sci., Central China Normal Univ., Wuhan, China)

James Simmons's previous study on automatic gain control in the big brown bat has suggested the contractions of middle-ear muscle and forward masking impairing hearing sensitivity during vocalization. The influence of vocal production can be extremely severe for bat species that produce constant-frequency (CF) sonar signals. However, perceptual hearing sensitivity and the underlying mechanisms in CF bats remain virtually unexplored. In this work, combing a 2-AFC psychophysical setup and an electrophysiological setup, we measured the hearing sensitivity of a CF bat, *Hipposideros pratti*, either in a passive listening (PL) task to detect pure tones, an active listening (AL) task to detect pure tones triggered by its vocalization, or a phantom Echo task. Behavioral data show that *H. pratti* had the best hearing sensitivity of approximately 0 dB SPL in the PL task, but nearly 40 dB worse in the Echo task. In the AL task, all bats gradually increased call frequency by 0.8 to 1.1 kHz to overcome self-generated auditory masking. Preliminary neurophysiological data indicate that the neural responses in the midbrain inferior colliculus can be highly plastic. Together, our data suggest that echolocating bats dynamically control the hearing sensitivity during vocalization in a complex manner.

Session 2aAO

Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Memorial Session for Jeffrey A. Nystuen I (Hybrid Session)

Jennifer Miksis-Olds, Cochair
University of New Hampshire, 24 Colovos Rd., Durham, NH 03824

Kay L. Gemba, Cochair
Physics Dept., 833 Dyer Road, Bldg 232, NPS, Monterey, CA 93943

Jie Yang, Cochair
Applied Physics Lab, University of Washington, 1015 NE 40th St., Seattle, WA 98105

Chair's Introduction—8:30

Invited Papers

8:35

2aAO1. Dr. Jeffrey A. Nystuen—A PAL to acoustical oceanography. Jennifer Miksis-Olds (Univ. of New Hampshire, Ctr. for Acoust. Res. & Education, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu)

Dr. Jeffrey A. Nystuen was a visionary, innovator, proud student of life-long science learning, and friend. Jeff pioneered the development of the Passive Aquatic Listener (PAL), which was initially the Acoustic Rain Gauge (ARG). The PAL is a low-noise, broadband (100 kHz sampling rate), low-duty cycle, underwater recorder that operates according to an onboard, adaptive sampling protocol. The PAL was one of the first autonomous passive acoustic recorders that could sample for a full year with a frequency range capable of recording dolphin echolocation signals. While originally designed to detect rainfall and estimate the rate of ocean precipitation, the PAL has been used to better understand marine mammal patterns and trends, wind and storms, anthropogenic activity, and sea ice dynamics. The versatility of the PAL has evolved throughout its use from a standalone, archival autonomous recorder to a compact system integrated into Argo floats. PALs have been deployed in the Arctic, Indian, Atlantic, and Pacific Oceans in addition to many regional seas, coastal waters, freshwater lakes, and springs! The vast number of interdisciplinary questions and collaborations centered around the PAL technology highlights Jeff's creative thinking, which has left a strong legacy in the field of acoustic oceanography.

8:55

2aAO2. Listening to the raindrops from underwater for weather and climate. Barry B. Ma (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, barry@apl.uw.edu)

Jeffrey A. Nystuen is a pioneer in using the passive acoustic method to measure oceanic rainfall. Under Nystuen's efforts, a self-contained passive acoustic recording device called the acoustic rain gauges were deployed all over the world, from a brackish pond in Florida, Carr inlet in Puget Sound, Crater Lake in Oregon, BioSphere II in Arizona, then to the South China Sea, the Pacific Ocean, the Taiwan strait, the Ionian Sea in Greece, the Aleutian Islands and ocean station Papa. This simple device records long-term spectra of sound pressure levels with minimum data screening, could detect the presence of oceanic rainfall, and provide estimates of rain rate and surface wind speed. Nystuen's inspirational works over the past three decades set an example and path for future scientists and engineers to foster curiosity, pursue exploration and make discoveries.

9:15

2aAO3. A real-time underwater soundscape classification system. Mojgan Mirzaei Hotkani (Elec. and Comput., Dalhousie Univ., 6299 South St., Halifax, NS B3H 4R2, Canada, m.mirzaei@dal.ca), S. B. Martin (JASCO Appl. Sci., Dartmouth, NS, Canada), and Jean-François Bousquet (Elec. and Comput., Dalhousie Univ., Halifax, NS, Canada)

The study of underwater soundscapes, pioneered by Dr. Jeff Nystuen, includes information about geophonic, biological, and anthropogenic processes, many of which have distinct spectral characteristics. The contributors to the underwater ambient sound field can be quantified and classified using knowledge of these spectra. Long-term acoustic data recordings from a wide variety of depths and locations with high sampling frequencies have been analyzed to develop a robust soundscape classification algorithm based on Dr. Nystuen's methods. Each one-minute of data has been evaluated to classify the soundscape into wind, rainfall, drizzle, heavy shipping, light shipping, other vessel activity, and biological phenomena. The power spectral density (PSD) level at twelve frequencies in the range of 0.03–30 kHz, as well as the spectral slope for the frequency range between 8 and 15kHz and kurtosis are used for the passive classification algorithm. After classification, the wind speed was quantified as a cubic function of PSD at 6 kHz and recording depth. The wind

speed estimated from the acoustics compared very well to satellite data for speeds lower than 15 m/s. The classification algorithms are being embedded on a processor using Xilinx's Zynq System-on-Chip that produces a 32-kHz hybrid millidecade spectrum in real-time on a logarithmic scale.

9:35

2aAO4. Early ocean ambient sound monitoring, precursor to soundscapes today, influenced by J. Nystuen. Bruce M. Howe (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bhowe@hawaii.edu) and Rex Andrew (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In the early 1990s, as the Acoustic Thermometry of Ocean Climate (ATOC) program was starting, simple statistics of ocean ambient sound were unknown. For instance, what fraction of time does the sound level at some frequency exceed a certain value? When the Cold War ended, US Navy SOSUS arrays became available for "dual use" science. At the Applied Physics Laboratory, University of Washington, we began to collect ambient sound data in 1994 from Navy arrays in the Pacific. Initial data spanning 2 years were presented by Curtis *et al*. [*JASA* (1999)]. Data and corresponding analyses were separated according to process, i.e., shipping, marine mammals, and wind, as well as for the total. This monitoring effort continued for nearly two decades as reported by Andrew *et al*. [*JASA* (2011)], with other collaborators involved (Metzger and Mercer). Throughout, J. Nystuen was always providing guidance and advice in the data analysis and interpretation of the ambient sound data, in fact influencing the first author as graduate students together. We review these early results and connect them to present day understanding and data collection efforts that reflect the status of Ocean Sound as an Essential Ocean Variable of the Global Ocean Observing System.

9:55–10:10 Break

10:10

2aAO5. Sperm whale acoustic ecology investigations using Jeffrey Nystuen's Passive Aquatic Listeners. Nikoletta Diogou (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, Victoria, BC V8P 3E6, Canada, niki.diogou@gmail.com), Daniel Palacios (Oregon State Univ., Newport, OR), Stylianos Katsanevakis (Univ. of the Aegean, Mytilini, Greece), and Holger Klinck (Cornell Univ., Ithaka, NY)

When Jeffrey Nystuen invented the Passive Aquatic Listener (PAL) for quantifying rainfall and wind speed using underwater sound spectra, he considered all biological sounds in the recordings as *noise*. Using this *noise* data, I built a doctoral thesis, comprised of three published papers. None of this would have happened without Jeff's continuous encouragement and support. Here, we present results from analyzing (1) five years (2007–2012) of PAL recordings from Ocean Station PAPA (OSP) in the offshore Gulf of Alaska; and (2) 19 months of PAL data from two sites, Pylos and Athos Stations (in the Hellenic Trench and North Aegean Trough respectively), in the Greek Seas, investigating the acoustic ecology of sperm whales (*Physeter macrocephalus*). Results of the bioacoustic analysis revealed the year-round presence of sperm whales at OSP and the Ionian Sea, with higher detections during the warm seasons. The sperm whale time series from OSP was correlated with *in situ* and remotely-sensed oceanographic variables to improve our understanding of global-warming-driven changes in the pelagic ecosystem of the NE Subarctic Pacific. Results from the Hellenic Trench emphasize the risk from increased shipping noise and contribute to the conservation efforts for the small, endangered sperm whale population in an understudied region.

Contributed Papers

10:30

2aAO6. Detection and estimation of rainfall from broadband acoustic signals. C. Mallary (Univ. of Massachusetts Dartmouth, Dartmouth, MA), C. J. Berg (ECE, Univ. of Massachusetts Dartmouth, North Dartmouth, MA), John R. Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu), and Amit Tandon (Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Jeff Nystuen contributed major advances in our understanding of the physical acoustics of rain falling on the ocean, culminating in his pioneering work quantifying rainfall from the shape of the acoustic spectrum. Ma and Nystuen [JAOT (2005)] exploited Vagle's wind spectrum to self-calibrate hydrophones for long deployments as acoustic rain gauges. Their algorithm predominantly relied on 3 narrowband frequencies to detect rainfall, and correlated the rainfall amount with the power spectral density (PSD) at 5 kHz. Recent research at UMass Dartmouth built upon the foundational work of Nystuen, Ma and others to examine how much additional information can be gleaned from broadband acoustic spectra. These algorithms exploit Principal Component Analysis and Linear Discriminant Analysis for rainfall detection, coupled with Error-Correcting Output Codes (ECOC) for quantizing rainfall estimation. Testing on 5 months of 3-minute PSDs from a noisy cove found a detection probability of $78 \pm 5\%$ with a $1.1 \pm 0.3\%$ false alarm rate. Moreover, ECOC-based hourly rainfall estimates achieved 0.97 ± 0.01 correlation with rainfall measurements at a co-located meteorological station. This talk plans to present results from a recent 6 week coastal deployment in deeper water. [Work supported by ONR/UMassD MUST program]

10:45

2aAO7. Summary of underwater ambient sound from wind and rain in the northeast Pacific continental margin. Felix Schwock (Elec. and Comput. Eng., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, fschwock@uw.edu) and Shima Abadi (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA)

Analyzing underwater ambient sound from various sources such as ships, marine mammals, rain, and wind is crucial for characterizing the ocean environment. While efforts to analyze ocean ambient sounds have been ongoing since the 1940s, networks such as the Ocean Observatories Initiative (OOI) provide modern large-scale recording setups for a more in-depth analysis. Here we will summarize results from analyzing over 11,000 h of wind generated ambient sound and 280 h of ambient sound during rain collected between 2015 and 2019 by two OOI hydrophones deployed in the northeast Pacific continental margin. The hydrophones record continuously at depths of 81 and 581 m with a sample rate of 64 kHz. Meteorological data are provided by surface buoys deployed near the hydrophones. We compare our results to data obtained from a large-scale recording setup in the tropical Pacific Ocean (Ma *et al*., 2005). In contrast to their results, we found that sound levels during rain in the northeast Pacific Ocean are highly dependent on the wind speed over a wide frequency range. This implies that large-scale distributed sound measurements are necessary to accurately characterize underwater ambient sound from wind and rain across the globe. [Work supported by ONR.]

Session 2aBAa

Biomedical Acoustics and Physical Acoustics: Clinical Perspective of Biomedical Acoustics I (Hybrid Session)

Tyrone M. Porter, Cochair
Biomedical Engineering, University of Texas at Austin, 107 W Dean Keetom St., TX 78712

Flordeliza S. Villanueva, Cochair
Medicine/Cardiology, University of Pittsburgh, UPMC Presbyterian, Ste. A-351, Pittsburgh, PA 15213

Chair's Introduction—7:40

Invited Papers

7:45

2aBAa1. Neuronavigated focused ultrasound for clinical bbb opening in Alzheimer's and brain cancer patients. Elisa Konofagou (Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, ek2191@columbia.edu)

Treatment of neurological diseases has been at the forefront of medical research for more than a century with limited success due partly to the blood-brain barrier (BBB) that impedes both delivery and biodistribution of the drug delivered. Focused ulltrasound methodologies in conjunction with systemically administered microbubbles have been shown capable of transcranially and transiently opening the BBB over the past two decades. More recently, those efforts have resulted into clinical translation in a variety of brain diseases such as brain tumors and neurodegenerative disease such as Alzheimer's (AD) and Parkinson's disease. Our group has been developing a neuronavigated system for opening the BBB at the patient's bedside safely and efficiently, i.e., within 20–30 min with real-time cavitation mapping. The findings of two clinical studies are reported herein. In the first clinical study, five AD patients underwent FUS with microbubbles with successful BBB opening in their prefrontal cortex while 60% exhibited amyloid reduction in the sonicated hemisphere. In the second clinical study, four diffuse intrinsic pontine glioma (DIPG) subjects underwent BBB opening similar procedure with successful BBB opening in the pons. An overview of the aforementioned findings together with the most recent clinical outcomes will be presented.

8:15

2aBAa2. Gene and drug delivery using ultrasound-targeted microbubble cavitation. Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, UPMC Presbyterian, Ste. A-351, Pittsburgh, PA 15213, villanuevafs@upmc.edu)

Therapeutic delivery of oligonucleotides, such as RNAs, has the potential to treat heretofore "undruggable" targets in diseases such as cancer. However, systemic intravascular delivery of RNAs to tumor tissue is limited by the vascular endothelium, which provides a barrier to transendothelial passage of large, charged molecules. To address RNA delivery challenges, we and others have developed an approach employing ultrasound targeted microbubble cavitation (UTMC), which utilizes 2–4 μm diameter gas-filled microbubbles that carry RNA on the shell and also serve as an ultrasound imaging agent. As intravenously injected microbubbles travel through the microcirculation, localized ultrasound induces microbubble cavitation, causing localized payload release from the microbubble. In addition to delivering the RNA payload, UTMC increases endothelial permeability, thus increasing local extravasation of the payload, while sparing non-insonified tissue (reduced off-target effects). Microbubbles are non-immunogenic and protect loaded RNA against nuclease digestion. Because UTMC concentrates the RNA at the tumor site, lower systemic RNA doses can be used, resulting in fewer off-target effects. A unique feature is that the microbubble that carries and releases the RNA also permits simultaneous US imaging of microbubble location. This lecture will review these unique features of UTMC for drug delivery and current work in this area.

8:45

2aBAa3. Understanding the mechanisms of ultrasound-targeted microbubble cavitation-mediated blood brain barrier opening. Grace E. Conway (Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 969.3, Pittsburgh, PA 15213, gec36@pitt.edu), Anurag N. Paranjape, Xucai Chen, and Flordeliza S. Villanueva (UPMC Heart and Vascular Inst., Univ. of Pittsburgh, Pittsburgh, PA)

Ultrasound-targeted microbubble cavitation (UTMC) transiently opens the blood brain barrier (BBB). We previously determined that UTMC induces BBB hyperpermeability through an influx of calcium. As activation of RhoA is a calcium-dependent pathway that causes cytoskeletal reorganization, leading to the breakdown of tight junctions, we tested the hypothesis that UTMC-induced activation of RhoA leads to BBB hyperpermeability. We utilized a transwell model with brain endothelial cells and astrocytes on opposite sides of a support membrane. Ultrasound (1 MHz, 250 kPa, 10 μs pulse duration, 10 ms pulse interval) was applied in the presence of lipid microbubbles for 20 s. BBB permeability was assessed using dextran flux and transendothelial electrical resistance (TEER) measured across the membrane. Integrity of tight junctions was evaluated by staining for ZO-1. UTMC reduced TEER ($p < 0.05$), confirming reduced barrier integrity. One hour after UTMC, there was a significant decrease in ZO-1 mean pixel intensity ($p < 0.05$). Treatment of cells with Rho inhibitor II (Y16) significantly reduced UTMC-induced dextran flux across the BBB ($p < 0.05$) and UTMC-induced decrease in TEER. In our co-culture model of the BBB, UTMC induces hyperpermeability through modulation of tight junctions, at least in part through a RhoA-dependent mechanism.

9:00

2aBAa4. Calcium influx via mechanosensitive piezo1 channels regulates ultrasound targeted microbubble cavitation-induced endothelial hyperpermeability. Anurag N. Paranjape (Vascular Medicine Inst., Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15213, anuragnparanjape@gmail.com), Xucai Chen, and Flordeliza S. Villanueva (Vascular medicine Inst., Univ. of Pittsburgh, Pittsburgh, PA)

Ultrasound targeted microbubble cavitation (UTMC) enhances drug-delivery across the endothelial barrier, but the underlying molecular mechanisms require further investigation. Here we explore the role of mechanosensitive piezo1 channels in regulating UTMC-induced hyperpermeability. Human coronary artery endothelial cells on transwells showed that UTMC (1 MHz, 250 kPa, 10 cycles, 10 ms interval for 10 s) caused hyperpermeability, demonstrated by reduced transendothelial electrical resistance (TEER) (1.7-fold, $p < 0.05$) and increased dextran flux across the monolayer (twofold, $p < 0.01$ for 70 kDa dextran, 1.6-fold, $p < 0.05$ for 10 kDa dextran). UTMC enhanced the paracellular permeability as shown by transient increase in inter-endothelial gaps. Inhibition of Ca^{2+} influx using mechanosensitive channel inhibitor GsMTx4, or piezo1 siRNA abrogated UTMC-induced hyperpermeability. Reduction in Ca^{2+} influx diminished inter-endothelial gaps and stress-fiber formation, suggesting Ca^{2+} influx is required, at least in part, for UTMC-induced hyperpermeability. UTMC increased nitric oxide (NO) production and inhibition of endothelial NO synthase with L-NAME abrogated UTMC-induced hyperpermeability, indicating a role for NO. Immunostaining showed that UTMC caused reorganization of adherens-junction protein VE-cadherin from a linear to interrupted pattern, which is known to be associated with hyperpermeability. Further elucidation of these pathways will aid in clinical translation and optimization of UTMC for delivery of cell-impermeant drugs.

9:15–9:45 Break

Invited Papers

9:45

2aBAa5. Combining stimuli-responsive colloids and focused ultrasound for drug delivery. Tyrone M. Porter (Univ. of Texas at Austin, 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu)

Over the years my research group has engineered colloids that respond to changes in pressure or temperature locally induced by focused ultrasound. We designed thermosensitive liposomes that released small or large molecular cargo rapidly when heated above normal body temperature by only a few degrees. Potentially, these liposomes can be combined with MR-guided focused ultrasound for noninvasive and image-guided triggered drug delivery. Separately, we developed liquid perfluorocarbon nanoemulsions that can be vaporized with high amplitude acoustic pulses. The resultant bubbles can then be used to facilitate cavitation-mediated permeation of biological interfaces, including cell membranes and the blood-brain barrier. This presentation will review some of our key observations and will discuss the potential for translating these technologies.

10:15

2aBAa6. Promoting immunotherapy of cancer with focused ultrasound. Richard J. Price (Biomedical Eng., Univ. of Virginia, Box 800759, Health System, Charlottesville, VA 22908-0001, rprice@virginia.edu)

Immunotherapy (ITx) has revolutionized cancer treatment. Nonetheless, for several indications, many patients do not respond to ITx due to poor immune cell infiltration and/or limited ITx penetration into solid tumors. Both thermally ablative (T) and mechanical (M) forms of focused ultrasound (FUS) may be capable of overcoming these challenges. For example, the pre-clinical application of T-FUS in combination with myeloreductive gemcitabine elicits immunological control of 4T1 breast tumors and metastases, results that inspired the Br54 clinical trial for early stage breast cancer patients at UVA. Meanwhile, our Br48 clinical trial, which combines αPD-1 ITx with T-FUS in Stage III/IV breast cancer patients, shows that the well-tolerated treatment yields reduced regulatory T cells and transcriptional alterations consistent with inflammasome activation, interferon gamma signaling and $\alpha\beta$ T cell activation. Regarding M-FUS, in pre-clinical brain tumor studies, we have used ImmunoPET imaging to optimize the timing of ITx antibodies (i.e., αCD47) administration with M-FUS-mediated blood-tumor barrier opening. This readily translatable treatment controlled GL261 tumors and improved survival markedly over previous studies that required multi-fold more αCD47 administration to achieve an effect. Overall, our pre-clinical and clinical results support the use of both T-FUS and M-FUS in combination with ITx for multiple cancer indications.

10:45

2aBAa7. Focused ultrasound in the human brain: Current and emerging applications. Nir Lipsman (Sunnybrook Health Sci. Ctr., Univ. of Toronto, 2075 Bayview Ave., A139, Toronto, ON M4N 3M5, Canada, nir.lipsman@sunnybrook.ca)

MR-guided Focused ultrasound (MRgFUS) is a disruptive medical technology, and its implementation in the clinic represents the culmination of decades of research. Lying at the convergence of physics, engineering, imaging, biology and neuroscience, FUS offers the ability to non-invasively and precisely intervene in key circuits that drive common and challenging brain conditions. The actions of FUS in the brain take many forms, ranging from transient blood-brain barrier opening to permanent thermoablation, among other mechanisms. The last decade has seen a dramatic expansion of indications for and experience with FUS in humans, with a resultant exponential increase in academic and public interest in the technology. Applications now span the clinical spectrum in neurological and psychiatric diseases, with insights still emerging from preclinical models and human trials. This presentation provides an overview of clinical trials at various stages of developing using FUS and describes the potential impact, and future directions, of FUS on the landscape of brain therapies.

Contributed Paper

11:15

2aBAa8. Clinical experience with a commercial histotripsy system for the ablation of liver tumors. Osman Ahmed (Univ. of Chicago, Chicago, IL) and Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu)

Liver cancer is a major burden on the American public, with an estimated 30k new diagnoses in 2022. The limitations of current treatment options motivate alternative approaches to improve patient outcomes. Histotripsy is a focused ultrasound therapy that ablates tissue with bubble cloud activity. The capacity of this technology to address liver lesions is currently under investigation (clinicaltrials.gov identifier NCT04572633). In this talk, we will review recent clinical experience with a commercial histotripsy system (HistoSonics, Inc., Ann Arbor, MI) for one site in this trial. Patients with hepatocellular carcinoma or liver metastases admitted to the University of Chicago Medical Center were screened for inclusion criteria, including lesion size ($<$3 cm diameter) and depth ($<$1 cm from liver capsule), the availability of acoustic window, among other specifications. To date, one procedure was performed, with a second scheduled. Identifying the lesion was the most challenging aspect of the procedure, in part due to differences in the window of the ultrasound imaging used for *in situ* guidance and pretreatment diagnostic imaging with MRI. Post treatment imaging confirmed ablation of the targeted region, demonstrating technical success. Early experience indicate histotripsy is a strong candidate as the modality of choice for liver lesions.

11:30–12:00
Panel Discussion

Session 2aBAb

Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics: New Technology Developments for use in Focused Ultrasound Therapy I

Lawrence A. Crum, Chair
Applied Physics Lab, University of Washington, 4662 175th Ave., SE, Bellievue, WA 98006

Chair's Introduction—8:00

Invited Papers

8:05

2aBAb1. The verasonics platform for ultrasound-guided focused ultrasound preclinical studies. Peter Kaczkowski (Verasonics, 11335 NE 122nd Way, Ste. 100, Kirkland, WA 98034, peterkaczkowski@verasonics.com) and Juvenal Ormachea (Verasonics, Kirkland, WA)

Verasonics, in partnership with Sonic Concepts (Bothell, WA, USA), has developed a turnkey platform for Ultrasound-Guided Focused Ultrasound (USgFUS) therapy with performance over a wide range of acoustic regimes. Built around the Verasonics Vantage HIFU ultrasound research system, it uses a 150 mm diameter, f1 HIFU transducer with 64 (0.5 MHz) or 128 (1.1 and 2 MHz) elements arranged in a spiral pattern that produce a highly focused field with low sidelobes over a 3D steering volume. Guidance and monitoring are provided by a coaxially mounted 128-element broadband phased array imaging transducer that can be rotated about the HIFU axis. Coupling to the subject is achieved by means of a membrane sealed water-filled cone through which degassed and temperature regulated water is circulated. The USgFUS applicator is mounted on an articulated arm that can be mechanically locked with a button actuated servo mechanism. Graphical software enables a conventional therapeutic workflow including imaging with any of several B-Mode and Doppler modalities, positioning of the applicator, focal zone exposure planning, therapy delivery, interleaved ultrasound monitoring using any supplied imaging mode or with Thermal Strain Imaging (TSI), and post-therapy imaging. This talk will describe the platform and provide examples of its capabilities using experiments in scattering phantoms and *ex vivo* tissues.

8:25

2aBAb2. Office-based kidney stone management with new ultrasound technologies. Adam D. Maxwell (Urology, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, amax38@uw.edu), Wayne Kreider, Yak-Nam Wang, Bryan W. Cunitz, Barbrina Dunmire, Jeff Thiel, and Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Despite the growing incidence of urinary stones, the fundamental interventions for urinary stones have remained the same for several decades. However, technical innovations in ultrasound have enabled new strategies for stone management. Our team aims to employ these methods to establish a new paradigm to treat stones in an office or clinic rather than a surgical suite. Technologies include new imaging methods to detect and characterize stones, burst wave lithotripsy to fragment stones using focused ultrasound rather than shock waves, and ultrasonic propulsion to promote clearance of fragments, all using a single ultrasound platform. This presentation will describe the principles and development of these technologies, as well as the fruitful partnerships between teams of basic scientists, engineers, physicians, and industry that have enabled rapid development of these concepts to successful clinical trials. [Work supported by NIDDK P01 DK043881 and the Focused Ultrasound Foundation.]

8:45

2aBAb3. Treatment of breast cancer with focused ultrasound: Opportunities and challenges. Allison Payne (Univ. of Utah, 729 Arapeen Dr., Salt Lake City, UT 84109, allison.payne@hsc.utah.edu)

Improved early detection methods and treatments have led to the reduction of breast cancer mortality, yet there remains a need for more conservative, more efficacious and less invasive breast cancer treatments. Focused ultrasound (FUS) is a promising non-invasive treatment for localized breast cancer. Magnetic resonance guided FUS leverages breast magnetic resonance imaging (MRI) to plan, monitor and assess treatments under MRI guidance to attain tumor destruction with focused ultrasound, achieving excellent cosmetic, therapeutic, and safety results. MRgFUS is one of the most attractive emerging minimally invasive procedures for breast cancer. This talk will present different focused ultrasound mechanisms that can be used to treat breast cancer and discuss the design considerations that must be considered when developing transducers, sonication protocols and imaging techniques for the treatment of breast cancer with therapeutic ultrasound. Preliminary results from ongoing breast cancer MRgFUS clinical trials will be presented and future opportunities and ongoing challenges for this exciting technology will be elucidated.

9:05

2aBAb4. Fully electronically steerable high power ultrasound phased arrays for therapy—A review of progress. Kullervo Hynynen (Medical Biophys., Univ. of Toronto/ Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca)

When combined with imaging-guidance focused ultrasound (FUS) provides means for localized delivery of mechanical energy deep into tissues. This focal energy deposition can modify tissue function via thermal or mechanical interactions with the tissue. Traditionally, ultrasound has been focused using spherically curved phased arrays that allow limited electronic steering around geometric focus of the array. To have a fully electronically steered arrays requires the transducer element size to be < wavelength/2 of the ultrasound wave. One way to accomplish this is to use sparse arrays with small transducer elements, the other is to have fully populated arrays with thousands of elements and required electronic complexity. However, the electrical impedance of small transducer elements is high and their efficiency is low and it has been difficult to manufacture arrays with adequate acoustic power. In this talk we will describe how these challenges can be solved with novel manufacturing methods and how we developed complete image-guided systems for both animal and clinical testing. We will also describe our experience with these systems for both brain and body treatments.

9:25

2aBAb5. Sound advice—Sound design. Mark E. Schafer (School of Biomedical Eng., Sci., and Health Systems, Drexel Univ., 3141 Chestnut St., Bossone 708, Philadlephia, PA 19104, mes544@drexel.edu)

Ultrasound energy offers unique capabilities to treat a wide range of medical conditions, through numerous distinct physical interactions with tissue. Unlike diagnostic ultrasound, where safety and efficacy are easily demonstrated and the underlying technology has been available for decades, therapeutic ultrasound applications have explored new regimes and therefore required novel technologies and approaches. This presentation will describe the journey from discovery to clinic, based on examples from the author's personal experience as a consultant and entrepreneur. Developing therapeutic ultrasound products involves several specific and sometimes unique steps. It generally starts with the observation of a therapeutic effect, or a realization that an existing approach could be applied to a different clinical situation. Moving from "N = 1" to a repeatable treatment paradigm can be both exciting and discouraging. The next key steps, which are the hardest, involve translating the initial idea/data/concept into a commercial design, or at least a plan for one. The regulatory, financial, commercial, and clinical environments can be as challenging as the purely technical hurdles, and all factor into the development process. While creating new clinical ultrasound treatments and devices can be difficult and time consuming, it can also be immensely rewarding and satisfying.

9:45–10:05 Break

Contributed Papers

10:05

2aBAb6. Cardioprotective efficacy of ultrasound-targeted nitrofatty acid microbubbles in rat myocardial ischemia-reperfusion injury model. Muhammad Wahab Amjad (Medicine, Univ. of Pittsburgh, 3550 Terrace St., 959 Scaife Hall, Pittsburgh, PA 15213, MUA56@pitt.edu), Soheb Anwar Mohammed, Marco Fazzari, Xucai Chen, Bruce Freeman (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Terry Matsunaga (Dept. of Biomedical Eng. and Dept. of Medical Imaging, Univ. of Arizona, Tucson, AZ), and John J. Pacella (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Over 1 million Americans suffer from acute myocardial infarction (AMI) annually. Although mortality from AMI has decreased, post-AMI congestive heart failure is increasing due to microvascular obstruction (MVO). MVO comprises mechanical obstruction, along with oxidate stress and inflammation. Currently available treatments for MVO are not consistently effective. Hence, we have been developing ultrasound (US)-targeted microbubble cavitation (UTMC) as a potential treatment for MVO. Nitrofatty acids (NFA) are pleiotropic signaling molecules with broad anti-inflammatory actions, potentially beneficial for the treatment of MVO. NFA are amphipathic thus can seamlessly integrate into the phospholipid shell of MBs. Thus, we have constructed microbubbles with NFA (NFABs) to treat ischemia-reperfusion injury (IRI) with UTMC in the rat myocardial model. Left anterior descending (LAD) coronary artery was ligated for 30 min, allowing for IRI. After 15 min of reperfusion, UTMC + NFABs therapy was administered. Echocardiography measurements were recorded at baseline, during ligation and post-treatment. Left ventricular fractional shortening (%) was calculated and NFA concentration in cardiac tissue was determined. UTMC with NFABs exhibited promising efficacy in improving fractional shortening post IRI, and in targeted myocardial delivery of NFA. Studies assessing ejection fraction, myocardial area at risk, histopathology, inflammatory burden, oxidative stress and cytoprotective biomarkers are underway.

10:20

2aBAb7. The efficiency of ultrasonic glymphatic manipulation-based intrathecal drug delivery depends on the physiological states. Haijun Xiao (Eng. (College of Arts and Sci., Lakeshore), Radiation Oncology (Stritch School of Medicine, Health Sci. Campus, Maywood), Loyola Univ. Chicago, Maywood, IL), Binita Shrestha (Biomedical Eng., Univ. of Texas, Austin, Austin, TX), Gabriel Gallegos, Nuala Kalensky, Dhruv Patel (Eng. (College of Arts and Sci., Lakeshore), Radiation Oncology (Stritch School of Medicine, Health Sci. Campus, Maywood), Loyola Univ. Chicago, Chicago, IL), Maurizio Bocchetta (Cancer Biology, Stritch School of Medicine, Loyola Univ. Chicago Health Sci. Campus, Maywood, IL), Tyrone M. Porter (Biomedical Eng., Univ. of Texas, Austin, U.S., TX), and Muna Aryal (Eng. (College of Arts and Sci., Lakeshore), Radiation Oncology (Stritch School of Medicine, Health Sci. Campus, Maywood), Loyola Univ. Chicago, 1032 W Sheridan Rd., Chicago, IL 60660, maryal@luc.edu)

Recently, for the first time, we discovered that brain-wide application of low-pressure transcranial focused ultrasound can be used to enhance the parenchymal penetration of intrathecally administered imaging agents to a large portion of the brain through the glymphatic pathway. This proof-of-concept study only investigated the delivery efficiency in fully anesthetized animals (2.5% isoflurane). Here, we aim to determine how different anesthetic conditions affect the delivery efficiency as the glymphatic system is altered by the physiological states. We assigned three rat experimental groups: heavy-3% isoflurane (n = 4), moderate-2.5% isoflurane (n = 4), and light-1.5% isoflurane (n = 5), co-delivered two different-sized imaging tracers intrathecally (IRDye + trypan blue, IRDye labeled IgG antibody + trypan blue), and exposed ultrasound (650 kHz, 0.2 MPa for 10 min). We measured heart rate, respiratory rate, oxygen level, perfusion, and body temperature during the test and imaged the *ex vivo* brain using an optical imaging system. We observed that the distribution of the delivered agents in the brain parenchyma is higher in light anesthetic groups as compared to

moderate and higher ones, and it depends on the molecular size of the agents as well. Future work is to increase the number of animals in each group, perform statistical analysis to confirm the preliminary findings, and determine the delivery efficacy of the technique on metastatic cancer model. M. Aryal *et al*., *J Control Release* (2022), PMID:35798095.

10:35

2aBAb8. *In vivo* soft tissue aberration correction for histotripsy using acoustic cavitation emissions. Ellen Yeats (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd, Ann Arbor, MI 48109, yeatsem@umich.edu), Greyson Stocker, Jonathan R. Sukovich, Zhen Xu, and Timothy L. Hall (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Phase aberration by heterogeneous soft tissues decreases the focusing quality of transabdominal histotripsy, thereby increasing the energy required to generate cavitation for therapy and elevating the risk of unsafe tissue heating. *In vitro* studies have shown that the acoustic signals emitted by histotripsy cavitation bubbles can serve as 'point sources' for aberration correction (AC). This study assessed the efficacy of cavitation emission-based AC *in vivo*. A 750-kHz, 260-element, receive-capable histotripsy phased array was used to generate cavitation at 3 locations in the livers of 3 pigs ($n = 9$). For each location, cavitation emission signals were received and cross correlated to determine a set of corrective phase delays. Then, the array was re-fired at increasing increments of driving power both with and without aberration correction. It was found that AC decreased the transducer power required to generate cavitation by 16%–48%. This result suggests that AC using the cavitation emissions from an initial test pulse can substantially reduce the energy delivered to intervening tissues during histotripsy therapy and thus increase treatment safety.

10:50

2aBAb9. Low-intensity triggered release with spatial and temporal control using antibubbles. Athanasios Athanassiadis (Heidelberg Univ., Im Neuenheimer Feld 225, IMSEAM - AG Fischer, Heidelberg 69120, Germany, thanasi@uni-heidelberg.de), Nicolas Moreno-Gomez (Heidelberg Univ., Heidelberg, Germany), Albert T. Poortinga (Eindhoven Univ. of Technol., Eindhoven, the Netherlands), and Peer Fischer (Heidelberg Univ., Heidelberg, Germany)

One approach to ultrasound therapy is to use therapeutic agents that can be activated by focused ultrasound when they reach a specific site in the body. Commonly, such agents are loaded on the surfaces of microbubbles, which respond strongly to ultrasound and shed the payload. However, microbubbles require high pressures and mechanical indices to burst (typically above 200 kPa, $MI > 0.2$). Moreover, the quantity and types of therapeutic payload that can be delivered by microbubbles are limited because payloads must be attached to the microbubble surface. Here, we show how these limitations can be overcome by using stabilized antibubbles as an ultrasound-responsive carrier. Antibubbles are liquid droplets encased within an air bubble. Because therapeutic payloads can be encapsulated in the core, larger volumes can be carried per antibubble. Additionally, by carrying payloads in the volume rather than on the surface, a wider variety of payloads can be carried. Through experiments we demonstrate that antibubbles respond strongly to ultrasound and can release payloads with pressures below 50 kPa ($MI = 0.05$) for certain formulations. By modifying the formulation, we show that the release pressure and temporal release profile can be tuned. Finally, we show that the bursting is highly selective in space, demonstrating that antibubbles can be used for precise delivery of payloads using shaped, low-intensity ultrasound fields.

11:05

2aBAb10. Heterogenous angular spectrum approach based holograms for trans-skull focused ultrasound therapy. 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)

Focused Ultrasound (FUS) therapy in the brain entails accounting for strong distortions in the transmitted wavefields introduced by the skull. While these aberrations are compensated by MR-guided FUS (MRgFUS) phased array-based systems by adjusting the phase and amplitude of individual transducer elements, their cost and reliance on MRI may limit their widespread adoption. This work presents phase-only acoustic holograms coupled to a single transducer element as a safe, effective, and affordable alternative. We used the Heterogenous Angular Spectrum Approach (HASA) to propagate between the transducer and target plane(s) and applied automated differentiation for gradient computation to iteratively optimize the phase of the holographic lens. To test our design approach, we employed experimentally validated k-wave simulation. Our results show a 9 dB increase in the peak signal-to-noise ratio (PSNR) of the target acoustic field at 1 MHz as compared to the case where aberrations are uncorrected. Furthermore, we addressed the effect of frequency and aperture on the quality of hologram construction. We showed the feasibility of implementing it in a clinical-scale FUS system targeting deep (>6 cm) brain structures. In summary, the combination of HASA with automated differentiation is an efficient and accurate tool for hologram design through heterogeneous media.

11:20

2aBAb11. Design and fabrication of an acoustic tweezer system for kidney stone manipulation. Mohamed A. Ghanem (Univ. of Washington, 1013 NE Boat St., Seattle, WA 98105, mghanem@uw.edu), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Ekaterina Kuznetsova (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Acoustic radiation forces can surround and trap objects to manipulate them in three dimensions. It has been demonstrated that the radiation forces can maneuver solid heavy objects similar to kidney stones in live animals under safe acoustic exposure levels. However, initial experiments identified limitations on the steering range of objects because of the array design, and tissue aberrations. Here, we present a method to design an in-house 256 multi-element array to improve the steering range of 2–5 mm stones. We modeled the acoustic field for various geometrical parameters including the element size, transducer focusing, and F-number at a frequency range from 0.5 to 1 MHz. The parameters were then used to simulate the trapping radiation forces on stone models to finalize the design of the array. The array was fabricated with a center frequency of 950 kHz and a focal distance located at 120 mm and an F-number of 1.5. A holography scan was performed to characterize the transducer output. Various trapping beams were measured and compared to simulations. The resulting radiation forces produced were on the order of the weight of the trapped 2-5 mm stone models. [Work supported by NIH P01-DK043881, K25-132416, and Applied Physics Laboratory SEED fellowship.]

Session 2aCA

Computational Acoustics, Architectural Acoustics, Physical Acoustics, and Musical Acoustics: Real-Time Computing II (Hybrid Session)

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

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Institute, School of Architecture, 110 8th Street, Troy, NY 12180

Invited Papers

10:05

2aCA1. Real-time processing approaches for interactive room acoustic auralization in the rtSOFE. Bernhard U. Seeber (Audio Information Processing, Tech. Univ. of Munich, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de)

Room acoustic simulation of early reflections with the mirror-image method for arbitrary room geometries is an exponential problem, which, if computed to a high reflection order, will quickly exceed available computational and memory resources. The real-time Simulated Open Field Environment (rtSOFE) is freely available room-acoustic simulation and auralization software based on the mirror-image method. For auralization, rtSOFE synthesizes room impulse responses for each playback channel from the list of mirror images. A convolver program receives these seconds-long room impulse responses and convolves them for each sound source, sums across sources, and performs the playback equalization. Both programs perform computations in multiple threads in parallel using the OpenMP library. Many operations are implemented with Intel Advanced Vector Extensions (AVX) to increase processing speed. Image source trees are terminated after a definable number of invisible parent sources, keeping the number of computed mirrors manageable. With this, rtSOFE is capable of computing room impulse responses to high reflection orders in interactive real-time settings. I will present the software concept and discuss the approaches that speed up the computation.

10:25

2aCA2. Event- and sample-based audio processing: A shotgun marriage. Miller Puckette (Music, UCSD, 9500 Gilman Dr., La Jolla, CA 92093, msp@ucsd.edu)

Real-time audio computation tends to fall into two very different paradigms. On one hand we deal with continuous streams of audio sample frames having a fixed number of channels and a fixed data type. Our main conceptual tool is the Nyquist theorem. On the other hand, musical scores and sporadic control streams (for instance MIDI or network packets) do not fit comfortably into any fixed data type and need not fall on any fixed sample rate. In this talk I'll describe the choices taken in the design of the Pure Data real-time music and audio programming environment. The design makes it possible to make exactly reproducible real-time audio computations that combine two computational paradigms, one for vectorized data streams (intended for, but not limited to, audio), and the other to support control computations with sub-sample-accurate time tags.

10:45

2aCA3. Real-time co-location in dynamic virtual soundscapes across multi-occupant immersive rooms. Mincong Huang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, huangm5@rpi.edu), Samuel Chabot (Rensselaer Polytechnic Inst., Troy, NY), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Immersive rooms, a type of virtual reality system consisting of human-scale panoramic visual and acoustic display systems and distributed sensing apparatus for occupant motion, have been increasingly adopted for dynamic and interactive applications. While these applications enable multi-user audiovisual immersion and navigation from a single physical location, they have yet to propagate along multiple homogeneous system infrastructures in a networked manner. In this work, we intend to co-locate two physically-remote immersive rooms – at EMPAC and the CRAIVE-Lab, respectively – in a single system of shared environments developed in Unity and embedded with virtual soundscapes. This system actively monitors spatial properties of both immersive rooms' dynamic virtual footprint and their corresponding occupants. It generates virtual sound sources. both procedurally and through spatially-aware user inputs. The sound sources are rendered in real time via an algorithm synthesizing a ray-traced early reflection window and a parameterized late reverberation estimate from in-scene geometries. The co-located virtual soundscapes, displayed in individual immersive rooms through their respective multi-channel wave field synthesis loudspeaker systems, are shared as such that the user interaction in one physical location has holistic effects on the experience of virtual environments across all associated physical locations. [Work supported by NSF IIS-1909229 & CNS-1229391.]

11:05–11:20 Break

11:20

2aCA4. Interpolation of scheduled simulation results for real-time auralization. Philipp Schäfer (IHTA, RWTH Aachen Univ., Kopernikusstraße 5, Aachen 52074, Germany, philipp.schaefer@akustik.rwth-aachen.de), João Garrett Fatela, and Michael Vorlaender (IHTA, RWTH Aachen Univ., Aachen, Germany)

The approach of auralization allows rendering plausible audio signals based on purely synthesized data. One key aspect for this is the simulation of sound propagation. Using sophisticated physics-based models, even complex situations—like sound propagation through an inhomogeneous, moving atmosphere—can be considered. However, even with fast approaches based on geometrical acoustics such simulations might not be real-time capable when being integrated in the auralization processing chain. This is because the update rate of the simulations is significantly lower than the audio block rate which can lead to artifacts, especially when considering effects like the Doppler shift. In this contribution, we propose a method aiming to run an artifact-free auralization despite a relatively low and potentially irregular simulation update rate. For this purpose, the simulations are run in a thread separate from the main processing chain and the respective outputs are interpolated. The method is applied to an aircraft flyover scenario considering curved sound propagation in a stratified, moving atmosphere.

11:40

2aCA5. Implementing neural networks in low-latency audio applications. Thiago Henrique Gomes Lobato (HEAD Acoust. GmbH, Ebertstraße 30a, Herzogenrath 52134, Germany, thiago.lobato@head-acoustics.de), Roland Sottek (HEAD Acoust. GmbH, Herzogenrath, Germany), and Michael Vorlaender (IHTA, RWTH Aachen Univ., Aachen, Germany)

The use of neural networks is becoming increasingly prevalent due to their ability to represent complex relationships and solve complex problems. However, implementing these models in systems that require low-latency output can be challenging, especially for practitioners who are used to developing their models in controlled environments like Python notebooks. Another issue is the high computational cost of complex models, which limits the minimum possible latency. This paper presents approaches for deploying models in audio applications, discusses the advantages and disadvantages of each approach, and presents strategies to reduce the inference cost of models without significantly sacrificing accuracy, using techniques such as model quantization. To illustrate these methods, example implementations of real-time beamforming deconvolution and real-time music DSP processing are shown.

Session 2aEA

Engineering Acoustics, Signal Processing in Acoustics, Architectural Acoustics, Underwater Acoustics, Physical Acoustics, and Noise: Microphones: Design, Development, and Characterization I

Thomas E. Blanford, Cochair
The Pennsylvania State University, State College, PA 16804

Zane T. Rusk, Cochair
The Pennsylvania State University, 104 Engineering Unit A, University Park, PA 16802

Chair's Introduction—9:00

Invited Papers

9:05

2aEA1. Microphones: A bit of history. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu)

The earliest commercially successful microphone design was the carbon microphone, whose original patent was filed by Emile Berliner in 1877. An interesting variant of the basic carbon microphone was a dual diaphragm microphone patented by Granville Woods in 1883. The basic carbon microphone provided adequate performance at low cost and remained in service in the telephone system for at least the next century. Other microphone technologies can also be found in the early patent literature, though their commercial acceptance was delayed until electronic amplification was possible. The eventual development of vacuum tube amplifiers enabled the development of the condenser microphone, whose initial patent was filed in 1916. Condenser microphones provide the improvement of wide bandwidth and flat response compared to carbon microphones. The invention in the 1960s of high quality electret materials enabled the manufacture of low cost electret microphones that quickly displaced carbon mics in telephones and most commercial devices for at least the next 40 years. Then, in the early years of the 21st century, condenser microphones implemented in silicon, also known as MEMS microphones, took over in high all volume applications. This paper will discuss this history and the current state of the art.

9:25

2aEA2. Sixty years of contributions to the world of microphones. James E. West (Elec. and Comput. Eng., Johns Hopkins Univ., 2327 Boston St. Unit 19, Baltimore MD, 21224, Baltimore, MD 21224, jwest10@jhu.edu), Ian M. McLane (Elec. and Eng. Dept., Johns Hopkins Univ., Baltimore, CA), and Valerie Rennoll (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

For almost 60 years, electret microphones have been the preferred sensors for applications in communications, mainly because the microphones are linear over a broad frequency range and rather simple to manufacture. Because the electret microphone can be mass produced with only slight differences in phase and frequency response, multiple units can be combined to form a variety of directional arrays ranging from second-order unidirectional to two-dimensional arrays for focusing on a specific area. While electret microphones and arrays have similar utility for monitoring lung and heart sounds from the body, the body sounds captured can be easily corrupted by noise external to the body. Advanced signal processing techniques can mitigate contributions from airborne noise but are computationally intensive. By modifying the acoustic impedance of the electret microphone's diaphragm to match that of the body, we are able to capture high-fidelity heart and lung sounds without corruption from airborne noise. This redesign of the original electret microphone could provide a method to continuously monitor lung and heart sounds from a subject regardless of their surrounding noise environment.

9:45

2aEA3. The six prototype microphone elements that led to the Shure Unidyne I Model 55. Michael S. Pettersen (Corporate History, Shure, Inc., 5800 West Touhy Ave., Niles, IL 60714, petterm@shure.com)

Benjamin B. Bauer (1913–1979) was an ASA Fellow and awarded the ASA Silver Medal (1978). He held over 100 patents for acoustical/audio technology, with his 1941 patent being arguably the most significant: the Uniphase Acoustical Network principle, integral to the Shure Unidyne model 55 microphone. Introduced in 1939 and still manufactured today, the Shure Unidyne was the first moving coil unidirectional microphone using a single dynamic element. Six prototype mic elements for the Unidyne, fabricated by Bauer in the late 1930s, reside in the Shure archive. This presentation provides details on each prototype using photos, drawings, and Bauer's text from his lab notebooks, providing insight into the creation of the one of the world's most famous microphones.

10:05

2aEA4. Trends in the development of measurement microphones. Per Rasmussen (GRAS Sound & Vib., 33 Skovlytoften, Holte 2840, Denmark, pr@grasacoustics.com)

Measurement microphones are, in principle, simple transducers that convert dynamic pressure fluctuations into an electrical output signal. The range of applications is, however, extremely wide—from measuring exceptionally low-level signals (below 0 dB; for example, measuring tiny insects) to the 190 dB signal of large rocket launches. The frequency range of interest spans from below 0.1 Hz for sonic boom measurements to 200 kHz or more for detailed investigations of very short impulses or wind tunnel scale modelling. At the same time, the individual application may mandate specific mounting conditions with specialized form factors for the microphone and associated preamplifier design. This trend towards application-specific testing has led to the development of a wide range of specialized microphones and accessories. Still, this wide variety of measurement microphones must measure accurately and be calibrated to be traceable to the basic unit of measure as, for example, defined by the SI system. This is ensured by maintaining reference to the basic IEC definition of measurement microphones in the IEC 61094 series of standards for microphones and calibration methods.

10:25–10:40 Break

10:40

2aEA5. Microphone vibration sensitivity: Frequency response, measurement error, and usage of measured responses in lumped element simulations. Charles B. King (R&D, Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60143, charles.king@knowles.com)

Microphones are designed to respond to acoustic pressure fields, but they can also be excited by external sources of mechanical vibration. In hearing aids, feedback is a difficult problem to resolve due to the high gain used to amplify sounds. A new method to measure a microphone's "intrinsic vibration sensitivity" was introduced at the 183rd ASA conference in Nashville. This paper will describe details of a practical measurement and how these details impact measurement error. It will demonstrate how the measured response can be integrated into lumped circuit models for analysis in SPICE or FEA programs. Utilizing measurement error and lumped circuit models, a description of how error of the complete system varies over frequency will be shown.

11:00

2aEA6. Comparative evaluation of omnidirectional and directional micro-electromechanical system microphone performance. Carly Stalder (Soundskrit, Inc., 94 Rue Bourget, Apt. 201, Montreal, QC H4C 2M2, Canada, carly.stalder@soundskrit.ca) and Stephane Leahy (Soundskrit, Inc., Montreal, QC, Canada)

As the need for directionality becomes a key requirement in audio applications, directional microphones have begun to enter the micro-electromechanical system (MEMS) design and market space, and their performance is approaching that of top-of-the-line omnidirectional MEMS microphones. This presentation examines and compares the performance limitations for both types of MEMS microphones and suggests more comprehensive methods of characterization that allow the qualities of directional MEMS microphones to be fully captured. Mechanical thermal noise caused by Brownian motion of air particles, measured with a laser Doppler vibrometer system, are shared and discussed for a variety of microphones and test structures. Parameters such as signal-to-noise ratio, total harmonic distortion, and frequency response are re-examined in order to capture the value of a microphone that exhibits directional behavior. Insights and suggestions are made for improvement and future work.

11:20

2aEA7. MEMS microphone arrays for quench detection in superconducting cables. Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu), Aidan Beckett, Rishabh Chaudhary, Luisa Chiesa (Mech. Eng., Tufts Univ., Medford, MA), Amish Desai (Tanner Res., Monrovia, CA), Odin Doolittle (Mech. Eng., Tufts Univ., Medford, MA), Michael Emerling (Tanner Res., Monrovia, CA), Peter Moore (Mech. Eng., Tufts Univ., Medford, MA), Makoto Takayasu (Plasma Sci. and Fusion Ctr., Massachusetts Inst. of Technol., Cambridge, MA), and Boning Zhang (Mech. Eng., Tufts Univ., Medford, MA)

An acoustic array is proposed as a quench detection method in superconducting magnets. A quench occurs when the current density in the superconductor exceeds a critical value, resulting in a loss of superconductivity and rapid local heating. This event is destructive and must be rapidly detected. It is thought that the quench may act as an acoustic source (Takayasu, 2019), which could be detected and localized by a microphone array inserted into the cryogenic coolant. A main advantage of this method is that acoustics propagate 1000 times faster than the normal zone propagation velocity in HTS conductors, providing for fast detection times. To demonstrate this concept, we first characterized the performance of a piezoelectric MEMS microphone and several potential preamplifiers under cryogenic conditions. An acoustic sense node was then constructed that operates down to 10 K. A cryogenic probe incorporating the MEMS array was used to study a quench event in a segment of a 2 mm wide REBCO tape. Quench experiments were carried out in a 7.6 cm diameter, 111 cm tall cryostat in Helium gas at 20 to 50 K. The MEMS array clearly detects a quench induced failure. Other observed acoustic features of unknown origin will be described.

Session 2aMU

Musical Acoustics: Musical Acoustics General Topics II

Kurt Hoffman, Chair
Whitman College, Walla Walla, WA 99362

Contributed Papers

7:40

2aMU1. Formant-harmonic interactions and vowel perception. May Pik Yu Chan (Dept. of Linguist., Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu) and Jianjing Kuang (Linguist., Univ. of Pennsylvania, Philadelphia, PA)

The technique of "turning over" in singing which causes perceivable vowel quality shifts has been attributed to the crossing over of H2 over F1. The present work seeks to validate these perceptual claims. A modified AXB discrimination task was completed by 63 speakers, consisting of Klatt synthesized vowels. Versions of the vowels (/i, u, ɑ, ɔ/) varying in the F_1 (+/50 Hz) dimension, the F_2 (+/−100 Hz) dimension or both F_1 and F_2 dimensions were used as reference "A/B" tokens, while the "X" token varied along 9 semitones in an F0 continuum. F1 of the vowel aligned to H2 at step 5 of the F0 continuum. Listeners judged whether the vowel quality of token 'X' (F0 modified) was closer to "A" or "B" (F1/F2 modified). Results find that listeners perceive the vowel to have a lower F1 during H2-F1 crossover, but the vowel height is perceived to be lower before and after the alignment. This pattern was more robust in the vowel height than the vowel backness dimension, with similar results across vowels. These findings were confirmed with logistic regression models. We conclude that formant harmonic crossovers have psychoacoustic effects on vowel perception, holding implications in vowel modification in singing.

7:55

2aMU2. Comparison of vocal vibrato across multiple singers and vowels. Hanna Berger (Audio and Music Eng., Univ. of Rochester, 500 Joesph C. Wilson Blvd, Rochester, NY 14627, hberger6@u.rochester.edu) and Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Vibrato is a common technique used by singers to add expressiveness to their voice, which involves oscillating slightly above and below the target pitch. Analyzing the features of vibrato could be useful in projects such as identifying singers based on their voice or creating a more realistic autotune. The parameters of the vibrato explored included frequency modulation width, frequency modulation rate, and amplitude modulation width. To collect data on these parameters, various singers of different voice parts were recorded singing scales on different vowels with and without vibrato. The singers were recorded in an anechoic chamber to minimize the impact of the room on the samples. The scales were then cut down into individual pitches that could be analyzed. Using MATLAB, the vibrato parameters of the voice samples were calculated and graphed. The data was then compared between vowels sung by the same singer, and between different singers.

8:10

2aMU3. Singing in different performance spaces: Associations between room acoustic and singers perception. Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu), Yvonne Gonzales Redman (School of Music, Univ. of Illinois Urbana-Champaign, Champaign, IL), Joshua D. Glasner (Speech-Lang. Pathol., Delaware Valley Univ., Philadelphia, PA), and Dario D'Orazio (Univ. of Bologna, Bologna, Italy)

While the acoustic design of concert halls and other performance venues has evolved over time, the fact remains that the acoustic environments are generally designed for the benefit of the listening audiences rather than that of the performers. However, singers adapt their voice to room acoustics. This paper expands upon our current understanding of how a room's acoustic environment influences a singer's performance from the perceptions, observations, and lived experiences of performers. The subjects were nine classically-trained singers. Subjects sang the same aria in five different performance venues. After the performance, the singers filled out a questionnaire. Questions were subdivided into five major sections that were designed to capture their perceptions of (1) overall impressions; (2) background noise levels; (3) voice support; (4) Voice clarity. It was observed the overall impression of the room was mostly correlated with STv and IAC-C_{late}. The perception of reverberation and the noisiness were correlated with C80 and EDT. The perceived voice support and clarity were mostly correlated with STv. This finding indicates the importance of auditory feedback in singers' performance and the need for an acoustical design that takes into account the performers perception.

8:25

2aMU4. Simulation of multiphonic vocalization: How many sound sources can the lungs support simultaneously? 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)

The hypothesis tested was that the lung pressure required for multiphonic vocal sound sources (some combinations of true folds, false folds, aryepiglottis folds, tongue base, velum, lips) is the sum of all the individual threshold pressures. The approach was strictly computational. Fluid transport and wave propagation in the airways were produced by an abbreviated Navier–Stokes solution. Each of the sound sources was modeled with three narrow 0.2 cm sections having viscous and elastic wall properties that could oscillate in isolation. The length of the supraglottal vocal tract was 17.5 cm. The simulation was used to compute phonation threshold pressure, radiated power from the mouth, degree of aperiodicity, and glottal efficiency.

8:40

2aMU5. Electronic architecture—Improving room acoustics using time variant electro-acoustic systems. Steve Barbar (E-Coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@ecousticsystems.com)

In 1990, we presented a technical paper at the Audio Engineering Society that described a new and unique system that enables microphones to be positioned as much as 50 feet from the sound source while producing high reverberation level without coloration from acoustic feedback. Since that time, hundreds of these systems have been installed throughout the world in both a wide range of applications, as well as wide variety of venues. We will discuss successful solutions for acoustical problems that are difficult to address using conventional architecture.

8:55

2aMU6. An investigation of rooms with reflection-free zones using finite-difference methods in curvilinear coordinates. Samuel D. Bellows (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, samuel.bellows11@gmail.com) and Timothy W. Leishman (Brigham Young Univ., Provo, UT)

Small-room acoustical designers have applied the concept of early reflection-free zones (RFZs) to recording studio control rooms and other critical listening environments. The rooms' unique shapes and treatments create listening regions with minimal early reflections to improve audio monitoring. One standard tool for studying room acoustics is the finite-difference (FD) method. However, it typically uses regular Cartesian grids, which lead to stair-casing discretization effects for arbitrarily shaped boundaries, including those of the RFZ room. This work presents how FD methods in curvilinear coordinates overcome these difficulties. Both modal and time-domain analyses yield insights into the acoustical differences between RFZ and rectangular shoebox rooms.

9:10

2aMU7. Effect of musical instrumentation on perception of reverberation in simulated listening environments. Christopher M. Jasinski (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, jasinski@hartford.edu) and Robert Celmer (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, West Hartford, CT)

The reverberance of a performance venue is often characterized by an important metric, reverberation time. In order for architects, acousticians, and performers to better understand the value of knowing and predicting the reverberation time in different spaces, the just-noticeable-difference (JND) is often cited. The established JND for reverberation time is approximately 5% of the reference reverberation, noted in the standard ISO 3382-1. This value was determined using impulsive sound, meaning it may not provide a complete picture for applications related to music. This study aimed to evaluate several elements of musical performance that may alter the perception of reverberation in a simulated listening environment. Primarily, the effect of instrumentation was investigated, including the use of solo, small ensemble, full orchestra, and vocal recordings for comparison. Results were generated using test participants listening to multiple soundfields and providing subjective evaluation. Reverberant soundfields were created in an anechoic listening environment and auralizations were produced using ODEON room acoustics software. Results indicated statistically significant mean differences in reverberation perception between musical instrumentation groups used during experiments. [Work supported by The Paul S. Veneklasen Research Foundation.]

Session 2aNS

Noise: Assorted Topics on Noise

James E. Phillips, Chair
Intertek, 4703 Tidewater Ave., Suite E, Oakland, CA 94601

Contributed Papers

7:45

2aNS1. Challenges in post-construction sound monitoring for rural wind projects. Ken Kaliski (RSG, 55 Railroad Row, White River Junction, VT 05001, ken.kaliski@rsginc.com)

Postconstruction sound monitoring for rural wind projects has unique challenges. While background sound levels tend to be lower than more urban areas, the sound levels from the wind turbines also tend to be lower, making signal to noise ratios rather low. Thus, differentiating wind turbine sound with wind-induced sound can be difficult, especially when wind speeds are variable. Other issues include safety, wildlife, harsh weather conditions, access to remote sites, difficult protocols, and noise contamination. These issues will be discussed in the context of several recent postconstruction sound monitoring campaigns. Each campaign uses wind turbine shutdowns to measure adjacent background sound levels so that turbine-only sound levels can be calculated. Advantages and disadvantages of the wind turbine shutdown method will be discussed and compared with other methods that don't require wind turbine shutdowns, such measuring under high wind shear conditions (low ground wind speeds and high hub-height wind speeds), proxy locations, and wind speed binning.

8:00

2aNS2. Predicting vehicle category using psychoacoustic indicators from road traffic pass-by noise. Ablenya Barros (Univ. of Antwerp, Groenenborgerlaan 171, Antwerp 2020, Belgium, ablenya.barros@uantwerpen.be), Michiel Geluykens (KU Leuven, Antwerpen, Belgium), Frederico Pereira (Comput. Graphics Ctr. - Univ. of Minho, Guimaraes, Portugal), Luc Goubert (Belgian Rd. Res. Ctr., Zaventem, Belgium), Elisabete Freitas (Univ. of Minho, Guimaraes, Portugal), and Cedric Vuye (Univ. of Antwerp, Antwerpen, Belgium)

A set of road traffic pass-by noises containing more than 2000 vehicles was recorded following the Statistical Pass-By (SPB) methodology (ISO 11819-1:2022). Besides the acoustic descriptors, psychoacoustic indicators (loudness, sharpness, roughness, fluctuation strength) were retrieved for each pass-by of the three vehicle categories defined in the standard (passenger cars, dual-axles and multi-axles heavy vehicles). A fourth vehicle category, comprised of delivery vans, was also investigated. All psychoacoustic indicators significantly differed among vehicle categories, meaning that not only intensity descriptors but also temporal and spectral features of pass-by noise distinguish those classes. With enough instances and a balanced dataset across groups, a machine-learning classification algorithm was trained with 70% of the dataset to predict vehicle categories using the psychoacoustic indicators. Classification accuracy on the test set reached 72.5%. Accuracy losses were primarily caused by 25% of the actual passenger cars being misclassified as vans and vice-versa. Pooling these two categories increased accuracy to 82%. With more descriptors from road traffic pass-by noise than uniquely its maximum noise level, limiting definitions of vehicle categories may be overcome. As a result, measurements such as the SPB can become broader and vehicle fleets worldwide more consistently represented in terms of noise perception.

8:15

2aNS3. The effect of ambient noise within a vehicle on the audibility of safety sirens. Madeline K. Webber (Miami Univ., 3965 South Beechgrove Rd., Wilmington, OH 45177, webberm3@miamioh.edu), Norman Lee (Biology Dept., St. Olaf College, Northfield, MN), and Christopher Beer (Phys., Miami Univ., Oxford, OH)

Driver response times to safety vehicle sirens can make the difference between life and death. Unfortunately, siren audibility is impeded by additional auditory stimuli (such as music), which more than 90% of drivers listen to. To better understand siren effectiveness in alerting drivers, ambient noise (including such additional stimuli) is measured with respect to a typical sedan to understand masking. Quantifying the sound propagation characteristics, specifically diffraction and attenuation, of sirens at set distances is done by measuring the A-weighted sound pressure levels outside and inside the sedan. Measurements are taken to simulate an emergency vehicle approaching from behind. Initial findings demonstrate that dBA levels are reduced more than two-fold when propagating inside a sedan with all noise sources off (engine, etc.). Ambient noise is studied in multiple designs, including engine idling or driving, and music at various sound pressure levels. This enables quantifying human detection thresholds to determine minimal masking values in various types/levels of ambient noise. Human detection thresholds are determined using established signal-to-noise ratio standards. Findings shall be discussed in terms of siren effectiveness in alerting drivers in the presence of ambient noise to provide insights to further the safety of the public and emergency personnel.

8:30

2aNS4. Effect of hearing protection attenuation on impulse insertion loss and kurtosis. William J. Murphy (Stephenson and Stephenson, Res. and Consulting, 2264 Heather Way, Forest Grove, OR 97116, wmurphy@sasrac.com) and Gregory Flamme (SASRAC, Forest Grove, OR)

Kurtosis (the fourth standardized moment of the sound pressure) has been used to assess the additional risk of hearing loss for complex or impulsive noise exposures. Murphy *et al*. [2012] reported the impulse peak insertion loss of five hearing protection devices (HPDs) for firearm impulse noise. Fackler *et al*. [2017] reported a spectral insertion loss for several HPDs assessed with firearm and shock tube impulse noise sources. Murphy [2019] previously reported the effect of five hearing protection devices on the kurtosis and insertion loss assessed with jackhammer noise. Unprotected Jackhammer noise exhibited kurtosis values between about 15 to 17, whereas protected exposures exhibited kurtosis between about 3 to 12. Anderson and Argo [2022] reported that insertion loss was unaffected by kurtosis level for seventeen HPDs measured on an acoustic test fixture under headphones. This paper will apply the complex transfer functions from Murphy *et al*. [2012] to the jackhammer noises measured by Murphy [2019]. The levels of the unoccluded noises transformed to the ear canal of the fixture will be compared to occluded jackhammer noises levels measured in the fixture.

8:45

2aNS5. Frequency compensation for in-ear noise dosimetry. John Parkins (Red Tail Hawk Corp., 1111 Locust St., Unit 8i, Philadelphia, PA 19107, jparkins@rthcorp.com)

Personal noise dosimeters (PNDs) are used in the prevention of noise induced hearing loss (NIHL) by monitoring noise exposure. Exposure of a worker is typically estimated by measuring the noise level at the shoulder location and subtracting the estimated protection provided by any worn hearing protection devices. Current noise dosimeters in the marketplace are deficient because the noise level at the shoulder does not necessarily represent the noise exposure in the ear canal due to shadowing, canal resonances and other acoustical phenomena. Moreover, workers often deliberately insert earplugs to shallow canal depths to better hear their environments, and earmuff attenuation is inconsistent due to variations in head geometry. Thus, the estimated hearing protection is often in error. In-ear personal noise dosimeters (iPNDs) can solve these problems because they incorporate a microphone within a hearing protection earplug that measures the actual noise level in an ear canal. However, the acoustics of the in-ear measurement are quite different compared to the shoulder measurement, and in-ear measurements must be frequency compensated to achieve valid results. This paper discusses in-ear noise dosimetry, why frequency compensation is needed and how to compensate the microphone signal. [This work was funded by a CDC SBIR grant.]

9:00

2aNS6. Hydraulic flow noise characterization of spool valves using experimental techniques. Carter A. Paprocki (Mech. Eng., Michigan Technol. Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cap6296@psu.edu), Sriram Malladi (Mech. Eng., Michigan Technolog. Univ., Houghton, MI), and Andrew Barnard (Acoust., Penn State, State College, PA)

This research was conducted to characterize and develop a deeper understanding of the hydraulic flow noise for a spool valve. Hydraulic noise was analyzed to study the interaction of the valve spool and the flow noise generation to isolate where in the spool articulation the most noise was generated. The goal was to correlate turbulent fluid flow in pressurized hydraulic circuits to different spool positions while isolating what factors were the main drivers of spool noise generation. The experiments were conducted on a valve controlled by a closed-circuit hydraulic system, using this system fluid-born noise could be isolated from other noisy mechanisms and a noise profile of each could be measured. It was found that internal spool geometry and flow channel open area played the largest role in valve noise generation for the valve. In the future, more valves could be tested to develop a robust profile library using the test methods developed.

9:15

2aNS7. Measurements of ship noise using an acoustic camera: A first survey. Johan A. Bocanegra, Davide Borelli (Dept. of Mech. Eng., Univ. of Genova, Genova, Italy), and Corrado Schenone (Dept. of Mech. Eng., Univ. of Genova, Via Opera Pia 15/A, Genova I-16145, Italy, corrado.schenone@unige.it)

While the acoustic impact of harbours is becoming an increasingly important issue, to the point of limiting their growth, the measurement of noise from ships and port operations is still rough. Two of the main problems that are commonly encountered are source overlapping and large distance from the emitting sources. Source overlapping prevents from identifying the specific contribution coming from each acoustic source. The large distance from the emitting sources, for instance the funnel of a ship or the engine of a transtainer crane, makes the measurement inaccurate. By using the acoustic camera it is possible to tackle these two problems at the same time. Noise measurements can be made with equipment far from noise source, while distinguishing the contribution from each single source emitting within the sound field. The paper demonstrates this enhanced measurement technique through a set of surveys performed inside French and Italian ports. The ship emissions are analyzed through the acoustic camera for ship on the way, maneuvering inside the harbour, and berthed at wharf. The effectiveness of beamforming technique is discussed, investigating the measurements accuracy, their qualitative and quantitative significance, and the possibility to introduce specific measurement standards for ships based on this methodology.

9:30

2aNS8. A study on acoustic performance of dissipative silencer for fluid-filled pipe system according to acoustic models of absorbing material. Haesang Yang (Seoul National Univ., Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, coupon3@snu.ac.kr), Jongmoo Lee, and Woojae Seong (Seoul National Univ., Seoul, South Korea)

In many industrial systems involving pipe or duct structures, sound transmission modeling is critical to noise treatment and control. A typical noise treatment method applied to pipe structures is to use a silencer. In particular, a dissipative silencer with a sound absorbing material has a noise control ability to reduce noise through successive sound reflections by impedance mismatch while dissipating incident sound energy as heat. In this study, the acoustic performance of a dissipative silencer lined with multi-layered absorbing materials for fluid-filled pipe system is investigated. Herein, the absorbing materials are modeled applying poroelastic and viscoelastic theories. In the analysis, the eigenvalues and eigenfunctions in the dissipative silencer are determined from the rigid wall boundary condition and the interfaces between the absorbing materials, and then the acoustic performance is compared by obtaining the transmission loss using the mode matching method. Also, a parametric study is conducted to investigate the effect of the number and arrangement of layers on the acoustic performance of the silencer according to the absorbing material properties.

Session 2aPAa

Physical Acoustics and Structural Acoustics and Vibration: Acoustics and Elasticity of Consolidated and Unconsolidated Granular Materials I (Hybrid Session)

John Yoritomo, Cochair
Acoustics, Naval Research Lab, 4555 Overlook Ave., SW, Washington, D.C. 20375

Richard Weaver, Cochair
Physics, Univ. Illinois, 1110 W. Green, Urbana, IL 61801

James Bittner, Cochair
Michigan Technological University, P.O. Box 22, Houghton, MI 49931

Chair's Introduction—8:00

Invited Papers

8:05

2aPAa1. Multiscale acoustics in dense granular media; geophysical implications. Xiaoping Jia (Institut Langevin, ESPCI Paris, PSL Univ., Paris, France, xiaoping.jia@espci.fr) and Arnaud Tourin (Institut Langevin, ESPCI Paris, PSL Univ., Paris, France)

We overview sound interactions with dense granular packings, dry, wet or saturated by liquid, in which acoustic waves are served not only as a probe to monitor the fragile and heterogeneous contact networks, but also as a pump to fluidize the medium and trigger the shear instability. First, we describe the linear sound propagation in jammed granular solid: coherent waves and multiple scattering [1], as well as acoustic probing of elastic heterogeneities [2], and plastic rearrangements [3]. Then, we investigate the nonlinear acoustic waves dominated by the hysteresis due to micro-slips at contacts [4] and the huge softening of shear wave velocity in transient elastography [5]. Finally, we discuss the triggering of granular avalanches by high-amplitude ultrasound [6], through the acoustic lubrication of contacts (micro-plasticity) without visible grains motions. This work helps to understand landslides triggered by seismicity. [1] X. Jia, Phys. Rev. Lett. *93*, 154303 (2004). [2] S. Wildenberg *et al*., Europhys. Lett. **115**, 34005 (2016). [3] Y. Khidas and X. Jia, Phys. Rev. Express **85**, 051304 (2012). [4] X. Jia, T. Brunet, and J. Laurent, Phys. Rev. Express **84**, 020301(R) (2011). [5] J. Brum *et al*., Phys. Rev. Eexpress **99**, 042902 (2019). [6] J. Léopoldès, X. Jia, A. Tourin, and A. Mangeney, Phys. Rev. Express **102**, 042901 (2020).

8:35

2aPAa2. Non-equilibrium strain and elastic hysteresis in consolidated granular media. 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 (Inst. of Thermomechanics of the Czech Acad. of Sci., Prague, Czechia)

The physical origin of hysteretic elasticity in consolidated granular media is still debated. The hypothesis of non-equilibrium strain slowly relaxing towards an equilibrium residual strain, dependent on the amplitude of the applied strain, is a possible candidate for the explanation of slow dynamics (conditioning and relaxation) observed in rocks, concrete and other granular media. Starting from relaxation experiments, i.e., the slow recovery of the modulus/velocity to its linear value after the sample has been externally perturbed, we introduce here an analytical expression for the temporal evolution towards equilibrium. The corresponding differential equation is then used to separate hysteresis from classical nonlinearity in both Static and Dynamic Acoustoelastic experiments. The simple theory proposed allows to explain most of the experimental results for sandstones: generation of butterfly loops (hysteresis) in the dependence of velocity on strain, anisotropy effects on shear wave velocities and presence of slow dynamic effects during creep in static testing.

9:05

2aPAa3. Nonlinear elastic properties of granular media: The effects of stress, relative humidity, and grain shape. Jacques Riviere (Penn State Univ., 212 Earth Eng. Sci. Bldg., University Park, PA 16802, jvr5626@psu.edu), Linying Gao, and Parisa Shokouhi (Penn State Univ., University Park, PA)

The long-term goal of this work is to elucidate the microphysical mechanisms responsible for the nonlinear elastic properties of rocks and granular media. Nonlinear elasticity in such media arises from the weak junctions between grains, however the precise microphysical mechanisms at play remain to be determined. In our previous work, we found that (1) in rocks, the nonlinear elastic parameters cluster in two groups, suggesting that there are two main physical mechanisms, (2) despite their differences in grain shape, the elastic nonlinearity of spherical glass beads and angular fine sand is of the same magnitude, however (3) the elastic nonlinearity increases with

relative humidity (RH) for glass beads, while it remains quite constant for fine sand. To further refine these observations and reduce scatter, here, we present preliminary results using a new setup that allows us to quickly vary the relative humidity of a single sample, rather than comparing different samples at different RH. We monitor the nonlinear elastic response of our samples using Dynamic Acousto-Elastic Testing and systematically vary the stress and RH. Interpretation of these new results in light of our previous work are discussed.

9:35

2aPAa4. Reflections on the physics of slow dynamics in rocks. James A. TenCate (Geophys. Group, Los Alamos National Lab., Earth and Environ. Sci., Los Alamos, NM 87545, tencate@mac.com)

Slow Dynamics is a peculiar property exhibited by many sedimentary rocks and was first reported in the mid 1990s. Slow dynamics refers to the process of changing the elastic state of a rock, softening its macroscopic stiffness with application of an AC acoustic drive. What makes slow dynamics unique is that the elastic state of the rock recovers as log(time) once the acoustic drive is turned off. Many creep-like mechanisms have been proposed to explain the effect, notably it resembles recoverable/reversible DC creep. The most common explanation is due to the effects of water, or a form of it, in the spaces between grains. However, it has been exceedingly difficult to pin down the real physics. One experiment with Berea sandstone at high vacuum and temperatures showed that slow dynamics still persists even at these conditions. Followup experiments are in progress, using a pure quartz Fontainebleau sandstone to avoid any effects of clays. Another experiment is planned which will mimic conditions on Mars where water in any form is of great interest to planetary geologists. [Work supported by the Office of Basic Sciences, DOE and by Laboratory Directed Research and Development funding.]

10:05

2aPAa5. Slow dynamics in a single bead with mechanical conditioning and transient heating. Richard Weaver (Phys., Univ. of Illinois, 1110 W. Green, Urbana, IL 61801, r-weaver@illinois.edu) and Sangmin Lee (Civil and Environ. Eng., Univ. of Illinois, Urbana, IL)

The ultrasonically-measured contact stiffness of an aluminum bead confined between two slabs diminishes on mechanical conditioning, and then recovers like log(t) after the conditioning ceases. Here that structure is evaluated for its response to transient heating and cooling, with and without accompanying conditioning vibrations. It is found that, under heating or cooling alone, stiffness changes are mostly consistent with temperature dependent material moduli; there is little or no slow dynamics. Hybrid tests in which vibration conditioning is followed by heating or cooling lead to recoveries that begin like log(t) and then become more complex. On subtracting the known response to heating or cooling alone we discern the influence of higher or lower temperatures on slow dynamic recovery from vibrations. It is found that heating accelerates the initial log(t) recovery, but by an amount more than predicted by an Arrhenius model of thermally activated barrier penetrations. Transient cooling has no discernable effect, in contrast to the Arrhenius prediction that it inhibits recovery. [Supported by the DOE, DE-SC0021056].

Session 2aPAb

Physical Acoustics, Education in Acoustics, and Structural Acoustics and Vibration: Acoustics Demonstration Extravaganza

Daniel A. Russell, Chair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802

Chair's Introduction—9:45

Invited Paper

9:50

2aPAb1. Acoustics demonstration extravaganza. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

The Education in Acoustics and Physical Acoustics committees are pleased to present an Acoustics Demonstration Extravaganza - a showcase of demonstrations and apparatus to inspire and challenge your understanding of acoustics and vibration phenomena. In the spirit of the "Circus of Acoustics" demonstration show from the June 2002 ASA 143 Pittsburgh meeting [*J. Acoust. Soc., Am.*, **111**, 5, Pt 2, p. 2451, (2002) session 4pPAb] this session will consist of several short demonstrations covering a wide variety of acoustics and vibration topics. Among the demonstrations intended to be shown are the following (there may be some last-minute additions or substitutions depending on equipment and presenter availability):

Simple Sound Sources: Directory patterns and frequency spectrum characteristics of monopole, dipole, and quadrupole sources, the pressure doubling effect of a baffle, and a beam-forming array.

Radiation of Sound from Plates: Demonstration of the radiated intensity from various vibrational modes of a simply supported thin plate.

Strings of Pearls: Modes of vibration for a N-DOF mass-spring system and standing waves on a compound string comprised of a two string segments with different densities.

Burning Wire of Death: Standing waves on a nichrome wire carrying a high current so it glows red at nodes.

The Song of the Singing Rod: The singing rod demonstration is commonly used to describe longitudinal waves in a free-free rod. However, if the length of the rod is adjusted just right, a nonlinear pulsating effect is observed resulting from modal coupling of transverse bending modes and either subharmonic longitudinal modes or torsional modes.

Nonlinear Oscillations in a Membrane: A membrane is driven with a large amplitude, so the vibration becomes nonlinear. As the frequency sweeps up and then down, the amplitude jumps and follows a hysteresis loop characteristic of a stiffening nonlinear spring.

Demonstration apparatus of the Cochlea (designed by Robert Keolian): A mechanical hydrodynamic analog model of the basilar membrane in the cochlea illustrating von Békésy's classical passive tonotopic traveling wave as an explanation of how the ear responds to different frequencies.

Acoustic Resonator Rockets: A pair of lightweight Helmholtz Resonators, balanced on a pivot, will rotate when exposed to a loud amplitude sound at their resonance frequency.

Acoustic Filters: An audible and visual demonstration of low-pass, high-pass, and band-stop acoustic filters using PVC pipes.

Vocal Tract Models: A collection of short pipes with varying cross-sectional areas (designed by Takayuki Arai) that produce vowel sounds and may be used to model the acoustics of speech.

Speed of Sound: A series of short demonstrations that illustrate the effect of temperature on the speed of sound in air, sound speed dependence on gas composition, and the effect of bubbles on the speed of sound in water.

Fluid Loading of a Plate: A comparison of the vibration response of a vibrating plate, and the audibly radiated sound, when the plate is in air versus submerged in water.

Rectangular Waveguide: Plane waves will propagate down a waveguide without decay at all frequencies, but non-plane wave modes will propagate only if driven above the cut-on frequency. When driven below the cut-on frequency, non-plane modes become evanescent and decay exponentially.

Baseball Bat Piano: The hollow cylindrical barrels of metal and composite softball bats exhibit cylindrical shell vibrational modes. A collection of softball bats with hoop-mode frequencies corresponding to a musical scale may be played like a xylophone.

Session 2aPP

Psychological and Physiological Acoustics: David Green and Psychoacoustics (Hybrid Session)

William A. Yost, Cochair
College of Health Solutions, ASU, PO Box 870102, ASU, Tempe, AZ 852870102

Robert A. Lufti, Cochair
Univ. of South Florida, Tampa, FL 33620

Chair's Introduction—7:55

Invited Papers

8:00

2aPP1. Dave Green and psychoacoustics. William A. Yost (College of Health Solutions, ASU, PO Box 870102, ASU, Tempe, AZ 852870102, william.yost@asu.edu)

Dave Green's career spanned four decades all devoted to the study of psychoacoustics. It is hard to imagine anyone whose work has had a greater impact on the field. Dave's interest in psychoacoustics began as an Experimental Psychology student at the University of Michigan. He is most widely recognized in the field of psychophysics for his pioneering contributions to Signal Detection Theory (SDT), which he helped develop in the Electronic Defense Group (EDG) of the Department of Electrical Engineering in the early 1950s. The culmination of this work is given in his seminal text "Signal Detection Theory and Psychophysics" (co-authored with John Swets) in 1966. His later work on Profile Analysis in the 1980s helped established a new view of auditory perception. As a consultant for Bolt, Beranek, and Newman (BB&N), Dave provided numerous publications and reports that have served beneficial to society, including his testimony regarding the reenactment of the assassination of John F. Kennedy in 1979. More than 50 students, post-docs, and colleagues, many who have made important contributions to psychoacoustics, have worked directly in Dave's labs and received his mentoring. To say that Dave Green is a psychoacoustic icon would be an understatement.

8:20

2aPP2. A profile of profile analysis. Christopher Conroy (Dept. of Biological and Vision Sci., SUNY College of Optometry, 33 W 42nd St., New York, NY 10036, cconroy@sunyopt.edu)

Throughout the 1980s and early 1990s, David M. Green and his collaborators published a series of papers and a book detailing a view of intensity discrimination that they termed profile analysis. The basic idea was that, in certain situations, masked intensity discrimination is accomplished through an analysis of spectral shape, rather than a successive comparison of intensities in the spectral region of the intensity change. In other words, masker energy in spectral regions far removed from the intensity change can have a profound influence—either positive or negative, depending, among other factors, on the listener's uncertainty and expectations regarding the masker's spectral shape—on intensity-discrimination performance, an idea that diverged sharply from classical models based on a single inflexible channel. Consequently, Green's work on profile analysis both spurred on, and became part of, a broader movement within psychoacoustics that emerged around the same time, in which classical theory and methods began to comingle with higher-level concepts (uncertainty, expectation, etc.). This talk will use Green's own reflections, along with those of his collaborators, to trace the development and theoretical consequences of his work on profile analysis, with a particular emphasis on its relation to this broader historical movement.

8:40

2aPP3. Profile analysis and neural fluctuations: New perspectives on a classic stimulus. Daniel R. Guest (Dept. of Biomedical Eng., Univ. of Rochester, 75 E River Rd., Minneapolis, MN 55455, daniel_guest@urmc.rochester.edu) and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Profile analysis tests the ability to discriminate sounds based on patterns in amplitude spectra. Prior work has largely interpreted profile-analysis data using the power-spectrum model of masking. Under this model, performance relies on analyzing the output of a peripheral bandpass filterbank, and thresholds reflect limits on performance due to frequency selectivity and neural noise. Although this model successfully captures some basic trends in profile-analysis data, it has difficulty explaining others, such as poorer performance at high frequencies. We hypothesize that these trends can instead be explained by midbrain sensitivity to neural fluctuations. Profile-analysis stimuli contain rich temporal modulations, which elicit fluctuations in neural responses that are shaped by the auditory periphery and encoded by average discharge rates in the midbrain. We used physiologically realistic models to simulate midbrain responses to profile-analysis stimuli over a wide range of frequencies, sound levels, and component numbers/spacing. Some features of profile analysis that

are difficult to explain with the power-spectrum model, such as frequency dependence and the effects of hearing loss, were readily accounted for by midbrain tuning to neural fluctuations. These results inform the role of fluctuations and effects of hearing loss on discrimination of complex sounds. [Work supported by NIH R01 DC010813.]

9:00

2aPP4. The ideal observer: The forgotten second half of signal detection theory. Robert A. Lutfi (Univ. of South Florida, Tampa, 4202 E. Fowler Ave., Tampa, FL 33620, rlutfi@usf.edu)

In his seminal text "Signal Detection Theory and Psychophysics" (1966), coauthored with John Swets, Dave Green identifies two parts to the theory. The first entails the psychophysical methodology and procedures required to separate sensory and decision processes in detection, the second compares human performance to that of a theoretical ideal observer that optimizes decisions based on the statistical properties of signals and the demands of the detection task. The first part has received far wider application in psychophysics, but there are demonstrable advantages to applying both. In this talk I will give examples from work in our lab where the analysis of ideal observers framed in the methodology has been used to (1) give mathematical meaning to the constructs of stimulus uncertainty and similarity in informational masking; (2) determine sensory constraints on the use of invariant acoustic cues in sound source identification; and (3) isolate a fixed effect of listeners in multi-talker speech segregation across experiments involving vastly different stimuli and psychophysical tasks. [Work supported by NIDCD grant R01 DC001262.]

9:20

2aPP5. The influence of temporal features of the signal on detectability: Data and signal-detection analyses. Beverly A. Wright (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, b-wright@northwestern.edu) and Huanping Dai (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Dave Green had an enormous influence on auditory research via signal-detection theory. A major contribution of signal-detection theory is the concept and application of the ideal detector, which establishes the absolute optimal detection performance. To bridge between the ideal performance and the less-than-ideal human performance, Green and Swets (1966) described an energy detector. The energy-detector model relaxed assumptions of the ideal detector to account for the phase insensitivity and limited frequency selectivity of humans, but it retained the assumption that the time window is matched to the signal duration. However, the time-window assumption had no clear support from human data. To help fill this gap, we examined the detectability of signals with expected versus unexpected temporal properties. The results of experiments and signal-detection analyses suggest that listeners listen selectively to signal duration, implying that they match the time window to the overall signal duration as was assumed in the energy-detector model. Additional results suggest that listeners do not also listen selectively to the temporal structure of the signal, implying that the temporal constraint is based on overall signal duration. We are grateful for the opportunity to have worked with Dave.

9:40

2aPP6. Temporal pitch processing in an animal model of normal and electrical hearing. John C. Middlebrooks (Otolaryngol., UC Irvine, Rm. 116, Medical Sci. Bldg E, Irvine, CA 92697-5310, j.midd@uci.edu), Matthew L. Richardson, Harrison W. Lin (Otolaryngol., UC Irvine, Orange, CA), and Robert P. Carlyon (MRC Brain and Cognition Unit, Univ. of Cambridge, Cambridge, United Kingdom)

Cochlear-implant users typically exhibit only limited sensitivity to temporal fine structure (TFS). We are developing a cat animal model in which to explore alternative modes of electrical stimulation that might improve TFS sensitivity. Normal-hearing cats were trained to detect changes in rates of acoustical pulse trains. The pulses were non-resolved harmonic complexes that were band-limited to stimulate the basal cochlear turn, which is the location that is accessible to feline cochlear implants. The rate sensitivity of that temporal pitch was like that of human listeners but shifted upward in rate by $\sim$1 octave. Scalp recordings from sedated cats showed a frequency following response (FFR) across all tested rates, up to 600 pulses per second. Changes in pulse rates elicited an Acoustic Change Complex (ACC) that showed a rate sensitivity parallel to that of perception. That suggests that the ACC can be used as a surrogate for psychophysical performance. Cats that were implanted chronically with conventional intra-scalar cochlear implants exhibited electrically-evoked FFR and ACC that showed rate sensitivity largely similar to that in normal-hearing cats. These results validate the cat animal model for ongoing studies of transmission of TFS with electrical cochlear stimulation. [Work supported by the NIDCD and the Wellcome Trust]

10:00–10:20 Break

10:20

2aPP7. Combining signal-detection theory and interaural cross-correlation approaches to account for the binaural abilities of normal-hearing listeners and listeners with "slight" hearing loss. Leslie R. Bernstein (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, Farmington, CT 06032, lbernstein@uchc.edu)

Outstanding among the manifold contributions David M. Green made to psychophysics in general and psychoacoustics in particular was his explication and application of Signal Detection Theory (SDT). Several experimental contexts will be discussed in which a SDT approach to modeling has yielded successful and intuitively appealing accounts of measures of binaural processing. The focus will be, primarily, on relatively recently published empirical data and quantitative modeling from our laboratory. Those reports demonstrate that data obtained in binaural detection experiments conducted across the last five decades can be accounted for by combining a signal-detection-based decision variable with a cross-correlation-based model of binaural processing. Key to the development of the unified account of those experimental results was (1) the inclusion of "internal noise" within stages of the model; and (2) the calculation and inclusion of the variability of the interaural correlations of the outputs of the model for both masker-alone and signal-plus-masker conditions. It will be shown how the SDT-inspired approach has also proven useful in determining and explaining why some listeners with slight, but clinically negligible, elevations in audiometric thresholds exhibit reliable and meaningful deficits in both binaural detection and binaural discrimination tasks. [Work supported by Office of Naval Research (N00014-15-1-2140; N00014-18-1-2473)]

10:40

2aPP8. David Green and the auditory demonstration tapes. William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu)

In 1976, Dave Green received a grant from the National Science Foundation to create a series of auditory demonstrations, intended for use in experimental psychology courses or labs. He assembled a group of collaborators from the Harvard Laboratory of Psychophysics and elsewhere that ultimately created a set of ten cassette tapes with 20 demonstrations. Demonstrations included critical band, asymmetry of masking, binaural beats, temporal integration, categorical speech perception, combination tones, Shepard tones, and periodicity pitch among others. The demonstrations were well chosen to cope with the limitations of the cassette medium. Ten years later, cassette tapes had been replaced by compact discs, and the Eindhoven group with Houtsma, Rossing, and Wegenaars created a similar set of digital demonstrations. Those recordings are now available from the Acoustical Society of America as the "Auditory Demonstrations IPO-NIU-ASA" CD. This talk will compare those two sets of recorded demonstrations and indicate the influence that the Harvard tapes had on the ASA CDs, what was kept, what was changed, what was improved, and what was not.

11:00

2aPP9. Recollections of David M. Green. Ervin R. Hafter (Dept. of Psych., Univ. of California, Berkeley, CA 94720, ErvHafter@gmail.com) and Dennis McFadden (Psych., Univ. of Texas, Austin, TX)

In addition to being wickedly smart, David Green had many characteristics that contributed to his great success as a scientist, scholar, mentor, and friend. ERH and family interacted with the Greens over many decades, during sabbatical years, and in travel to exotic places. DM worked with Dave on two major projects—the acoustical reenactment of the JFK assassination, and an NRC committee studying the effects of intense sounds on marine mammals. Recollections will be shared, some true. Dave's work was marked by an unflagging determination to fully understand the problem at hand, and he was amazingly quick at comprehending new information and organizing it into his extraordinary memory. While current and future generations of scientists will remember him for the paradigm shift that his work produced in experimental psychology and neuroscience, those of us who knew him personally will remember an honest, generous, kind, and unpretentious fellow who was just as knowledgeable about sports, art, and current events as about science. We also will not forget that he was a truly fun guy who enjoyed nothing better than a good joke about the world's foibles or about his own notorious frugality.

11:20

2aPP10. Dave Green as a mentor. Elizabeth A. Strickland (SLHS, Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, estrick@purdue.edu)

In addition to his influence on the field through his science, Dave Green also had a direct impact through his role as a mentor for many students, post docs, and visitors to his labs. In this talk, I will discuss Dave's history as a mentor, and reflect on experiences as the last post doc to come to the lab.

11:40

2aPP11. David Green as a mentor: Recollections from his last Ph.D. student. Jungmee Lee (Commun. Sci. and Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD1017, Tampa, FL 33620, jungmeelee@usf.edu)

David Green's significant contributions to psychoacoustics were already widely recognized when I began as his Ph.D. student. At that time, I was new to this country and had little idea of how really huge his presence was in the field. Working with him was at first difficult, he could be quite intimidating. But slowly, as my training progressed, I began to realize how very lucky I was to be his student. Our relationship continued well after my Ph.D. up until he passed away. I want to share my many memorable stories about him as a lifetime mentor and friend.

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics, Computational Acoustics, and Physical Acoustics: Acoustic Metamaterials I

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnett Rd., Austin, TX 78758

Alexey Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, Bethesda, MD 20817

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Brittany Wojciechowski, Cochair
Wichita State University, 1845 Fairmount St., Wichita, KS 67260

Chair's Introduction—8:00

Invited Papers

8:05

2aSA1. What is an acoustic metamaterial? Christina J. Naify (Appl. Res. Labs., The Univ. of Texas at Austin, 4555 Overlook Ave., SW, Washington, DC 20375, christina.naify@gmail.com), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Alexey Titovich (Naval Surface Warfare Ctr., Carderock Div., Bethesda, MD)

The topic of acoustic metamaterials has been included as a regular special session at the ASA for over 10 years, with hundreds of papers presented. In many of these presentations, the author begins with a definition of 'acoustic metamaterial.' While many of these definitions are similar, subtle differences exist in the defining characteristics. So what is a metamaterial? What are the defining features as accepted by the research community? This presentation will review some of the common themes in the definitions and include opportunity for light-hearted interactive audience participation to evaluate those defining themes.

8:35

2aSA2. Self-adapting band gaps in rotating elastic metamaterials. Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801, kmatlack@illinois.edu) and Ignacio Arretche (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Rotation in mechanical systems can drastically change their vibrational response. For example, it can enable new properties, such as break time-reversal symmetry and induce non-reciprocity in acoustic metamaterials, due to Coriolis forces and gyroscopic effects. Rotation can also be highly detrimental: for example, synchronous vibrations occur in rotating mechanical components and can be extremely damaging when they coincide with system resonances. In addition, centrifugal forces from rotation significantly affect the dynamics of rotating systems, through stress stiffening and spin softening, however their effects on acoustic and elastic metamaterials have rarely been explored. Here, we demonstrate that centrifugal forces in rotating elastic metamaterials enable band gaps that "self-adapt" to synchronous vibrations, attenuating these vibrations over a broadband frequency range. Specifically, this talk presents a reduced order model that captures stress stiffening effects on the band gaps of a rotating elastic metamaterial. The resonators are beams with tip masses oriented perpendicular to the rotational axis, such that stress stiffening causes their resonant frequency to be linearly proportional to the rotational speed. The reduced order models are validated by full finite element simulations of a 3D elastic metamaterial whose band gap self-adapts to synchronous vibrations that are linearly proportional to rotational speed.

8:55

2aSA3. Achieving any desirable dispersion curves using non-local phononic crystals. Arash Kazemi, Kshiteej Deshmukh, Fei Chen, Yunya Liu (Mech. Eng., Univ. of Utah, Salt Lake City, UT), Bolei Deng (MIT, Salt Lake City, UT), Xuan Zhu (Civil and Environ. Eng., Univ. of Utah, Salt Lake City, UT), Henry Fu (Mech. Eng., Univ. of Utah, Salt Lake City, UT), and Pai Wang (Mech. Eng., Univ. of Utah, 1495 E 100 S., MEK Bldg., Salt Lake City, UT 84112, pai.wang@utah.edu)

Phononic crystals and vibro-elastic metamaterials are characterized by their dispersion relations—how frequency changes with wave number/vector. While there are many existing methods to solve the forward problem of obtaining the dispersion relation from any arbitrarily given design. The inverse problem of obtaining a design for any arbitrarily given dispersion bands has only had very limited success so far. Here, we report a new design scheme for arbitrary dispersion relations by incorporating non-local interactions between unit cells. Considering discrete models of one-dimensional mass-spring chains, we investigate the effects of both local (i.e., springs between the nearest neighbors) and non-local (i.e., springs between the next nearest neighbors and other longer-range springs) interactions. First, we derive the general governing equations of non-local phononic chains. Next, we examine all design constraints for a linear, periodic, passive, statically stable, non-gyroscopic, and free-standing system. Finally, we perform analytical calculations and numerical simulations to solve the inverse problem. The results illuminate a new path toward novel wave manipulation functionalities, such as ordinary and higher-order critical points with zero-group-velocity (ZGV) modes, as well as multi-wavelength and multi-speed propagations of the same mode at the same frequency.

Contributed Papers

9:15

2aSA4. Contactless manipulation of subwavelength-structured acoustic metasurfaces. Samuel Keller (Mech. Eng., Univ. of Minnesota-Twin Cities, 111 Church St. SE, Minneapolis, MN 55455, kell2521@umn.edu), Matthew Stein, Yujie Luo, and Ognjen Ilic (Mech. Eng., Univ. of Minnesota-Twin Cities, Minneapolis, MN)

The use of acoustic waves to actuate materials without physical contact is relevant in a variety of fields, including assembly, imaging, and holography. Current acoustic manipulation techniques can produce high-precision, non-invasive and non-contact forces, but they are generally limited to small (subwavelength) objects due to the nature of acoustic trapping potentials. In this study, we describe an approach for overcoming these limitations by using metasurfaces to control the refraction of sound waves. We present theory, simulations, and experimental results showing how a deliberately engineered metasurface can steer the momentum of sound waves and therefore control the intensity and the direction of acoustic radiation pressure. To illustrate the potential applications of this concept, we demonstrate acoustic metasurfaces which exhibit dynamical phenomena such as self-guidance and contactless acoustic pulling. We expand this analysis to survey how this concept can be implemented in different metasurface unit cell topologies and how it scales with different acoustic frequencies. Our results combine actuation with acoustic metamaterial physics to provide novel degrees of freedom to control acoustic forces beyond the limits of traditional wave-matter interactions. Nat. Commun. **13**, 6533 (2022).

9:30

2aSA5. Surface wave suppression in granular media using a partially embedded meta-barrier. Lalith Sai Srinivas Pillarisetti (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg., University Park, PA 16802, ljp5518@psu.edu), Cliff Lissenden, and Parisa Shokouhi (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Existing studies on meta-barrier designs for suppressing surface modes in inhomogeneous granular media are limited to meta-barriers comprising vertical or horizontal oscillators. However, a clear understanding of the surface mode hybridizations with the local resonances of the resonators is lacking and could help realize more realistic meta-barrier designs for surface wave control in granular media. In this work, we propose a meta-barrier comprising partially embedded rod-like resonators to study the hybridization of the fundamental surface modes: PSV_1 and PSV_2, with the longitudinal and flexural resonances of the resonators that enable surface mode suppression. The hybridized dispersion curves and frequency-domain finite element simulations with the meta-barrier demonstrate preferential hybridization of the PSV_1 mode with the longitudinal resonance of the resonators, enabling PSV_1 mode suppression above the hybridized longitudinal-resonance mode cut-off frequency. Unlike the PSV_1 mode, PSV_2 hybridizes with both longitudinal and flexural resonances, making it challenging to suppress the mode. However, we observe that enhancing the filling fraction of the meta-barrier promotes PSV_2 mode suppression as the flexural-resonance hybridized modes are localized under the meta-barrier. The time-domain finite element simulation using a broadband excitation source further numerically validates the meta-barrier design for applications in vibration mitigation and seismic isolation of structures.

9:45

2aSA6. Analysis of ideal models of real nonlocal acoustic and elastic metamaterials. Nathan Geib (Appl. Res. Labs., The Univ. of Texas at Austin, 1587 Beal Ave., Apt. 13, Ann Arbor, MI 48105, geib@umich.edu)

Nonlocal coupling in acoustic and elastic systems has received growing attention for its potential to expand the capabilities of active metamaterials. Previously, we have shown experimentally how nonlocality imposed through action at a distance, where signals from disturbances at one location are used to drive actuators at separate locations, gives rise to unique scattering characteristics, namely highly nonreciprocal transmission. We showed how the mathematical models for these experimental systems were excellent predictors for real system behavior, and as such, represented powerful experimental design tools. However, these models were elaborate, incorporating the characteristics of real sensors and actuators, electronic controllers, and material properties and geometries, consequently obscuring the underlying nonlocal physics responsible for their unique behavior. Here, we present simplified models of these real systems that feature closed form expressions for system scattering characteristics, allowing for the identification of relevant dimensionless quantities and revealing how the interactions of those quantities affect both system performance and stability. We compare and contrast ideal models for both acoustic and elastodynamic nonlocal systems, highlighting how each ideal model is qualitatively representative of its real counterpart, thus providing a pathway to a deeper understanding and further exploration of nonlocality in wave-bearing systems.

10:00–10:15 Break

10:15

2aSA7. Multiple scattering model for collections of bianisotropic acoustic scatterers. A. J. Lawrence (Appl. Res. Labs., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, ajlawrence@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The magnitude of an acoustic point source is traditionally described as monopole, dipole, or higher-order multipole source strengths. The polarizability tensor, commonly used in electromagnetics, can be used to describe the scattering from an acoustically small heterogeneity in a background medium as monopole, dipole, and higher order multipoles due to the pressure and particle velocity at the scattering site. Recent research on bianisotropic acoustic (or Willis) metamaterials has employed the acoustic polarizability tensor as a convenient descriptor of the scattered fields from scatterers with asymmetries since non-zero off-diagonal terms indicate the existence and relative strength of Willis coupling. In the present work, we employ the polarizability tensor in a multiple scattering formulation for N arbitrary, point scatterers. The polarizability tensor for an example scatterer is extracted from finite element simulations by generalizing the approach of Su and Norris [Phys. Rev. B **98**, 174305, (2018)]. We then provide a metasurface example to demonstrate the ability of the multiple scattering approach to consider variability in scatterer position and orientation in the calculation of the total scattered field.

10:30

2aSA8. Source-driven homogenization theory for electro-momentum coupled scatterers. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of at Austin, 10000 Burnet Rd., Austin, TX 78758, cgokani@arlut.utexas.edu), Samuel P. Wallen, Mark F. Hamilton, and Michael R. Haberman (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of at Austin, Austin, TX)

Willis materials are metamaterials whose subwavelength asymmetry couples the macroscopic pressure-strain and momentum-velocity relations. Recently, the design space of these metamaterials has been expanded to consider asymmetric piezoelectric scatterers and thereby couple the electric field-electric displacement relation to the constitutive equations in the Willis form. The existence of this so-called electro-momentum coupling was first predicted using dynamic homogenization of heterogeneous piezoelectric media [J. Mech. Phys. Solids **134**, 103770 (2020)] but it can also be understood through a generalized polarizability tensor α that calculates the electro-acoustic field scattered from a single asymmetric piezoelectric inhomogeneity as a function of local fields [Proc. Mtgs. Acoust. **46**, 065002 (2022)]. We present a multiple-scattering dynamic homogenization method that extends the work of Sieck *et al*. [Phys. Rev. B **96**, 104303 (2017)] using the generalized polarizability tensor. The model considers the microscale scattered pressure, velocity, and electric fields in a one-dimensional periodic lattice of identical scatterers to find analytical expressions for the effective macroscale fields. The macroscale fields are then used to find the effective electro-momentum coupling constants in terms of the polarizability of the individual scatterers and the concentration of scatterers in a background medium. [Work supported by DARPA and ARL:UT McKinney Fellowship in Acoustics.]

10:45

2aSA9. Computational analysis of sub-wavelength scatterers exhibiting electro-momentum coupling. Samuel P. Wallen (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu), Benjamin M. Goldsberry (Appl. Res. Labs. at The Univ. of Texas at Austin, Austin, TX), Chirag A. Gokani, and Michael R. Haberman (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

The macroscopic response of acoustic metamaterials with sub-wavelength asymmetry may be described by the so-called Willis constitutive relations, which include coupling between the acoustic pressure and momentum density. One method to describe the behavior of acoustically-small, Willis material building-blocks is via a polarizability matrix relating the monopole and dipole scattering moments to the local pressure and velocity fields, providing a metric for the design of microscale structures leading to macroscopically observable Willis coupling. Additionally, heterogeneous, piezoelectric media with sub-wavelength asymmetry have been shown to exhibit coupling between momentum density and electric fields, a material response known as electro-momentum coupling. Recent studies have generalized the polarizability concept to include coupling between acoustic and electromagnetic fields for theoretical point scatterers in two and three dimensions, and derived analytical bounds based on conservation of energy. In this work, we computationally study the coupled, acousto-electromagnetic polarizability of heterogeneous, piezoelectric scatterers and assess the practicability of achieving electro-momentum coupling in physically realizable geometries. Numerical results are obtained using a custom finite element approach, which includes the fully electrodynamic nature of the system and accounts for large differences in the characteristic wavelengths of the acoustic and electromagnetic fields. [This work was supported by DARPA.]

11:00

2aSA10. Experimental demonstration of a spatiotemporally modulated sound diffusing metasurfaces. Janghoon Kang (Mech. Eng., The Univ. of Texas at Austin, Walker Dept. of Mech. Eng., 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, jh3010.kang@utexas.edu) and Michael R. Haberman (Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Spatiotemporal modulation of material properties has been studied across physics in an effort to control propagating waves. Using a semi-analytical approach, we have recently shown that the performance of a conventional sound diffuser can be significantly improved through spatiotemporal modulation of its surface admittance [Kang *et al*., Appl. Phys. Lett. **121**, 181703 (2022)]. The improved performance is a result of energy in scattered sound at the incident wave frequency plus and minus harmonics of the modulation frequency of the temporal modulation of the input admittance. The additional frequencies of reflected sound increase the diversity of propagating diffraction modes and therefore increased angular distribution of scattered energy. This work presents the design of a sound diffuser element to implement the spatiotemporal modulation. The rigid bottom of a straight-tube diffuser well is replaced by a piezo disc shunted by a negative impedance converter (NIC), in which a capacitance can be controlled by varying a resistor value in the NIC. We present impedance tube measurements that show the presence of modulation harmonics in the reflected wave for single-frequency incident wave caused by a modulation of an acoustic admittance, and the diffusion performance measured in the anechoic chamber is compared with the non-modulated case.

11:15

2aSA11. Generative optimization of sound insulation of composites using deep learning networks. Sheng Sang (Eng. Sci., Bethany Lutheran College, 700 Luther Dr., Mankato, MN 56001, ssang@blc.edu) and Chen Xu (Comput. Sci., The Univ. of Tennessee at Chattanooga, Chattanooga, TN)

Composites have been widely used in the field of acoustics due to their extraordinary ability of sound insulation. To date, the design of acoustic composites relies primarily on the expertise of engineers and experimental tests. This preliminary study outlines a deep learning (DL) based approach to optimize the microstructure of the composite bars to achieve the best performance in sound insulation. This approach first trains DL networks using data generated using finite element simulation to predict the pressure amplitude and energy of the output waves, then a genetic algorithm (GA) uses the DL model as its evaluation function and generates new designs. The results indicate that a combination of DL and GA can generate Pareto-optimal designs to satisfy the specific needs of engineering projects. We demonstrate that starting with a quite small data set (less than of all possible designs) and applying a DL approach is an efficient and robust method to obtain optimal designs. The DL model is accurate in its predictions which enables the GA to find unique composite designs that are optimal for sound insulation.

11:30

2aSA12. Chirality-selective transmission of acoustic orbital angular momentum through lossy metasurfaces. Jun Ji (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., Graduate Program in Acoust., State College, PA 16802, jbj5431@psu.edu), Junfei Li (Elec. and Comput. Eng., Duke Univ., Durham, NC), Nikhil JRK Gerard (Acoust., Penn State Univ., Raleigh, NC), Xiuyuan Peng, Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Wave fields with acoustic orbital angular momentum have drawn great attention because of its wide range of applications such as acoustic communication and particle manipulation. We designed, fabricated, and

experimentally validated a type of lossy metasurface which enables an extremely selective transmission of acoustic wave filed depending on the chirality of its orbital angular momentum. The metasurface is composed of a phase gradient metagrating (PGM) and microperforated panel (MPP). By harnessing the multiple internal reflection of PGM and the carefully engineered loss of MPP, the left-handed incident vortex has a near-zero power transmission while its right-handed counterpart has a 60% power transmission. Such an extremely selective transmission was experimentally validated by retrieving the full scattering matrix using a generalized twelve-microphone method, which shows a good agreement with numerical simulations.

11:45

2aSA13. Bandstructure of phononic crystal of rotating heterogeneous cylinders: Time periodicity due to pure mechanical modulation. Dmitrii Shymkiv (Phys., Univ. of North Texas, 1155 Union Circle #311427, Denton, TX 76203, dmytroshymkiv@my.unt.edu), Matthew Li, and Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX)

Existing designs of elastic spatiotemporal periodic structures usually require electric/piezoelectric elements that provide temporal modulation of elastic properties. While this method has advantages in many aspects, it usually allows relatively low modulation depth. Here we propose a design for spatiotemporal elastic media that is experimentally feasible, involves pure mechanical elements, and may provide any desired temporal modulation depth. In our scheme the scatterers of a phononic crystal are cylinders composed of two (or more) semi-cylinders (parts) of different materials. The cylinders are capable for rotation with high speed in a stationary fluid environment that provides time-dependent medium for propagating sound wave. Adjusting the elastic contrast between the materials of solid semi-cylinders the necessary modulation depths can be achieved. Different regimes of temporal modulation, from adiabatic to rapid oscillations can be realized by varying the frequency of rotation. We report momentum/energy gaps and other features of the bandstructure that are theoretically obtained using plane wave expansion. Nonreciprocal effects are studied for a phononic crystal with a square unit cell containing two heterogeneous cylinders rotating synchronically with a constant phase shift. [This work is supported by the NSF under EFRI Grant No. 1741677.]

TUESDAY MORNING, 9 MAY 2023 LOS ANGELES/MIAMI/SCOTTSDALE, 9:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication and Psychological and Physiological Acoustics: Clear Speech(es) Across People, Places, and Time I

Rajka Smiljanic, Cochair
Univ. of Texas at Austin, 305 E. 23rd St., B5100, Austin, TX 78712

Georgia Zellou, Cochair
UC Davis, 1 Shields Ave., Department of Linguistics, Davis, CA 95616-3086

Invited Papers

9:00

2aSC1. Clear speech processing benefits beyond intelligibility. Rajka Smiljanic (Univ. of Texas at Austin, 305 E. 23rd St., B5100, Austin, TX 78712, rajka@austin.utexas.edu)

A robust clear speech intelligibility benefit for a variety of talkers, listeners, and communication challenges is well-documented. In this talk, I will review research that focuses on how conversational to clear speech modifications facilitate linguistic processes and cognitive functioning beyond word recognition in noise. In one line of work, using a visual-world paradigm, we showed that clear speech enhanced speech segmentation and reduced lexical competition. In another, we showed that clear speech benefit extended to the improved sentence recognition memory and recall of words and sentences. Finally, in a series of experiments using a dual-task paradigm, we showed that hearing clear speech increased reaction times on a concurrent visual task suggesting that the clear speech processing benefits may arise through the increased engagement of the attentional resources toward the more salient hyperarticulated speech. The results contribute evidence that clear speech facilitates signal-dependent sensory processing as well as deeper linguistic processing abstracted from the input speech. These clear speech findings have implications for our understanding of perceptual mechanisms that underlie improved speech perception, including the use of cognitive resources and listening effort.

9:20

2aSC2. Audio-visual clear speech: Articulation, acoustics and perception of segments and tones. Yue Wang (Linguist., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A1S6, Canada, yuew@sfu.ca), Allard Jongman, and Joan Sereno (Linguist., Univ. of Kansas, Kansas City, KS)

Research has established that clear speech with enhanced acoustic signal benefits segmental intelligibility. Less attention has been paid to visible articulatory correlates of clear-speech modifications, or to clear-speech effects at the suprasegmental level (e.g., lexical tone). Questions thus arise as to the extent to which clear-speech cues are beneficial in different input modalities and linguistic domains, and how different resources are incorporated. These questions address the fundamental argument in clear-speech research with respect to the trade-off between effects of signal-based phoneme-extrinsic modifications to strengthen overall acoustic salience versus code-based phoneme-specific modifications to maintain phonemic distinctions. In this talk, we report findings from our studies on audio-visual clear speech production and perception, including vowels and fricatives differing in auditory and visual saliency, and lexical tones believed to lack visual distinctiveness. In a 3-stream study, we use computer-vision techniques to extract visible facial cues associated with segmental and tonal productions in plain and clear speech, characterize distinctive acoustic features across speech styles, and compare audio-visual plain and clear speech perception. Findings are discussed in terms of how speakers and perceivers strike a balance between utilizing general saliency-enhancing and category-specific cues across audio-visual modalities and speech styles with the aim of improving intelligibility.

9:40

2aSC3. Clear speech adaptations and effortful speaking and listening in background noise across the lifespan. Outi Tuomainen (Univ. of Potsdam, Karl-Liebknecht-Straße 24-25, Potsdam 14476, Germany, tuomainen@uni-potsdam.de), Linda Taschenberger, Stuart Rosen, and Valerie Hazan (Univ. College London, London, United Kingdom)

Our study investigates strategies speakers use to clarify their speech and compensate for the impact of background noise in everyday settings. We recorded 114 individuals (8-80 years) while they carried out an interactive problem-solving task (diapix) in quiet, background speech and background non-speech noise. A secondary bell-press task was added to increase cognitive load. We measured (i) articulation rate, median fundamental frequency (f0) and intensity which reflect clear speech adaptations and speaking effort; (ii) subjective ratings of listening effort; and iii) secondary task performance reflecting distractibility (d-prime). Regardless of age, speakers increased their f0 and intensity more in non-speech noise than in quiet or background speech. However, they made more errors in the secondary task and reported greater listening effort when the background noise was speech. Older participants (>50 years) increased speaking effort in both noise types, but younger children (<13 years) increased speaking effort only in background speech. These results indicate that, in terms of speaking effort, background speech (more cognitively demanding) is effortful for children and older adults. However, background noise (more peripheral interference) is mostly effortful for older adults. In terms of listening effort and distractibility, we observed greater interference in background speech regardless of participant age.

10:00

2aSC4. Clear speech research in dysarthria. Kaila L. Stipancic (Communicative Disord. and Sci., Univ. at Buffalo, 114 Cary Hall, Buffalo, NY 14214, klstip@buffalo.edu) and Kris Tjaden (Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY)

A sizeable group of theoretically grounded studies has defined clear speech production for neurotypical talkers as well as the perceptual consequences of neurotypical clear speech. Although less robust in size compared to the neurotypical literature, clear speech research focused on the clinical population of dysarthria has grown substantially over the last decade. The growth stems, in part, from the positive effects of clear speech on intelligibility, a perceptual construct central to the clinical management of dysarthria. We will review dysarthria research from our lab examining the effects of clear speech on speech acoustics and subsequently, on perceptual measures of speech. Factors which may contribute to variability in clear speech outcomes in dysarthria will be considered. These factors include but are not limited to (1) dysarthria neuropathology; (2) the instructions provided to elicit a clear speaking style; (3) the amount of effort exerted by the speaker; and (4) the choice of measures for evaluating efficacy. Finally, we will offer suggestions for future research that may accelerate translation of clear speech research to clinical practice.

10:20–10:40 Break

10:40

2aSC5. Dual-task costs of clear and loud speech: Implications for talkers with Parkinson disease. Jason A. Whitfield (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health & Human Services, Bowling Green, OH 43403, jawhitf@bgsu.edu)

Decades of research suggest that the speech production adjustments associated with adopting a clearer or louder speaking style enhance speech intelligibility, making these speech styles a common talker-oriented strategy in speech production interventions for talkers with dysarthria secondary to Parkinson disease (PD). Recent data suggest that explicit instructions to speak louder or clearer than usual may increase the processing load and effort associated with speech production. Empirical data will be presented that demonstrate these higher-effort speech styles may require greater attentional resources than habitual or conversational speech. A group of neurotypical talkers performed a sentence repetition task using Habitual, Loud, and Clear speech styles in isolation and while performing a concurrent visuomotor tracking task. Lip and jaw kinematics, speech intensity, and accuracy on the visuomotor task were collected to quantify performance on each task. Group-level data suggest that when faced with the competing task demands of the visuomotor task, neurotypical talkers prioritized the articulatory and phonatory adjustments associated with Clear and Loud speech at the cost of secondary task performance. Case examples from talkers with hypokinetic dysarthria resulting from PD will also be examined to frame and discuss clinical implications and considerations.

11:00

2aSC6. Adaptative and non-adaptive phonetic variation: Clear speech and second-language speech. Ann Bradlow (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

Two independent lines of research have revealed a somewhat surprising convergence between clear speech and second-language (L2) speech. Specifically, relative to conversational L1 speech, both clear speech and L2 speech typically exhibit slower speaking rates (fewer syllables per second), longer and more frequent inter-word pauses, and less segment- and syllable-level reduction. These phonetic features result in a general pattern of lower information density (more phonetic material produced for a given linguistic message or text) compounded by longer utterance durations to yield substantially lower information transmission rates for both clear speech and L2 speech relative to conversational L1 speech. However, clear speech and L2 speech diverge in terms of intelligibility. While clear speech is a highly effective strategy for enhancing overall intelligibility, L2 speech is typically characterized by greater likelihood of listener errors in word recognition accuracy. This pattern of convergent phonetics and divergent intelligibility highlights the complicated relationship between phonetic form and communicative efficiency. While clear speech is a listener-oriented talker adaptation to listener limitations, L2 speech reflects talker-oriented talker-listener misalignment. The phonetic similarities between these adaptative (clear speech) and non-adaptive (L2 speech) sources of variation underscore the dynamics of human speech communication with the talker-listener relation as the fulcrum.

11:20

2aSC7. Non-native clear speech: The roles of proficiency and speech target. Melissa M. Baese-Berk (Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu) and Misaki Kato (Linguist., Univ. of Oregon, Eugene, OR)

Previous work has demonstrated that native talkers are able to enhance intelligibility by using a speaking style typically referred to as "clear speech." However, it is less clear whether talkers who are speaking a language other than their native language are also able to produce speech in a style that results in intelligibility gains for listeners. In this talk, we will present a series of studies investigating the role of proficiency (higher versus lower proficiency) and target of clear speech (individual segments versus global properties of speech). The work suggests that both lower- and higher-proficiency non-native talkers reliably produce clear speech, but lower-proficiency talkers do not reliable produce a casual or reduced speaking style, and non-native talkers. Further, both groups of non-native talkers are more successful at producing clear speech when the target is a single segment, though this varies as a function of their experience with the particular segment in question. We will present both production and perception data as evidence for these claims. We will conclude by discussing the role of the listener and their familiarity with talkers and accents in our understanding of non-native clear speech.

11:40

2aSC8. Clear speech in the new digital era: Speaking and listening clearly to voice-AI systems. Georgia Zellou (UC Davis, Davis, CA, gzellou@ucdavis.edu) and Michelle Cohn (UC Davis, Davis, CA)

It's a new digital era: many people regularly talk to voice-activated artificially intelligent (AI) personal assistants, such as Siri, Alexa, and Google Assistant, that spontaneously and more naturalistically produce interactive speech. Human speech patterns toward these new voice-AI interlocutors can serve as a "Turing" test for discovering how people adapt their production and perception to improve communication during spoken language interactions. In this research program, we explore how interactions with voice-AI can influence human speech patterns during short-term interactions, and the potential that this has to lead to sound change within speech communities. In this talk I will present two case studies representing the types of studies we are conducting to explore how voice-AI interactions influence human speech patterns. First, we investigate how speakers adapt their speech when talking to devices, particularly to be better understood where the devices display comprehension difficulties. Second, we examine the intelligibility of text-to-speech (TTS) and how listeners can adapt their listening strategies during interactions with devices to better understand. Our work demonstrates that examining people's speech behavior when interacting with voice-AI can serve as a test to our scientific understanding of speech communication, language use, and even linguistic change.

Session 2aSP

Signal Processing in Acoustics: Signal Processing Potpourri

Bernice Kubicek, Cochair
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Chair's Introduction—8:40

Contributed Papers

8:45

2aSP1. Data-driven morphological component analysis. Hannah Kurdila (Penn State Univ., 10903 New Hampshire Ave., Silver Spring, MD 20993, hrk5147@psu.edu), Geoff Goehle, and Daniel C. Brown (Penn State Univ., State College, PA)

A common goal in signal processing is to decompose a signal into constituent components. Morphological Component Analysis (MCA) is a convex optimization technique that can be used to decompose a signal into a sum of sparse representations determined by chosen dictionaries. Previously, MCA has been shown to successfully decompose sonar data into long-duration and short duration components, using Enveloped Sinusoid Parseval (ESP) frames as the dictionaries. However, in its current state, ESP MCA is model-driven and requires prior selection of envelope parameters. This presentation describes the addition of hyper-parameter tuning to ESP MCA. Specifically, gradient descent is performed on an unrolled version of the ESP MCA regularization algorithm in order to learn the optimal envelope parameters used to construct the ESP frames. Addition of hyper-parameter tuning allows the dictionary atoms to adapt and thus improves sparse representation of the sensed data.

9:00

2aSP2. Hybrid analog-sampled adaptive active noise control. Yoav Vered (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, v.yoav@soton.ac.uk) and Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom)

Feedforward adaptive controllers are widely used for active noise control. For good broadband performance, the secondary path delay must be shorter than that of the primary path. In this work, a hybrid analog-sampled adaptive feedforward controller is developed to eliminate the additional delay associated with the sampling. An adaptive finite-impulse-response system is used for the sampled controller, and a state filtered adaptive linear combiner is used for the analog controller. Two approximations are used for the linear combiner. The first minimizes the added electrical noise by having individual state filters, all driven by the reference signal, while the second gives a superior bandwidth by driving each component filter with the output of the previous one. A sampled version of the normalized projection algorithm is developed for the analog controller. The different error paths used to filter the reference signal for the controllers' adaptation are emphasized. Finally, the performance of the proposed controller is assessed. The results highlight the capabilities of the suggested controller and the different state filters, especially their ability to achieve significant attenuation levels close to the causality limit and when long delays of the reference signal are required.

9:15

2aSP3. Micro-multipath interference and its effect on sub-minute acoustic fluctuations: Observations of mid-frequency acoustic propagation in the upper ocean. Jacob P. DeFilippis (Scripps Inst. of Oceanogr., Univ. California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, jdefilip@ucsd.edu), Bruce Cornuelle, Andrew J. Lucas (Scripps Inst. of Oceanogr., Univ. California San Diego, La Jolla, CA), William Hodgkiss (Scripps Inst. of Oceanogr., Univ. California San Diego, San Diego, CA), Luc Lenain, William Kuperman, and Matthew Alford (Scripps Inst. of Oceanogr., Univ. California San Diego, La Jolla, CA)

An experiment was conducted to investigate how variability in the acoustic channel impulse response at signal processing timescales of seconds to minutes is related to internal-wave-driven changes in the ocean sound speed field on timescales of tens of minutes. We find that vertical fine-scale structures ($\leq$10 m) embedded in the sound speed profile have strong second derivatives which allow for the formation of acoustic caustics as well as acoustic propagation pathways that interfere with each other (micro-multipaths). Both effects are sensitive to vertical displacement by internal waves, which can therefore produce sub-minute acoustic fluctuations from internal waves that have timescales longer than a minute.

9:30

2aSP4. Selective preconditioning of an active sound control system with virtual sensing. Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom) and Yoav Vered (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, v.yoav@soton.ac.uk)

The convergence speed of multichannel adaptive active noise control algorithms can be improved if the reference signals are preconditioned, so it is advantageous to whiten them and reduce their cross-correlations. However, this preconditioning depends on the properties of the reference signals, which will generally change over time. It may still be possible to approximate the properties of the measured reference signals using a limited number of exemplar cases. By classifying the measured reference signals into the closest of these cases, the most appropriate preconditioning can be selected for the adaptive algorithm at any one particular time. Also, one method of implementing virtual sensors is to use the additional filter method, which provides targets for the measured error signals to follow, generated from the reference signals. The performance of this virtual sensing is known to be degraded if the properties of the reference signals change, so this classification method could also be used to select the most appropriate additional filter.

9:45

2aSP5. Trust-worth multi-representation learning for audio classification with uncertainty estimation. Kele Xu (National Key Lab. of Parallel and Distributed Processing (PDL), 107, Yanwachi, Changsha 410073, China, kelele.xu@Gmail.com), Kang You, Ming Feng (Tongji Univ., Shanghai, China), and Boqing Zhu (National Key Lab. of Parallel and Distributed Processing (PDL), Changsha, China)

Multi-view learning has been explored for audio classification tasks, exploiting different representations of audio signals, ranging from MFCC, CQT, to raw signals. The quality of each view may vary for different audio signals, and the appropriate uncertainty quantification for each view has not been fully explored. In this work, we explore a trusted multi-view learning framework for classification tasks in order to fully incorporate different views. Our framework consists of three parallel branches of Transformer architectures (Gammatone spectrogram, log-Mel and CQT) and they are combined using the uncertainty estimation of different branch. In addition to computing the classification probabilities, the uncertainty of each representation can also be obtained using the framework. We firstly calculate the evidence based on feature vectors to obtain the probabilities and the uncertainty of classification problems for Gammatone, log-Mel and CQT branch. By integrating the confidence from each of the different representations using the Dempster–Shafer theory, the classification framework can provide higher accuracy and confidence. To demonstrate the effectiveness of the proposed framework, we conduct the experiments on the GTZAN dataset. The obtained results show that our method can reach the accuracy of 83.0%, which significantly outperforms single representation-based methods while providing uncertainty estimation for different views.

10:00

2aSP6. Auditory-inspired adaptive frequency tracking. Vijay Peddinti (Naval Undersea Warfare Ctr. Div. (NUWC), Howell St., Newport, RI 02841, vijaykumar.peddinti.civ@us.navy.mil)

One of the best frequency trackers to date is the human (mammalian) auditory system, which has evolved through millions of years of 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 features of the auditory system could be beneficial in developing superior frequency tracking and classification algorithms. An auditory inspired adaptive synchrony capture filterbank (SCFB) signal processing architecture for tracking signal frequency components was proposed in a related paper [JASA (2013)]. The SCFB architecture consists of a fixed array of traditional, passive linear, gammatone filters in cascade with a bank of three adaptively tunable bandpass filters that form a frequency-discriminator-loop (FDL). The SCFB exhibits many desirable properties for processing speech, music, and other complex sounds. In recent work (Dec 2021), the algorithm was modified using adaptive tuning parameters, and a generalized way to determine/suppress voiced and unvoiced (silent) regions. This modified algorithm estimates frequencies with higher accuracy even in the presence of closely spaced input tones. Preliminary analysis with synthetic, human speech and humpback/whale-call signals demonstrates that the revised algorithm performs well. This talk will focus on the latest updates.

10:15–10:30 Break

10:30

2aSP7. North Atlantic right whale calls monitoring: Signal processing meets deep learning. Abhilash Durgam (Elec. and Comput. Eng., Univ. of Maine, Orono, ME), Zafaryab Haider (Elec. and Comput. Eng., Univ. of Maine, Barrows Hall, Orono, ME 04469, zafaryab.haider@maine.edu), Samuel Aremu, Mersedeh Najishabahang, and Mahdi Al-Badrawi (Elec. and Comput. Eng., Univ. of Maine, Orono, ME)

The monitoring of marine mammals, such as North Atlantic Right whales (NARW), via their acoustic signatures, can contribute to the protection of these endangered species. The monitoring via passive acoustics, however, may deteriorate, due to the propagation properties of the underwater channel, whereas these signals are susceptible to noise masking and/or mixing as they travel long distances. A day (July 30th, 2018) of NARW annotated recordings (Provided by DCLDE), including seven imbalanced call types, were explored for detection, classification, and density estimation purposes. The call types involved right whale upcall, gunshot, moan, scream, minke whale pulse train, low-frequency pulse, and low-frequency down sweep. Classifying these imbalanced call types represented a real challenge for classical signal processing approaches and conventional deep learning networks (e.g., Convolutional Neural Networks). The remedy to these challenges involved data preprocessing (spectrogram cleaning/sifting) and the use of a state-of-the-art deep learning transform, self-supervised, that accounts for the imbalances in the dataset. The preprocessed spectrograms boosted the classification score at the deep learning network. The accuracy results for the training set exceeded 90% while the accuracy scored 81% for the testing set.

10:45

2aSP8. Using machine learning to determine the best wavelet for compressing acoustic data. Avery C. Landeche (Phys., Univ. of New Orleans, 644 Whitney Dr., Slidell, LA 70461, aclandec@uno.edu), Shaun Pies, Kendal Leftwich, and Juliette W. Loup (Phys., Univ. of New Orleans, New Orleans, LA)

This research uses acoustic data to determine the best method to find the optimal wavelet to use for data compression. The power spectral densities before and after wavelet decomposition, compression, and decomposition (decompression) are compared to assess the quality of the compression. Machine learning methods are tested to choose the optimal wavelet with the best qualities. Ultimately, this research will create a program to automatically give the user the best wavelet(s) to compress a particular acoustic data set. [This research is funded by the University of New Orleans Office of Research.]

11:00

2aSP9. Grid-agnostic personalized head-related transfer function modeling with neural fields. You Zhang (Elec. and Comput. Eng., Univ. of Rochester, 500 Wilson Blvd, Rochester, NY 14620, you.zhang@rochester.edu), Yuxiang Wang, Mark Bocko, and Zhiyao Duan (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

The spatial filtering effect brought on by sound propagation from the sound source to the outer ear is referred to as the head-related transfer function (HRTF). The personalization of HRTF is essential to enhance the personalized immersive audio experience in virtual and augmented reality. Our work aims to employ deep learning to predict the customized HRTF from anthropometric measurements. However, existing measured HRTF databases each employ a different geographic sampling, making it difficult to combine these databases into training data-hungry deep learning methods while each of them only contains dozens of subjects. Following our previous work, we use a neural field, a neural network that maps the spherical coordinates to the magnitude spectrum to represent each subject's set of HRTFs. We constructed a generative model to learn the latent space across subjects using such a consistent representation of HRTF across datasets. In this work, by learning the mapping of the anthropometric measurements to the

latent space and then reconstructing the HRTF, we further investigate the neural field representation to carry out HRTF personalization. Thanks to the grid-agnostic nature of our method, we are able to train on combined datasets and even validate the performance on grids unseen during training.

11:15

2aSP10. Universal global compact representation of head-related transfer functions with hemi-spherical harmonics and conformal mapping. Yuxiang Wang (Elec. and Comput. Eng., Univ. of Rochester, 250 e Squire Dr., Apt. 7, Rochester, NY 14623, ywang310@ur.rochester.edu) and Mark Bocko (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

The head-related transfer function(HRTF) is an essential part of spatial auditory display systems. In recent years, numerous HRTF databases are established to include human subjects' measurements, which enabled data-driven research projects such as HRTF prediction and personalization. However, in most cases, each HRTF database has its own unique measuring standard and source direction grid set. It's difficult to merge and efficiently represent the global HRTF information across multiple HRTF databases. Current presentation methods, such as principle component analysis (PCA) and spherical harmonics transform (SHT), are constrained by the source grid layout and regularization errors, which lays burdens for common feature extraction from different sets of HRTF measurements. In this work, we propose a novel approach for the global compact representation of HRTFs across different measuring standards, using hemispherical harmonics (HSH) and conformal mapping. The method takes into account the spatial domain covered by the typical HRTF measurements and proves to be less contained by the source direction grids. Both numerical and auditory model experiments are performed, to examine the representation error and consistency when including multiple HRTF databases with different measuring standards and grid layouts. In continuing work, we are using this method as a pre-processing approach for HRTF personalization utilizing multiple HRTF databases.

11:30

2aSP11. Binaural externalization processing method for object-based audio rendering. Christopher R. Landschoot (1943 W Warner Ave., Apt. 2, Chicago, IL 60613, crlandschoot@gmail.com) and Jean-Marc Jot (Virtuel Works LLC, Aptos, CA)

In both entertainment and professional applications, conventionally produced stereo or multi-channel audio content is frequently delivered over headphones or earbuds. Use cases involving object-based binaural audio rendering include recently developed immersive multi-channel audio distribution formats, along with the accelerating deployment of virtual or augmented reality applications and head-mounted displays. The appreciation of these listening experiences by end users may be compromised by an unnatural perception of the localization of frontal audio objects: commonly heard near or inside the listener's head even when their specified position is distant. This artifact may persist despite the provision of perceptual cues that have been known to partially mitigate it, including artificial acoustic reflections or reverberation, head-tracking, individualized HRTF processing, or reinforcing visual information. In this paper, we review previously reported methods for binaural audio externalization processing, and generalize a recently proposed approach to address object-based audio rendering.

11:45

2aSP12. Sound and voice information transfer enhancement of OLED TV in OTT environment. Hyungwoo Park (ICT, Dong-Seoul Univ., 76, Bockjeong-ro, Sujeong-gu, Seongnam-si, Gyeonggi-do 13117, South Korea, michael.park@du.ac.kr) and Sungtae Lee (LG Display, Gyunggi-do, South Korea)

People get a lot of information around them through visual, auditory, and various senses. In particular, audiovisual data is a very important information transfer factor and has developed various technologies and methods for a long time . In previous studies, we have found that a better effect is achieved when the focus of the screen matches the focus of the sound. To that end, it is now possible to create a sound by vibrating a modern display panel, such as an OLED (Organic Light-Emitting Diode). In our previous research, we have studied the research of creating a stereo sound by directly vibrating a single plane, the reason why the sound of this method is better than the ordinary audio system, the position of the exciter optimized for sound quality, the shape of the exciter and the enclosure of speakers. In this study, a research was conducted on the understanding of the audience's voice by the sound composition in an over the top (OTT) environment that supplies a lot of content recently. In the OTT environment, we try to provide picture quality and sound similar to that of a theater. However, the traditional audio method rather lowers the ability to transmit information of voice. The sound system using the proposed sound radiation system overcomes this problem and creates a clear voice, thereby solving the viewer's discomfort with the sound.

Session 2pAA

Architectural Acoustics, Noise, Structural Acoustics and Vibration, and ASA Committee on Standards: Classroom Acoustics II (Hybrid Session)

David S. Woolworth, Cochair
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Lily M. Wang, Cochair
Durham School of Architectural Engineering and Construction,
University of Nebraska - Lincoln, Omaha, NE 68182

Contributed Papers

1:35

2pAA1. Teacher's vocal health: Voice application systems and other work-related factors. Eric J. Hunter (Com Sci. & Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu), Lady Catherine Cantor Cutiva, Russell Banks (Commun. Sci. & Disord., Michigan State Univ., East Lansing, MI), Pamela Hallam (Educational Leadership and Foundations, Brigham Young Univ., Provo, UT), and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

School teachers' voice problems are widespread and influence the learning process. Voice amplification systems in classrooms are a common response and a preventative strategy. The primary purpose of this study was to (1) determine the relationship between teacher vocal fatigue and use of classroom amplification; and (2) identify factors influencing amplification usage. Teachers throughout the US completed questionnaires regarding vocal health, classroom conditions, lifestyle habits, and access to and use of vocal amplification systems. Responses indicated that biological female teachers who used voice amplification or taught grades kindergarten to middle school were more likely to report higher levels of vocal fatigue. Further, teachers with access to amplification systems were more likely to use them if they were teachers in lower grade levels, who smoked occasionally, drank alcohol frequently, and taught in larger capacity classrooms. In conclusion, it maybe that those using the amplifications are using them because of pre-existing voice issues. Finally, the work-related factors associated with the use of amplification systems (e.g., grade level, classroom capacity) may be indicators of adjustments to reduce the occurrence of voice problems among teachers. Ongoing studies reviewing the available classroom amplification equipment and use from a school administrators perspective will be discussed.

1:50

2pAA2. Survey of the classroom acoustics in Utah elementary and secondary schools and voice amplification systems. Megan R. Robertson (Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, robertsonrachel10@gmail.com) and Brian E. Anderson (Phys. & Astron., Brigham Young Univ., Provo, UT)

Poor acoustics in a classroom can lead to vocal strain and heath challenges for teachers using these rooms over many years. In some cases, voice amplification can be used to help alleviate vocal strain. The acoustics of various types of elementary and secondary school (K–12) classrooms in Utah were assessed with speech parameters calculated from impulse response measurements for teacher to student interactions (using a dodecahedron loudspeaker and microphone) as well as oral binaural impulse responses for the teacher's perception of the room (using a KEMAR mannequin with a mouth simulator). The use of and impact on speech intelligibility of voice amplification was also investigated. This study is part of an overall effort to better assess the degree to which rooms with poor acoustics may impact vocal strain.

Invited Papers

2:05

2pAA3. How do autistic people experience sound? William J. Davies (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., University Rd., Salford, Gtr Manchester M5 4WT, United Kingdom, w.davies@salford.ac.uk)

Autism is a lifelong neurodevelopmental condition diagnosed by differences in social interaction, communication, and imagination. Most autistic people also experience atypical sensory processing (e.g., a heightened sensitivity to sound or texture). The literature includes evidence of autistic hearing differences including hyperacusis, enhanced pitch perception, difficulties with speech in noise, and auditory attention. Most of these can make the classroom a challenging environment for an autistic child. However, despite the rapid growth of autism research, the autistic hearing experience remains poorly understood. Reasons for this may include the lack of involvement of autistic people in research, the medicalised deficit model of most research and a dominant intervention model which seeks to suppress behavioural responses in children. Anecdotal evidence from autistic people suggests a more nuanced and complete picture of autistic hearing will include strengths as well as weaknesses, variation with context and between individuals, and a need for more

accessible design. This paper will synthesise the main features from peer-reviewed and anecdotal evidence on autistic experience of sound, with particular consideration to classroom acoustics.

2:25

2pAA4. Indoor environmental quality (IEQ) factors in K–12 learning spaces: The impact of acoustic performance on students with autism spectrum disorder. Michael Lekan-Kehinde (Univ. of Minnesota, 1985 Buford Ave., St. Paul, MN 55108, lekan012@umn.edu), Abimbola Asojo (Univ. of Minnesota, St. Paul, MN), and B Sanborn (DLR Group, Chicago, IL)

Acoustics is one of the major factors that impact language acquisition, concentration, information retention, and general comfort within the environment needed for preparing children for adult life. Students learn by communication, making speaking and listening crucial. Autism Spectrum Disorder (ASD) is a neurodevelopmental condition that could be complex, multifactorial, and pervasive and is widely described under the umbrella term of Neurodiversity. Neurodiversity describes anyone with a different brain process. As the understanding of cognitive and neurosciences increases, the number of people identified as neurodiverse is nearly 30% of the global population. This research examined the available guidelines and standards for learning space with good acoustic quality, conducted expert interviews and a case study of a space designed to meet this guideline, and finally, make performance threshold and material specification recommendations for improvement to ensure better inclusion of students with ASD. The study involved a mixed methods approach using literature review, interviews, case study, and simulation of a school in the south-central part of the United States. The results describe the current solution used to ensure acoustic suitability for students with ASD and the limitations student with ASD encounter in spaces. This research makes design and material recommendations for improving learning spaces and concludes with an increased need for an integrated understanding of acoustics by all players in the design field.

2:45–3:00 Break

3:00

2pAA5. How can anyone learn or teach? Experiences of autistic people with sound in schools and universities. Carmen Rosas-Pérez (Heriot-Watt Univ., School of Energy, Geoscience, Infrastructure and Society, Edinburgh, United Kingdom, mr2008@hw.ac.uk), Laurent Galbrun, Mary E. Stewart (Heriot-Watt Univ., Edinburgh, United Kingdom), and Sarah R. Payne (Univ. of Surrey, Surrey, United Kingdom)

An inadequate sound environment in schools and universities can lead to poorer learning, higher levels of stress and worse overall well-being, ultimately affecting people's careers and opportunities. These short- and long-term impacts can be more accentuated in autistic students, teachers, and lecturers, as autistic and other neurodivergent people tend to have significant perceptual and sensory processing differences. This paper presents the data from our study on the experience of autistic people in daily life acoustic environments related to this topic. The study explored the experiences of 12 autistic adults through reflective thematic analysis of semi-structured interviews, providing much-needed first-hand perspectives on main challenges, impact, and common coping strategies. The results highlight the crucial importance that good acoustics has for the participants, some of whom are or have been teachers and lecturers. They also illustrate how the lack of better acoustic practices and understanding of their experiences, together with some trends like open-plan schools and offices, have had a disabling impact throughout their life since childhood.

3:20

2pAA6. Understanding the impact of acoustics on the learning ability among children with autism in the classroom. Catrin Tufvesson (National Agency for Special Needs Educations and Schools, Norra Neptunigatan 7, Malmö 21118, Sweden, catrin.tufvesson@icloud.com)

A pronounced ambition concerning the educational policy in Sweden is to organize an all-inclusive school, meaning that schools should be able to meet the needs of all children, irrespective of their capacities and conditions. However, there were no specific guidelines for the school environment as a learning environment. In my thesis The Human Environment Interaction-model is implemented to identify environmental factors that affect children with ADHD, autism, and Down's syndrome and their ability to concentrate in their learning environment at school. The issue was not to dismiss any of the additional resources these children are in need of; instead it was to suggest how to arrange learning environments in the most supportive way possible. This presentation will focus on aspects of acoustics which are—and which are not—beneficial for children with autism. The results from the thesis have been an underlying basis for further work at the National Agency for Special Needs Educations and Schools in Sweden, which will also be presented. The developed work at the Agency resulted in a tool with guidelines to make the learning environment more accessible for all children in existing classrooms, but it can also be implemented in the building process.

3:40

2pAA7. An experimental study of acoustical design, physiological measurements, and repetitive behaviors in children with autism. Shireen Kanakri (Construction Management and Interior Design, Ball State Univ., 12701 Broncos Dr., Fishers, IN 46037-7804, smkanakri@bsu.edu)

Researchers have known that sound is an aspect of the environment that can have an effect on children with Autism Spectrum Disorder (ASD) due to hypo or hypersensitivities that are commonly present in this population. What is less known is how sound is specifically correlated to behaviors and physiological phenomenon. In order to build a predictive measure of sound decibel level on behavior, children with ASD were observed in a controlled environment while exposed to varying acoustic levels. Behaviors, heart rate, ECG, and respiration were recorded during experimental sessions and results revealed that the children typically stayed on task consistently throughout the sound stimulation; however, significant ECG, HR, and respiration rate variances occurred at the different sound levels in all behaviors. The results found from this study do point to a predictive relationship when it comes to the influence of sound and physiological readings on behavior in ASD children.

4:00

2pAA8. Measurements of classroom acoustic conditions in which young neurodiverse persons learn in São Paulo and Nebraska.
Fernanda Caldas (Univ. of Campinas, 400 Albert Einstein, Av, Campinas, São Paulo 13083-852, Brazil, fernandacaldas0@gmail.com), Fernando Teodoro De Cillo (Univ. of Campinas, Campinas, São Paulo, Brazil), Samuel H. Underwood, Sanjay Kumar (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), Eloisa Valler Celeri, Bruno Masiero (Univ. of Campinas, Campinas, São Paulo, Brazil), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

Autism is a set of complex neurobehavioral conditions that involve, among other characteristics, hypo/hypersensitive responses to any sensory input, usually referred to as sensory processing disorder. The most common sensory difference associated with autistic individuals is an over or under response to visuals, touch, smells, tastes, balance, and sounds when compared to neurotypical individuals. Among these, auditory sensitivity has a significant influence. Research studies to date have revealed that autistic individuals are highly susceptible to surrounding sounds, and show that certain noises, such as loud sounds, classroom bells, sirens, the radio or the TV, and traffic noise, may induce uncomfortable and distracting behavior. In this study, we investigated the acoustic environment of classrooms/facilities designed for a more inclusive education. A number of room acoustic descriptors, such as reverberation time (RT), speech clarity (C50), speech transmission index (STI), sound levels, and auditory strength (S/N ratio) were measured in this regard. The experimental measurements were performed in several rooms and facilities focusing on autistic individuals in Campinas, São Paulo, in Brazil and in Omaha, Nebraska, in the United States.

4:20

2pAA9. Soundfield amplification for primary school students on the autism spectrum: Research findings and classroom applications. Wayne J. Wilson (Discipline of Audiol., School of Health and Rehabilitation Sci., The Univ. of Queensland, Queensland 4072, Australia, w.wilson@uq.edu.au), Keely Harper-Hill (Queensland Univ. of Technol., Queensland, Australia), Cerys Downing (Audiol., The Univ. of Queensland, Queensland, Australia), Rebecca Armstrong (Speech Pathol., The Univ. of Queensland, Queensland, Australia), Kelsey Perrykkad (Monash Univ., Monash University, Victoria, Australia), Mary Rafter (Psych., The Univ. of Queensland, Queensland, Australia), and Jill Ashburner (Autism Queensland, Autism Queensland, Queensland, Australia)

Sound field amplification (SFA) has recently attracted interest as an inclusive classroom adjustment for children on the autism spectrum. When functioning optimally, SFA improves the signal-to-noise ratio in the classroom by maintaining the teacher’s voice at a higher level than the classroom noise. In this paper, we will report on the results of a randomised controlled trial of SFA over two semesters for 13 children (9 males, aged 7.6 to 8.4 years) on the spectrum and 17 children (7 males, aged 7.6 to 9.3 years) not on the spectrum in their third year of schooling in 10 elementary schools in Australia. SFA was found to assist children on the spectrum to improve their skills in an area of phonological processing known to be important to reading acquisition, but not their skills in attention, memory or educational achievement. In conclusion, SFA continues to show sufficient potential for supporting children on the spectrum in the classroom to warrant its trialing on a case-by-case basis. Realistic expectations as to the short-term benefits of SFA are needed.

Session 2pAB

Animal Bioacoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Session in Honor of James A. Simmons II (Hybrid Session)

Laura Kloepper, Cochair
Department of Biological Sciences, University of New Hampshire, 230 Spaulding Hall, Durham, NH 03824

Alyssa W. Accomando, Cochair
Biologic and Bioacoustic Research, National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106

Invited Papers

1:00

2pAB1. Target ranging by echolocation in bats: Performance and mechanisms. Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames 200B, Baltimore, MD 21218, cynthia.moss@jhu.edu), Ninad B. Kothari (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Melville Wohlgemuth (Neurosci., Univ. of Arizona, Baltimore, MD), Jinhong Luo (Biology, Central Normal Univ., Baltimore, MD), Aditya Krishna, and Xiaoyan Yin (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

James Simmons's early research on sonar ranging in echolocating bats generated two groundbreaking discoveries: (1) Bats compute object distance from the time delay between sonar calls and echoes, and they discriminate echo-delay differences in the microsecond range [Simmons, https://psycnet.apa.org/doi/10.1121/1.1913559 (1973)] and (2) A population of auditory neurons show facilitated and echo delay-tuned responses, a posited neural substrate of sonar ranging [Feng *et al*., https://10.1126/science.705350 (1978)]. These findings spawned decades of biosonar research around the world, and this talk will summarize three new findings on the mechanisms of sonar ranging in the big brown bat, *Eptesicus fuscus*. (1) Echo-delay tuned neurons in the midbrain of the free-flying bat show 3D spatial tuning to echoes from physical objects, and sonar-guided attention evokes sharper delay-dependent response areas and shifts to shorter echo delays. (2) Local field potential recordings from auditory midbrain neurons in the passively listening bat encode the time interval between call-echo pairs in the microsecond range, with accuracy dependent on signal duration and bandwidth. (3) A population of hippocampal CA1 neurons encodes the distance of sonar objects in bats as they track moving targets, and responses of hippocampal neurons depend on the production of sonar calls that yield echo returns.

1:20

2pAB2. The quest for the biosonar image of the world. Rolf Müller (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Sci. Cir, (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu)

The large differences in the wavelength of visible light and the ultrasonic pulses of bat biosonar raise the question whether the sensory world of bats is the same as that of humans and other visual animals, completely different, or something in-between. Pioneering research on this question has focused on the range dimension of biosonar perception exploiting the straightforward relationship between echo delay and target range. Furthermore, since bat ears are typically no more than a single order of magnitude longer than the respective ultrasonic wavelengths, the beam patterns of the animals are fairly wide, which can be expected to result in a much better resolution for range than along the cross-range dimension. Bats that use their biosonar to navigate in densely vegetated environments receive echoes containing contributions from many unresolved scatterers, i.e., "clutter echoes." Extracting useful information from clutter has been studied using real-world data collected with biomimetic sonar systems and analyzed with deep-learning methods. The results demonstrate that useful information can be extracted from clutter echoes without the formation of an image, i.e., without a systematic spatial representation of resolved objects. Hence, many interesting aspects of how bats experience their sensory worlds may still be left to be discovered.

1:40

2pAB3. Auditory processing for action selection in bats. Angeles Salles (Biological Sci., Univ. of Illinois Chicago, 3400 N. Charles St., Ames Hall, Ste. 232, Baltimore, MD 21218, angiesalles@gmail.com), Emely Loscalzo (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Jessica Montoya (Biological Sci., Univ. of Illinois Chicago, Chicago, IL), Kevin Boergens (Phys., Univ. of Illinois Chicago, Chicago, IL), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Bats are auditory specialists, processing acoustic signals to guide their behaviors, including prey tracking, navigation and communication. In this talk I will provide an overview of my work related to how bats analyze and process signals for action-selection; specifically prey tracking and background clutter segregation—work inspired by research from Jim Simmons's lab. I will also give an overview of the line of research of my current lab, neural mechanisms for auditory processing of communication signals. There is strong

evidence that context plays a role in the processing of acoustic signals. Yet, the mechanisms that govern this process are still not fully understood. *Eptesicus fuscus* bats emit a wide array of communication calls, including food claiming calls, aggressive calls and appeasement calls. We developed a novel competitive foraging task to explore the role of behavioral context in auditory responses to social calls. With this approach, we recorded neural population responses from the IC of freely interacting bats. Analysis of our neural recordings from the IC show stronger population responses to individual calls during behaviorally aggressive events. These results indicate that behavioral context plays a role in the modulation of neuronal population responses to social vocalizations in the bat IC.

2:00

2pAB4. Out of the bat lab and into the field: Sensorimotor strategies of bats at high-speed flight. Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu), Ian Bentley (Saint Mary's College, Notre Dame, IN), Christian Harding (Univ. of Oxford, Oxford, United Kingdom), Caroline Brighton (Univ. of Oxford, Wytham, United Kingdom), Mohammad Izadi, Robert Stevenson (Univ. of Notre Dame, Notre Dame, IN), and Graham Taylor (Univ. of Oxford, Wytham, United Kingdom)

The experiments performed by James Simmons helped pave the way for our understanding of bat biosonar processing and "what it's like to be a bat." By conducting experiments with trained bats, he discovered key information on how bats use temporal and spectral cues in their echoes to perceive their environment. After spending my postdoc with Dr. Simmons in his lab, he encouraged me to venture to the field to explore the sonar behavior of bats in their natural environment. This talk will summarize my recent work exploring sensorimotor behavior of free-tailed bats when returning to their cave roost at speeds exceeding 60 km/h. By combining acoustic recordings with multiple-camera recordings, we reconstructed the 3D path of bats in flight both during and in-between pulse emission. From these data we examined how acoustic behavior changed depending on altitude and velocity, and whether changes in flight trajectory or kinematics coincided with the reception of echoes. Our results demonstrate that when echolocating at fast flight speeds, bats use a predictable, open-loop sensing strategy to navigate their environment. This may be advantageous for bats flying at fast speeds and experiencing high g forces, as the physiological consequences of closed-loop sensing may be too demanding.

2:20

2pAB5. Noise-induced vocal flexibility in Hipposiderid and Rhinolophid bats. Jinhong Luo (School of Life Sci., Central China Normal Univ., 152 Luoyu Rd., Wuhan 430079, China, jluo@ccnu.edu.cn), Tingting Wei, Nina Ma (School of Life Sci., Central China Normal Univ., Wuhan, China), and Aiqing Lin (Jilin Provincial Key Lab. of Animal Resource Conservation and Utilization, Northeast Normal Univ., Changchun, China)

Although James Simmons's pioneering work on bats' target ranging capability by echolocation has been widely acknowledged, his contributions to other fields of bat echolocation have often been underappreciated. For example, James Simmons was probably the first to observe that echolocating bats can employ remarkable vocal flexibilities to counter masking noise effects on target ranging [Simmons *et al.*, https://doi.org/10.1121/1.1914154 (1974)], a research topic that has only gained popularity in the bioacoustics community in recent two decades or so. In this talk, we present the latest data on noise-induced vocal flexibility of five species of bats, including *Hipposideros pratti* and four Rhinolophids. All bats emit echolocation calls consisting of a constant frequency component and a frequency modulated component. We first show that both Rhinolophid and hipposiderid bats possess a highly developed vocal flexibility to counter noise interference. Then, with *H. pratti* we demonstrate that the expression of vocal flexibility by the same individual was affected by several factors, such as the behavioral context or task, and history of training, which were believed to mediate the noise-induced vocal flexibility of humans only. These results emphasize the value of the comparative approach in understanding the vocal production control of mammals.

2:40

2pAB6. Paying attention to the man behind the curtain: Jim Simmons' contributions to bat echolocation. Michaela Warnecke (Reality Labs Res. at Meta, 16 Partridge Ave., Somerville, MA 02145, elabelabohne@gmail.com)

Echolocating big brown bats (*Eptesicus fuscus*) perceive their surroundings by broadcasting frequency-modulated (FM) ultrasonic pulses and processing returning echoes. These bats commonly navigate acoustically-cluttered environments, in which each broadcast is followed by multiple echoes at varying time delays and echo amplitudes. The bat must decode these echo cascades in order to create a coherent percept of its surroundings and adapt its flight and echolocation behavior in real time. Jim Simmons' physiological and behavioral work—which continues to fuel research questions to this day—has significantly helped our understanding of how this decoding might be possible. In this talk, I will revisit some of Jim Simmons' seminal work that contributed to our understanding of echolocation processing, and describe its influence on new generations of researchers who continue to build on Jim's important work.

3:00–3:15 Break

3:15

2pAB7. The soundscape of dense bat swarms: Differences based on location within the group. Keegan Eveland (Dept. of Biological Sci., Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 40 Gables Way, Durham, NH 03824, Keegan.Eveland@unh.edu), Robert Stevenson (Univ. of Notre Dame, Notre Dame, IN), Paul Domski (New Mexico Falconry Assoc., Tijeras, NM), and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

The laboratory work of Jim Simmons and colleagues has provided key information on how bats use harmonic structure to identify targets from background clutter. Although advantageous for detecting targets straight ahead, the question remains as to how bats modify echolocation, if at all, when identification of objects in the periphery may be needed, such as when flying in a dense group. Based on the work of Dr. Simmons and colleagues, we predicted that free-tailed bats in central positions of swarms may use different echolocation signals than bats on the periphery. We investigated this by quantifying the soundscape based on location within the group. To overcome the challenge of recording inside a dense swarm, we developed a microphone/camera unit carried by a trained Harris hawk that flew through the bat swarm. Frequency spectra were extracted and analyzed based on the frequency-dependent amplitude compared to position in the swarm. We found a significant difference in the soundscape between the 'Edge' and 'Middle' of the swarm for frequencies above 40 kHz, suggesting bats in central positions of the swarm produce echolocation signals with more energy in higher frequencies, which may aid in distinguishing their calls from conspecifics.

3:30

2pAB8. Neural responses of the auditory midbrain to naturalistic echolocation sequences in *Hipposideros armiger*. Zhongdan Cui (School of Life Sci., Central China Normal Univ., 152 Luoyu Rd., Hongshan District, Wuhan City, Hubei Province, China 430079, China, cui@mails.ccnu.edu.cn) and Jinhong Luo (School of Life Sci., Central China Normal Univ., Wuhan, China)

The functional properties of the auditory midbrain, inferior colliculus (IC), plays a critical role in James Simmons's SCAT model of target ranging by echolocation. This SCAT model is based solely on the big brown bat producing frequency-modulated (FM) sonar calls. By contrast, there has been very limited knowledge on how the auditory midbrain may encode target ranging information in bats that produce constant-frequency (CF) sonar calls. Here, we conducted single-unit recordings from the auditory midbrain of passively listening *Hipposideros armiger* using naturalistic echolocation sequences. We created echolocation sequences from recordings taken from individual bats swung on a moving pendulum that evokes Doppler shift compensation (DSC) of their sonar vocalizations. Our preliminary data revealed a population of IC neurons selectively responded to DSC information in the sequences. To our surprise, we did not find neurons that encode echo delays. Considering published data on FM bats, these results suggest that the functional role of auditory midbrain for echolocation may be distinct between FM bats and CF-FM bats.

3:45

2pAB9. Discrimination of simulated two-highlight echoes including phase manipulations by bottlenose dolphins (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), Alyssa W. Accomando (National Marine Mammal Foundation, San Diego, CA), Katie A. Christman (Psych., UC San Diego, San Diego, CA), Katelin Lally (National Marine Mammal Foundation, San Diego, CA), Austin O'Kelley (Psych., UC San Diego, La Jolla, CA), Dorian Houser (National Marine Mammal Foundation, San Diego, CA), and James Finneran (Code 56710, NIWC Pacific, San Diego, CA)

Dolphins potentially use spectral cues to discriminate inter-highlight interval (IHI) for passively presented, simulated two-highlight echo stimuli. To investigate this potential, dolphins were trained to listen to repetitive two-highlight "background" echoes and respond upon a change to "target" echoes with increased IHI. In the first experimental task, all highlights had the same phase, and the target differed from the background only in terms of IHI. In the second task, highlights within both background and target echoes had random phase angles. This resulted in a spectral interference pattern for the two-highlight echo with the same notch spacing as the first condition, but different absolute positions of the notches along the frequency axis. Discrimination thresholds for the constant-phase task were lower than those for the random-phase task at IHIs of 250 μs and lower. Thresholds for both tasks increased with increasing background IHI and were comparable at 375 and 500 μs. Within the dolphin auditory temporal window of $\sim$250 μs, IHI discrimination is therefore likely related to—but not strictly based on—the spacing of notches in the complex echo spectrum. Above 250 μs, IHI is likely determined through direct encoding of highlight timing in the auditory periphery. [Work funded by ONR.]

4:00

2pAB10. Effects of absolute range and echo phase on range discrimination in bottlenose dolphins (*Tursiops truncatus*). Katie A. Christman (Psych., UC San Diego, 2240 Shelter Island Dr., San Diego, CA 92106, katie.christman@nmmf.org), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), Tim Gentner (Psych., UC San Diego, La Jolla, CA), Dorian Houser, and Jason Mulsow (National Marine Mammal Foundation, San Diego, CA)

Echo-delay (range) resolution is a critical feature of animal biosonar. In 1973, Jim Simmons was the first scientist to test bat range discrimination thresholds at a variety of absolute ranges. He used a two alternative-force-choice paradigm with two phantom echo generators (PEGs) and compared the results to the bats' performance with physical targets. In the present study, methods similar to Simmons' 1973 study were used to test two dolphins' range discrimination thresholds using a two channel PEG system. Range discrimination thresholds were tested at seven different absolute ranges between 1.75 and 20 m. Ranges were simulated by manipulating echo-delay while relative echo level was held constant for all parameters. To examine potential effects of echo phase, measurements were repeated at a 7 m range while randomizing echo phase. For both subjects, range discrimination thresholds increased exponentially with absolute range. Randomizing echo phase did not significantly change thresholds, suggesting that dolphins use the envelope of the echo waveform to determine range versus the fine structure.

Session 2pAOa

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Megan Ballard, Chair
University of Texas at Austin, 10000 Burnet Road, Austin, TX 78665

Chair's Introduction—1:30

Invited Paper

1:35

2pAOa1. Passive acoustic exploration of the ocean. David R. Barclay (Dept of Oceanogr., LSC Bldg., Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H 4R2, Canada, dbarcl@gmail.com)

Encoded in the spectral density, spatial variability, and directionality (spatial coherence) of the ambient sound field is information on the generation mechanisms of sound and the properties of the ocean propagation environment and its boundaries. Through field and observatory measurements, and analytical and computational models of the underwater sound field, a research program has been pursued that asks, "What can we learn about the ocean by listening?" Large acoustic data sets have been exploited to develop an estimate of the effective source level per unit area of surface generated noise. In complement, a methodology for precisely partitioning the sound field into ship generated and wind generated components by exploiting the vertical noise directionality has been demonstrated. Models of the spatial properties of wind driven and ship generated sound have further been used to estimate the geoacoustic properties of the seabed, the depth of mix layer, depth-averaged pH, and localize a source in bearing and azimuth using only a pair of vertically oriented omnidirectional hydrophones. An autonomous passive acoustic profiler, The Deep Acoustic Lander (DAL), recently made measurements of the ambient sound field from the surface to the bottom of the Challenger Deep, Mariana Trench, precisely determining the mixing of a locally- and distantly generated contributions to the sound field. Meanwhile, DAL measurements at the Endeavour hydrothermal vent field have revealed components of the sound field generated by vent activity.

Session 2pAOb

Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Memorial Session for Jeffrey A. Nystuen II (Hybrid Session)

Jennifer Miksis-Olds, Cochair
University of New Hampshire, 24 Colovos Rd., Durham, NH 03824

Kay L. Gemba, Cochair
Physics Dept., 833 Dyer Road, Bldg 232, NPS, Monterey, CA 93943

Jie Yang, Cochair
Applied Physics Lab, University of Washington, 1015 NE 40th St., Seattle, WA 98105

Invited Papers

2:40

2pAOb1. Open ocean ambient noise data in the frequency band of 100 Hz–50 kHz from the Pacific Ocean: A legacy of Jeffrey A. Nystuen. 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), and Eric I. Thorsos (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Ocean ambient noise, spanning from a few hertz to tens of kilohertz, is often the limiting factor for sonar performance in target detection, location, and identification. In this frequency band, wind generated surface breaking waves produce bubbles near the surface that are the dominant ambient noise source. In this work, results from a long-term collaboration between NOAA/ NASA and ambient noise study pioneer, Jeffrey A. Nystuen, are presented. Specifically, two-decades of ambient noise data from six deep ocean moorings with companion surface meteorological measurements are used to validate ambient noise models. Excluding data during rainy periods, the ambient noise level is investigated under different wind speed ranges. For wind speeds exceeding 15 m/s, the ambient noise level displays a sharp drop-off and creates a "cross-over" as the spectral level at higher wind speeds and frequencies becomes lower than that at lower wind speeds. 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. This mismatch, currently under investigation, is likely due to attenuation when ambient sound propagates through the deeper and denser bubble layer under high sea conditions. [Work supported by NOAA, NASA, and ONR.]

3:00

2pAOb2. Variability analysis of low-frequency and very low-frequency ambient ocean noise. David L. Bradley (Dept. of Defense, Mark Ctr., Alexandria, VA 22350, david.bradley@unh.edu) and Anthony Eller (Appl. Ocean Sci., Springfield, VA)

Much of the information presented was made possible by the presence over the past several years of a collection of deep water hydroacoustic arrays with long-term, high data rate recorders of ocean ambient noise, deployed and maintained under the auspices of the United Nations Comprehensive Nuclear Test Ban Treaty Office (CTBTO). Not only does the long time span of the noise record allow high frequency resolution of annual and multi-year variability, but the high data rate supports as well the analysis of high temporal resolution minute-by-minute variability. The present analysis is directed at acoustic frequencies where the noise level and its variability are governed primarily by natural events : planetary—such as the influence of tides, oceanographic—such as the influence of El Nino, and meteorological—such as the influence of storms and routine weather. The analysis is focused on the fluctuations of a series of narrow band tonals at acoustic frequencies between 0.1 and 10 Hz. Results of the analysis display the time scales of the various processes that drive the fluctuations.

3:20

2pAOb3. Jeffrey Nystuen: An innovator in the field of ocean ambient sound. Martin Siderius (Portland State Univ., 1600 SW 4th Ave., Ste. 260, Portland, OR 97201, siderius@pdx.edu)

It is a testament to Jeff Nystuen's pioneering research that many of us now think about ocean ambient sound rather than ocean ambient noise. The term noise makes us think of this as a nuisance factor that needs to be overcome by a useful signal. However, Jeff changed this paradigm and showed how valuable environmental information can be extracted from the ambient sound field. Specifically, he developed the acoustic rain gauge and expanded our understanding of the sound rain generates in the ocean. He spent many years bringing this idea from a concept to a mature device that could be practically used in the ocean to estimate rainfall and measure other important ambient sound events. Following Jeff's early work, many other techniques that use ambient sound for environmental characterization have been developed. In this talk, we will pay tribute to Jeff's groundbreaking research. We will also explore how this work has led to improved models for ambient sound as well as applications to environmental sensing.

3:40–3:55 Break

Contributed Papers

3:55

2pAOb4. Mid-frequency acoustic localization of breaking waves. Ryan Saenger (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., San Diego, CA 92093, rsaenger@ucsd.edu), Luc Lenain, William Kuperman (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

During an experiment in deep water off the coast of Southern California, winds between 20 and 25 knots resulted in large breaking waves. A mid-frequency planar hydrophone array recorded underwater ambient noise while an airplane equipped with a high-resolution video camera recorded images of the sea surface above the array. Beams of ambient noise between 5 and 6 kHz were projected onto the sea surface and aligned in space and time with the aerial images. The array's resolution of the surface is coarse (20 by 50 m) due to its modest 1 meter horizontal aperture and relatively deep 130 meter deployment depth. Nonetheless, concentrated regions of high intensity in the acoustic surface projection match visible breaking events in the aerial images. [Work supported by the Office of Naval Research.]

4:10

2pAOb5. Ambient sound levels, directionality, and transport velocity around a seamount measured by a deep-drifting acoustic vector sensor. Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0206, La Jolla, CA 92093, athode@ucsd.edu), Dieter Bevans (Keyport, Naval Undersea Warfare Ctr., Keyport, WA), Alison B. Laferriere (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Eric Berkenpas, Mike Shepard (Second Star Robotics, Silver Spring, MD), and Lauren A. Freeman (NUWC Newport, Newport, RI)

In September 2022 an opto-acoustic drifter was deployed several times in the vicinity of the Kelvin seamount (38°49'N, 64°2'W) in the North Atlantic Ocean, which rises to a depth of 1500 m from a 5 km deep abyssal plain. The "SQUALL-E" drifter incorporates stereo video cameras, vertical (1.75 m aperture) and tetrahedral hydrophone arrays, and a 2-D Geospectrum M35 acoustic vector sensor. One overnight deployment descended to 250 m depth, and a another to 400 m depth, which resulted in highly different drift rates. The M35 sensor demonstrated the ability to estimate the directionality of ambient sounds between 1 and nearly 30 kHz. Sperm whales, delphinids, various vessels, and other discrete and distributed sources were detected. For both deployments we present distributions of ambient sound levels, azimuthal and elevation angles, and transport velocity. We discuss how these distributions evolved throughout the deployments. [Work sponsored by ONR TFO program.]

4:25

2pAOb6. Source separation and arrival angle estimation using a single acoustic vector sensor. Alison B. Laferriere (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0206, La Jolla, CA 92093-0206, alaferriere@ucsd.edu) and Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Extracting individual time signatures from two angularly-separated sources overlapping in time and bandwidth typically requires coherent array processing on multiple hydrophones, a process whose performance is degraded by limited aperture and spatial aliasing. Vector sensors, which measure both acoustic pressure and particle velocity, can measure an acoustic field's directionality in a single, compact sensor package. Standard beamforming techniques can be applied to the outputs of a vector sensor to suppress one source's signal, but its efficacy is limited by the angular insensitivity of the resulting cardioid beam pattern. Here we use fundamental physical principles to demonstrate that measurements on a 2-D vector sensor can completely determine the azimuths, amplitudes, and relative phases of two horizontal plane waves of the same frequency, achieving source separation and direction-of-arrival estimates for both. Interference between the waves creates a reactive intensity whose orientation is perpendicular to the bisector of the two wavenumber vectors. This observation leads to closed-form inversion formulas. Data examples of the technique are illustrated using overlapping humpback whale songs off Hawaii, and on weak airgun and bowhead whale signals embedded within intense directional noise from a nearby drilling site in the Beaufort Sea. [Work sponsored by ONR TFO.]

Invited Paper

4:40

2pAOb7. Video tributes to Jeffrey A. Nystuen. Jie Yang (Appl. Phys. Lab, Univ. of Washington, 1015 NE 40th St., Seattle, WA 98105, jieyang@uw.edu), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), and Kay L. Gemba (Naval Postgrad. School, Monterey, CA)

This video presentation is dedicated to the career and legacy of Jeffrey A. Nystuen.

Session 2pBAa

Biomedical Acoustics and Physical Acoustics: Clinical Perspective of Biomedical Acoustics II (Hybrid Session)

Tyrone M. Porter, Cochair
Biomedical Engineering, University of Texas at Austin, The University of Texas at Austin, 107 W Dean Keetom St., United States, TX 78712

Flordeliza S. Villanueva, Cochair
Medicine/Cardiology, University of Pittsburgh, UPMC Presbyterian, Suite A-351, Pittsburgh, PA 15213

Invited Papers

1:00

2pBAa1. Microvascular no reflow in acute MI—Mechanistic studies of sonoreperfusion. John J. Pacella (Medicine, Univ. of Pittsburgh, 3500 Terrace St., Pittsburgh, PA 15213, pacellajj@upmc.edu)

Congestive heart failure (CHF) following acute myocardial infarction (AMI) is on the rise. Microvascular obstruction (MVO) is the single most important contributor to post-AMI CHF; it occurs in up to 80% of patients with AMI and is associated with worse outcomes. To address this unmet need, we have been developing ultrasound targeted microbubble cavitation (UTMC) to achieve "sonoreperfusion" (SRP) for MVO. During SRP, US is applied to intravenously administered MBs as they transit the coronary microcirculation, which causes MBs to oscillate and generate shear stress. We have previously shown that SRP therapy restores perfusion during MVO through direct mechanical "chiseling" effects of MBs on microthrombi. However, MVO is more than just physical obstruction from microemboli; it is characterized by a milieu of inflammation and oxidative stress, a state of imbalance between the formation of damaging reactive oxygen species (ROS) and antioxidant defenses. One such defense is nitric oxide (NO), which has multi-level therapeutic potential for MVO. We have demonstrated that UTMC releases NO from endothelial cells through mechanotransduction, and that NO is a major contributor to the efficacy of SRP. We are now actively pursuing strategies to optimize SRP, not only through mechanical mechanisms but also by addressing oxidative stress.

1:30

2pBAa2. Sonothrombolysis. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Cardiovascular disease and stroke are the two leading causes of death worldwide (doi.org/10.1161/CIRCOUTCOMES.118.005375). Sonothrombolysis, ultrasound which accelerates thrombus breakdown in combination with thrombolytic drugs, has been under extensive investigation in the last two decades. An overview of *in vitro*, preclinical, and clinical sonothrombolysis studies will be provided. In a xenographic porcine cerebral thromboembolism model employing retracted human clots, 220 kHz ultrasound, in conjunction with Definity® increased the probability of early successful reperfusion with rt-PA. Combined treatment with rt-PA, Definity®, and ultrasound *in vivo* increased the rate of reperfusion up to 45 min faster than clots treated with rt-PA or saline alone. Tsivgoulis *et al.* (doi.org/10.1161/STROKEAHA.120.030960) performed a meta-analysis of sonothrombolysis safety and efficacy data in patients with acute ischemic stroke from 7 randomized-controlled clinical trials. Sonothrombolysis was associated with a 2-fold increase in the odds of complete recanalization compared with intravenous lytic alone. The likelihood of symptomatic intracranial hemorrhage was not significantly different between the 2 groups (7.3% versus 3.7% odds ratio). However, in spite of the higher recanalization rates no difference in 3-month clinical outcomes was uncovered, though this conclusion might have been underpowered statistically. Strategies to improve clinical sonothrombolysis outcomes will be discussed.

2:00

2pBAa3. Clinical trials of sonothrombolysis for Acute Coronary Syndromes. Wilson Mathias (School of Medicine, Univ. of Sao Paulo, Av. Dr. Eneas de Carvalho Aguiar, 44 Cerqueira Cesar, Sao Paolo 05403-000, Brazil, wilson.mathias@hc.fm.usp.br)

Diagnostic high mechanical index (MI) ultrasound in conjunction with microbubble infusion in patients with acute ST-elevation myocardial infarction (STEMI) may be a method of achieving early recanalization, (sonothrombolysis). To evaluate the safety, efficacy and prognostic impact of sonothrombolysis in patients with acute coronary syndromes (ACS). We will randomize 540 patients with anterior STEMI in the "High Ultrasound Mechanical Index and MicrobuBBLEs to reduce acute myocardial infarction burden" (HUBBLE-I). Sonothrombolysis (50 min) was performed using multiple intermittent high MI impulses when microbubbles were imaged within the risk area (Therapy Group) or only 3 projections of low MI imaging alone plus microbubbles (Control Group) in 5 patient subgroups: G1-STEMI pre-PCI; G2-STEMI after lytic therapy; G3-Unstable angina and non-STEMI; G4-STEMI post-PCI; G5-STEMI pre-PCI treated in ambulances. Recanalization, ejection fraction, troponin and MB-CK levels were acquired at baseline, pre-PCI and

follow-up. Global longitudinal strain was serially measured. The number of patients randomized to control versus therapy was 47 (23/24), as follows: G1- 12 (5/7); G2- 13 (9/4); G3- 14 (8/6); G4- 8 (4/4) and G5- 0(0/0). This multicenter, Phase III randomized trial in patients with ACS treated by sonothrombolysis will assess safety, efficacy, and prognostic impact of this therapy.

2:30

2pBAa4. Photoacoustic imaging for assessment of abnormal placental function during pregnancy. Carolyn Bayer (Biomedical Eng., Tulane Univ., 530 Lindy Boggs Ctr., Tulane University, New Orleans, LA 70118, carolynb@tulane.edu)

Insufficient placental function is a comorbidity of many pregnancy pathologies, including preeclampsia, and may be a driver of epigenetic changes affecting the future health of mother and child. Existing ultrasound imaging modalities assess placental anatomy, but provide insufficient information about function, limiting clinical care. Our research develops photoacoustic imaging of placental function to characterize progression and treatment of placental-related diseases. In photoacoustic imaging, short laser pulses generate acoustic signals from light-absorbing chromophores within the tissue. Since the optical absorption of many tissue chromophores is wavelength dependent, by varying the wavelength of the laser, estimates of the tissue composition can be obtained through analytical fitting of the acquired photoacoustic signal to the known optical absorption of the tissue chromophores, termed spectral photoacoustic imaging (sPA). sPA imaging was applied to demonstrate that ischemia of the placenta is a defining condition of preeclampsia in a preclinical animal model of the disease, and that therapeutics demonstrate a mechanism-specific reduction in the extent of ischemia in the preeclamptic placenta. Currently, tissue optical attenuation limits the imaging depth of photoacoustic imaging, but strategies are being developed to improve imaging depth and potentially enable clinical translation.

3:00–3:15 Break

Contributed Paper

3:15

2pBAa5. Analysis of microvasculature morphological features for prediction of metastatic lymph nodes in breast cancer patients via contrast-free ultrasound imaging. Soroosh Sabeti (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine & Sci., 200 First St. SW, Rochester, MN 55905, sabeti.soroush@mayo.edu), Giulia Ferroni (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), Nicholas B. Larson (Quantitative Health Sci., Mayo Clinic College of Medicine, Rochester, MN), Robert T. Fazzio (Radiology, Mayo Clinic College of Medicine and Sci., Rochetser, MN), Mostafa Fatemi (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine & Sci., Rochester, MN), and Azra Alizad (Radiology, Mayo Clinic College of Medicine and Sci., Rocheter, MN)

Axillary lymph nodes (ALNs) are generally among the first to be impacted by the metastatic spread of breast tumors. Consequently, evaluating their condition is of great importance as the removal of affected ALNs can potentially curtail the further spread of cancer cells to other parts of the body. Conventional ultrasound imaging suffers from low sensitivity and specificity and does not provide a comprehensive picture of the status of the node. In this work, we utilize an ultrasound microvessel imaging technique to visualize vascular structures within ALNs and, through several image processing steps, characterize the morphology of these structures. Several morphological metrics are used as biomarkers to differentiate between reactive and metastatic lymph nodes of patients with suspicious ALNs. A group of 68 patients is included in this study. Pathology examination results are used as the gold-standard reference for data labeling. A support vector machine (SVM) model is trained, and its performance is evaluated. Statistical analysis of the discriminative potential of individual biomarkers and receiver operating characteristic (ROC) curve analysis of the trained model are presented. Preliminary results suggest that a sensitivity of 0.88, specificity of 0.88, and area under the curve of 0.94 are achievable. Work supported by NIH-R01CA239548.

Invited Papers

3:30

2pBAa6. Hypertensive disorders of pregnancy and echocardiography in cardio-obstetrics. Malamo Countouris (Medicine, Univ. of Pittsburgh, 200 Lothrop St., Pittsburgh, PA 15213, countourisme@upmc.edu)

Hypertensive disorders of pregnancy, such as preeclampsia, are associated with adverse cardiovascular outcomes in the peripartum period including heart failure, peripartum cardiomyopathy, and stroke. Echocardiography provides a noninvasive way to assess cardiac structure and function and changes related to hypertensive disorders of pregnancy. In this talk, we will discuss expected cardiac changes measured by echocardiography among women with hypertensive disorders of pregnancy. We will also discuss other cardiac complications related to pregnancy (such as spontaneous coronary artery dissection and peripartum cardiomyopathy) and associated echocardiographic findings. Finally, we will discuss later life echocardiographic findings among individuals with prior hypertensive disorders of pregnancy.

4:00

2pBAa7. Toward the clinical application of ultrasound molecular imaging in the early detection of breast cancer. Jeremy Dahl (Radiology, Stanford Univ., 3155 Porter Dr., MC 5483, Palo Alto, CA 94304, jjdahl@stanford.edu), Rakesh Bam (Asieris, Mountain View, CA), Dongwoon Hyun, Arutselvan Natarajan, Farbod Tabesh, and Ramasamy Paulmurugan (Radiology, Stanford Univ., Palo Alto, CA)

Ultrasound molecular imaging has large potential in the early detection of breast cancer, particularly in women with radiographically dense breasts. In ultrasound molecular imaging, a molecularly-targeted ligand attached to a microbubble binds to proteins expressed on the tumor neovasculature to produce contrast that can identify cancer at an early stage. Keys to successful ultrasound molecular imaging are: (1) a molecular target highly specific to breast cancer; and (2) a sensitive imaging system that can visualize bound microbubbles while suppressing bubble signal from the background. In addition, the system must integrate with existing ultrasound imaging workflow in breast imaging clinics. We present our current progress on the development of a real-time ultrasound molecular imaging platform that consists of a novel microbubble targeted to the B7-H3 biomarker (a vascular biomarker highly specific to breast cancer and not expressed in benign disease processes) alongside a nondestructive and real-time imaging technology, based on a neural network design for contrast imaging to enable real-time high-sensitivity imaging of the novel targeted contrast agent. Although we present the results and current status of this system in preclinical models of breast cancer, the targeted microbubble is designed to be pharmaceutical grade to allow it to be injected in humans, and the imaging technology is designed such that it is compatible with current clinical workflow and operation of breast ultrasound imaging.

4:30–5:00
Panel Discussion

TUESDAY AFTERNOON, 9 MAY 2023 BELMONT, 1:00 P.M. TO 4:40 P.M.

Session 2pBAb

Biomedical Acoustics, Engineering Acoustics and Physical Acoustics: New Technology Developments for use in Focused Ultrasound Therapy II

Lawrence A. Crum, Chair
Applied Physics Lab, University of Washington, 4662 175th Ave., SE, Bellievue, WA 98006

Invited Papers

1:00

2pBAb1. Instrumentation for histotripsy ultrasound therapy: Evolution toward the clinic. 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), Jon Cannata (Histosonics, Inc., Ann Arbor, MI), J. Brian Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy is a method of ultrasound therapy using very intense, focused sound fields to fragment tissue causing controlled and localized necrosis principally through mechanical actions of cavitation. Highly specialized instrumentation (transducers and electrical drivers) is required to produce these extreme sound fields for research into cavitation physics and therapeutic applications of histotripsy. This paper will describe the evolution of histotripsy instrumentation from the first devices up to the latest phased-array systems with hundreds of channels and transmit-receive capability. While early histotripsy systems used tone bursts generators with limited bandwidth (called "class-D" circuits) for transducer excitation, more recent ones are based on a switched inductor principle to produce shorter duration acoustic emissions. This method has been tested for peak excitation voltages up to 5 kV and 40 A in compact modular systems suitable for phased-arrays. Receive of acoustic signals by the therapeutic array for aberration correction and cavitation mapping is achieved by sensing current generated by the transducer greatly simplifying transmit-receive circuitry while maintaining very large dynamic range. A novel distributed architecture was developed for data handling and signal processing using off-the-shelf FPGA evaluation modules interfaced by ethernet. These latest features will be key to successful clinical translation and widespread adoption of histotripsy.

1:20

2pBAb2. Technological considerations for delivering focused ultrasound exposures to the spinal cord. Meaghan O'Reilly (Sunnybrook Res. Inst., 2075 Bayview Ave., Rm C736a, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

Focused ultrasound opening of the blood-brain barrier (BBB) has reached clinical investigations and seems poised to revolutionize the treatment of brain disorders. Far less attention has been paid to the spinal cord and functionally equivalent blood-spinal cord barrier (BSCB). Small and large animal studies demonstrate that ultrasound and microbubbles can be used to modulate the BSCB in a similar manner to the BBB. However, the complex bony geometry of the vertebral column poses a challenge to the non-invasive delivery of therapeutic ultrasound exposures to the spinal canal. This talk will review work developing spine-specific ultrasound pulse trains and identifying optimal apertures for maximizing sound transmission through the vertebral arch.

1:40

2pBAb3. Sonobiopsy for expanding the application of therapeutic ultrasound in the diagnosis of brain diseases. Hong Chen (Washington Univ. in St. Louis, 6338 Washington Ave., University City, MO 63130, chenhongxjtu@gmail.com)

Focused ultrasound (FUS) was introduced for brain applications in the 1940s and has been employed as a therapeutic tool since then. FUS-mediated thermal ablation treatment of essential tremors was approved by the U.S. Food and Drug Administration (FDA) in 2016. FUS combined with microbubbles for inducing BBB disruption to deliver neurotherapeutic drugs to the brain is currently in early-phase clinical trials. We proposed to use FUS as a brain disease molecular diagnostic tool. Sonobiopsy uses FUS to release brain disease-specific biomarkers (e.g., DNA, RNA, and proteins) from the brain to the blood for the diagnosis of brain diseases by blood tests. Sonobiopsy does not involve any neurotherapeutic drugs, which may accelerate FDA approval of the FUS technique. Sonobiopsy opens new research opportunities that have the potential to substantially improve the diagnosis of a broad spectrum of brain tumors. This talk will present recent progress in the development of sonobiopsy technique for the diagnosis of brain cancer and Alzheimer's disease. This talk also will present recent achievements in the clinical translation of sonobiopsy.

2:00

2pBAb4. Therapeutic ultrasound to market. Kyle P. Morrison (Sonic Concepts, Inc., 18916 North Creek Parkway, Ste. 115, Bothell, WA 98011, kmorrison@sonicconcepts.com) and Francisco Chavez (Sonic Concepts, Inc., Bothell, WA)

Therapeutic ultrasound is proving to be viable in the clinic and is producing a high level of commercial activity in Therapeutic Ultrasound Systems globally across a wide variety of clinical indications. Recently there has been an increased level of commercial interest in systems delivering lower intensity ultrasound. Some examples of clinical indications include modulating nerves, sonodynamic therapy, or combining ultrasound with drugs to lower the threshold of required intensities. Regardless of the acoustic intensity required, new product inception will adopt an effective bio-acoustic result to match a clinical indication, creating a gap in product development. Sonic Concepts provides transmit ultrasound development services to fill this gap through a phased development program, accelerating first-in-man and time to market. Sonic Concepts has co-developed 30 + clinical systems, servicing customers with more than 100 years of accumulated experience, creative design techniques, and a spectrum of products and intellectual property to leverage. Examples of co-developed clinical systems will be presented, detailing the process from initial concept to first-in-man while using pre-clinical systems, HIFUPlex™ and NeuroFUS® to fast-track development.

2:20

2pBAb5. Technology development for acoustic hemostasis. Lawrence A. Crum (Ultrasound Technologies, 4662 175th Ave., SE, Bellevue, WA 98006, lacuw@uw.edu)

In trauma cases, the often immediate (and necessary) requirement is to stop uncontrolled bleeding; indeed, in combat, more than 50% of deaths result from exsanguination. With the development of ultrasound image-guided High Intensity Focused Ultrasound (HIFU) devices, "Acoustic Hemostasis" is not only possible but has the ability to terminate uncontrolled bleeding as well as for many applications of considerable benefit to medicine. In this lecture, various technology development devices and methodologies will be described that have the potential for commercial development for a wide variety of medical indications.

Contributed Papers

2:40

2pBAb6. Feasibility of MRI-guided focused ultrasound-mediated intranasal delivery in a large animal model. Siaka Fadera (Biomedical Eng., Washington Univ. in St. Louis, 4511 Forest Park Ave., St. Louis, MO 63108, s.fadera@wustl.edu), Lu Xu, Chih-Yen Chien, Yimei Yue (Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO), Dezhuang Ye (Mech. Eng. and Material Sci., Washington Univ. in St. Louis, St. Louis, MO), Chinwendu Chukwu, and Hong Chen (Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO)

Intranasal delivery provides non-invasive delivery of drugs from the nose to the brain bypassing the blood-brain barrier (BBB). We previously reported the focused ultrasound (FUS)-mediated intranasal delivery (FUSIN) of various agents in mice using FUS. However, no study has investigated the feasibility of FUSIN in large animals. The objective of this study was to investigate the feasibility of MR-guided FUS-mediated intranasal delivery in a large animal model. Pigs were used as the large animal model due to their clinical relevance to humans. Fluorescence-labeled albumin was mixed with an MRI contrast agent and administrated intranasally to the pig using a catheter. Followed by intravenous injection of microbubbles, FUS sonication was performed at two brain regions, the cortex and brainstem. T1-weighted MR images showed enhancement of MRI contrast agent at the olfactory region of the pig nose. Fluorescence imaging displayed enhanced fluorescence intensity at the FUS-targeted brain regions. These findings suggest that MR-guided FUS can achieve noninvasive and localized delivery of intranasally administered agents to the brain in large animals.

2:55–3:10 Break

3:10

2pBAb7. MRI-guided focused ultrasound drug delivery to DMG tumors in a mouse model. Payton Martinez (Biomedical Eng., Univ. of Colorado, Boulder, CO), Adam Green (Pediatric Neuro-Oncology, Univ. of Colorado, Aurora, CO), and Mark Borden (Biomedical Eng., Univ. of Colorado, 1111 Eng. Dr., Boulder, CO 80309-0427, mark.borden@colorado.edu)

Diffuse midline glioma (DMG) is the most common and unfortunately the most deadly brainstem tumor in childern. Current clincal treatment methods only involve palliative radiotherapy that does not significantly improve survival beyond its poor mean survival of 9 months. These tumors remain difficult to treat with chemotherapy as they maintain an intact blood brain barrier (BBB). Focused ultrasound (FUS) and microbubbles (MBs) has been shown to disrupt the BBB, allowing chemotherapeutics to enter the perenchyma. Panobinostat is one of the chemotherapeutics that has difficulty crossing the BBB, although it has promising effects *in vitro*. In this study, we hypothesized that MB + FUS could disrupt the BBB, allowing a higher concentration of Panobinostat to enter the tumor region and provide a therapeutic effect. Our results demonstrate that MB + FUS can successfully deliver Panobinostat to the pons with DMG tumors over 3-fold, while only slightly increasing the concentration of the forebrain. Passive cavitation detection during treatments showed harmonic cavitation throughout the 3-minute treatment, while not having any significant broadband cavitation. The combination of both Panobinostat and MB + FUS showed major decreases in tumor growth by ~71% by the end of the treatment weeks when compared to Panobinostat alone. A significant survival benefit was also demonstrated.

3:25

2pBAb8. Dynamics of focused ultrasound-enhanced glymphatic transportation in the mouse brain. Yan Gong (Washington Univ. in St. Louis, 4511 Forest Park Ave., St. Louis, MO 63108, yan.gong@wustl.edu), Kevin Xu (Washington Univ. in St. Louis, St. Louis, MO), Dezhuang Ye (Mech. Eng. and Material Sci., Washington Univ. in St. Louis, St. Louis, MO), Yaoheng Yang, and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

The glymphatic system, a perivascular network in the brain, regulates the exchange of cerebrospinal fluid and interstitial fluid in the perivascular space. Focused ultrasound combined with microbubbles (FUSMB) has been recently shown feasible to manipulate the glymphatic transportation by enhancing the intranasal delivery of agents in the perivascular space. The objective of this study was to reveal the dynamics of FUSMB-enhanced agent transport in the glymphatic system. A ring-shaped FUS transducer was confocally aligned with an objective of a two-photon microscope (2PM). Fluorescently labeled albumin was administered to the mouse nose, where it transported along olfactory nerve and trigeminal nerve from nose to brain and then spread in perivascular space in the brain. FUS sonication was performed after intravenous injection of microbubbles. *In vivo* 2PM recorded the dynamics of agent transportation before, during, and after FUS sonication. Time-lapse recording showed that FUSMB enhanced the agent accumulation at the perivascular space. FUS significantly increased the accumulation of the agent in the perivascular space by 1.2-folds immediately after the sonication ($p = 0.0325$), and 1.6-folds at 5 min after the sonication ($p < 0.0001$). In conclusion, this study showed, for the first time, direct evidence of FUSMB-enhanced glymphatic transportation in the mouse brain.

3:40

2pBAb9. Binary acoustic metasurfaces for transcranial focused ultrasound. Zhongtao Hu (Washington Univ. in St. Louis, Radiation Oncology, St. Louis, MO 63108, zhongtaohu@wustl.edu), Yaoheng Yang, Lu Xu, Yao Hao, and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

Transcranial focused ultrasound is a promising technique for noninvasive and spatially targeted diagnosis and treatment of brain diseases. Acoustic lenses were designed to correct the skull-induced beam aberration, but these designs could only generate static focused ultrasound beams inside the brain. Here, we designed and 3D printed binary acoustic metasurfaces (BAMs) for skull aberration correction and dynamic ultrasound beam focusing. BAMs were designed by binarizing the phase distribution at the surface of the metasurfaces. The phase distribution was calculated based on time reversal to correct the skull-induced phase aberration. The binarization enabled the ultrasound beam to be dynamically steered along wave propagation direction by adjusting the operation frequency of the incident ultrasound wave. The designed BAMs were manufactured by 3D printing with two coding bits, a polylactic acid unit for bit "1" and a water unit for bit "0." BAMs for single- and multi-point focusing through the human skull were designed, 3D printed, and validated numerically and experimentally. In summary, the proposed BAMs with subwavelength scale in thickness are simple to design, easy to fabric, and capable of correcting skull aberration and achieving dynamic beam steering.

3:55

2pBAb10. Airy beam-enabled binary acoustic metasurfaces (AB-BAM). Zhongtao Hu (Washington Univ. in St. Louis, Radiation Oncology, St. Louis, MO 63108, zhongtaohu@wustl.edu), Yaoheng Yang, Lu Xu (Washington Univ. in St. Louis, St. Louis, MO), Yun Jing (Acoust., Penn State Univ., State College, PA), and Hong Chen (Washington Univ. in St.Louis, St. Louis, MO)

Airy beams have offered great opportunities for ultrasound beam manipulation. However, one critical barrier that limits the broad applications of Airy beams in ultrasound is the lack of simply built device to generate Airy beams in water. This work presents a family of Airy beam-enabled binary acoustic metasurfaces (AB-BAMs) for underwater ultrasound beam manipulation. AB-BAMs are designed and fabricated by 3D printing with two coding bits: a polylactic acid unit acting as a bit "1" and a water unit acting as a bit "0." The distribution of the binary units is determined by the pattern of Airy beam. We demonstrate the capability of AB-BAMs in flexibly tuning the focal region size and beam focusing in 3D space. The focal depth of AB-BAMs can be continuous and electronical tuned by adjusting the operating frequency of the planar transducer without replacing the AB-BAMs. The superimposing method is leveraged to enable the generation of complex acoustic fields. Furthermore, the proposed 3D-printed AB-BAMs are simple to design, easy to fabricate, and low-cost to produce with the capabilities to achieve tunable focal size, flexible 3D beam focusing, arbitrary multipoint focusing, and continuous steerability, which creates unprecedented potential for ultrasound beam manipulation.

4:10

2pBAb11. Shaveless focused-ultrasound-induced blood-brain barrier opening in mice. Lu Xu (Washington Univ. in St. Louis, St. Louis, MO 63108, xu.lu@wustl.edu), Yan Gong, Chih-Yen Chien, and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

To demonstrate that focused ultrasound-induced blood-brain barrier opening (FUS-BBBO) in mice can be achieved without shaving hairs. We performed FUS-BBBO in mice by using oil as the coupling medium without shaving hairs. The hydrophobic nature of oil leads to a higher affinity to hair than water-based ultrasound gel. FUS-BBBO outcome was compared under three conditions: "oil + hairs," "gel + hairs," and "gel + no hairs." T2-weighted, T1-weighted MRI, and fluorescence imaging of the *ex vivo* brain slices were performed to measure the quality of coupling and outcome of FUS-BBBO. Results showed that "oil + hairs" consistently achieved high-quality acoustic coupling without trapping air bubbles (Figures A & B). FUS-BBBO outcome was not significantly different between the "oil + hairs" group and the "gel + no hairs" group based on T1-weighted MRI (Figures C & D) and *ex vivo* fluorescence imaging (Figure E). The FUS-BBBO efficiencies for both the "oil + hairs" and "gel + no hairs" groups were significantly higher than the "gel + hairs" group. This study demonstrated that FUS-BBBO in mice could be achieved without shaving hairs. Oil provides a simple solution for achieving effective acoustic coupling for transcranial FUS procedures.

4:25

2pBAb12. Sonoreperfusion with fibrin-targeted phase shift microbubbles for the treatment of microvascular obstruction. Soheb Anwar Mohammed (Dept. of Medicine, Univ. of Pittsburgh, 3350 Scaife Hall, Pittsburgh, PA 15213, som57@pitt.edu), Muhammad Wahab Amjad, Xucai Chen (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Maria F. Acosta, Dillion Hanrahan, Emmanuelle J. Meuillet, Evan C. Unger (Microvascular Therapeutics, Tucson, AZ), and John J. Pacella (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

More than 1 million Americans yearly have a new or recurrent acute myocardial infarction (AMI). Although the mortality rate from AMI has decreased in recent years, post-MI congestive heart failure is increasing due to microvascular obstruction (MVO), ultimately limiting myocardial salvage. We aim to address this unmet need by devising an image-guided therapy called sonoreperfusion (SRP) to resolve MVO by ultrasound-targeted microbubble cavitation (UTMC). In this study, we hypothesized that fibrin-targeted phase shift microbubbles (FTPSMBs; ∼200 nm) (Microvascular Therapeutics, Inc) would improve SRP efficacy compared to fibrin-targeted microbubbles (FTMBs; 1–3 μm) to treat MVO owing to the smaller size and more efficient microthrombi penetration. A rat hindlimb model of MVO was created by directly injecting freshly prepared porcine microthrombi into the left femoral artery under contrast-enhanced ultrasound imaging (CEUS) guidance, and treated with UTMC (1 MHz, 1.5 MPa, 5 ms pulse duration, 5-sec pulse interval) with concomitant administration of FTPSMBs/FTMBs (3 mL/hr). The treatment effect was accessed with CEUS. UTMC with FTPSMBs caused more rapid and complete reperfusion of rat hindlimb following MVO compared with FTMB, likely owing to their small size and more effective thrombus penetration. Studies to explore the underlying molecular mechanisms associated with SRP treatments are underway.

TUESDAY AFTERNOON, 9 MAY 2023 LINCOLNSHIRE 1/2, 1:00 P.M. TO 3:15 P.M.

Session 2pCA

Computational Acoustics, Structural Acoustics and Vibration, Physical Acoustics, Musical Acoustics, Acoustical Oceanography, and Noise: Wind Acoustics

D. Keith Wilson, Cochair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineer Research and Development Center (U.S Army ERDC-CRREL), 72 Lyme Rd., Hanover, NH 03755-1290

Gregory W. Lyons, Cochair
DEVCOM Army Research Laboratory, 2800 Powder Mill Rd., Adelphi, MD 20783

Chair's Introduction—1:00

Invited Papers

1:05

2pCA1. Mitigating wind noise in multichannel active noise control systems. Sipei Zhao (Ctr. for Audio, Acoust. and Vib., Univ. of Technol. Sydney, 32-34 Lord St., UTS Tech Lab, Botany, New South Wales 2019, Australia, sipei.zhao@uts.edu.au) and Yijing Chu (State Key Lab. of Subtropical Bldg. Sci., South China Univ. of Technol., Guangzhou, China)

Multichannel active noise control (ANC) systems have been investigated to reduce outdoor low-frequency noise, such as power transformer noise and traffic noise. However, in outdoor environments, wind noise is inevitable and significantly degrades the noise reduction performance of active control systems. Despite broad adoption in windy environments, porous microphone windscreens cannot eliminate low-frequency wind noise in microphones, especially when the wind speed is high. Assuming wind noise is additive and uncorrelated with the primary noise to be controlled, this paper analyses how the wind noise detected by the reference microphone impacts the control filters and the residual noise of a multichannel ANC system based on the multichannel filtered-reference least mean squares algorithm (MFxLMS). By extending our recent research on wind noise suppression in single-channel active noise control systems [Y. Chu, *et al.*, J. Acoust. Soc. Am. **152**(6), 3340–3345 (2022)], a modified MFxLMS algorithm based on a new cost function is proposed. Simulations with real-recorded wind noise are performed to verify the theoretical analysis.

1:25

2pCA2. Pseudospectrum-based methods for estimating the wind speed and direction based on closely spaced microphone signals. Daniele Mirabilii (WS Audiology, Henri-Dunant-Straße 100, Erlangen, Bavaria 91058, Germany, daniele.mirabilii@wsa.com) and Emanuël A. Habets (Int. Audio Labs. Erlangen, Erlangen, Bavaria, Germany)

Acoustic array processing can be employed to measure the wind speed and direction based on microphone signals. Turbulent pressure fluctuations picked up by microphones are referred to as wind noise. According to Taylor's frozen turbulence hypothesis, turbulent eddies retain their shape while advecting at nearly the mean wind speed and in the wind direction. It follows that wind noise propagates accordingly across a microphone array when the inter-microphone distance is smaller than the turbulence wavelength. This property can be exploited to track the orientation of the turbulence advection, and hence to characterize the wind flow. We propose beamforming and signal subspace-based methods to estimate the wind speed and direction using a compact microphone array. In particular, the pseudospectrum of measured wind noise is computed against candidate pairs of wind speed and direction. The wind speed and direction estimates are then obtained as the maximizers of the pseudospectrum. In addition, we extend an existing time difference of arrival-based method originally derived for three microphones to an arbitrary number of microphones. We evaluate the estimation accuracy of the proposed methods separately for the wind speed and direction. Possible applications include highly integrable, portable, and inexpensive anemometers for smart sensors or action cameras.

1:45

2pCA3. Characterization of infrasonic wind noise using the Discrete Orthonormal Stockwell Transform. Garth Frazier (NCPA, Univ. of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu)

The Discrete Orthonormal Stockwell Transform (DOST) (Stockwell, 2007) projects a finite-length, periodically sampled signal onto a set of orthonormal basis vectors that are localized in time and scale such that all basis functions at each scale are represented by a finite number of Discrete Fourier Transform (DFT) basis functions. It is based on the Stockwell Transform (Stockwell *et al*., 1996), sometimes referred to as the S-transform, which is applicable to continuously defined waveforms. Furthermore, the DFT basis functions corresponding to each DOST scale are non-overlapping. The shorter DOST scales correspond to higher-frequency DFT basis functions and greater bandwidths than the longer DOST scales. Thus, the DOST is a type of time-frequency transform and is suitable for non-stationary time-series analysis. Additionally, the DOST of an N-point record can be calculated efficiently in order N log N operations (Wang and Orchard, 2009). Several applications of the S-transform and the DOST have been documented including beamforming with array data (Collar and Frazier, 2022; Frazier, 2022). In this presentation, the DOST used is to examine the statistics of data measured with infrasound sensors as a function of the DOST scales, adaptively characterize the amplitude distributions of the DOST coefficients at each scale and perform near-optimal detection of transient signals. Results corresponding to infrasound data recorded under various conditions are presented.

2:05–2:20 Break

2:20

2pCA4. Progress in understanding seismic wind noise. Richard Raspet (NCPA, Univ. of MS, 145, Hill Dr., University, MS 38677, raspet@olemiss.edu), Craig J. Hickey, and Md. A. Samad (NCPA, Univ. of MS, University, MS)

Previous work at NCPA on predicting and measuring wind noise in the ground used a quasistatic calculation employing Shield's wind noise correlation function. This method predicted the vertical displacements at the surface well, but under predicted the horizontal displacement and predicted a rapid decay of levels with depth that was not observed in the measurements. A new measurement of the wind noise correlation coefficients using flush mount microphones was presented at the Seattle Meeting (2aPa1). In this talk additional measurements and improved methods for evaluating the coefficients are described. An improved calculation of the seismic levels due to wind induced pressures at the ground has been developed based on the wavenumber-frequency analysis developed for studying airframe noise due to turbulence. This calculation correctly predicts the surface vertical component and the near constant seismic level in the first 40cm of the ground. It still under predicts the horizontal component, but the under prediction is smaller than that predicted by the quasistatic calculation. [This research was sponsored by the Department of the Navy, Office of Naval Research under ONR Award No. N00014-18-2489.]

2:40

2pCA5. Tornado infrasound: Observations and potential mechanisms. Brian R. Elbing (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK, elbing@okstate.edu), Trevor C. Wilson, Chris Petrin, Real J. KC, Taylor Swaim, and Bryce B. Lindsey (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK)

Tornado warnings have not improved over the past 20 years, which is especially true in hilly terrain where radar cannot see near the ground due to the curvature of the earth. While Tornado Alley is best known for tornadoes, most tornado related deaths occur in the southeastern US where hilly terrain is more prevalent. Tornadoes emit sounds at frequencies below what humans can hear (infrasound), and there is strong evidence that these sounds carry information about the forming of the tornado as well as its size. In addition, these very low frequencies can travel well beyond the line-of-sight. Currently, this information is not used to guide warnings because we do not understand what makes it. Our team has been working on identifying the fluid mechanism for the past few years and there are four commonly proposed mechanisms that are consistent with observations; radial oscillation, latent heat effect, pressure relaxation, and shear instability. In this presentation each mechanism will be discussed relative to available observations. In addition, an overview of current field and laboratory testing motivated by the success and limitations of each proposed mechanism will be discussed.

Contributed Paper

3:00

2pCA6. Infrasound noise mitigation on high altitude balloons. Taylor Swaim (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK, tswaim@okstate.edu), Kate Spillman, Emalee Hough, Zach Yap, Jamey Jacob, and Brian R. Elbing (Mech. and Aerosp. Eng., Oklahoma State Univ., Stillwater, OK)

Wind noise significantly limits the use of infrasound for remote sensing using ground-based sensors. Recently, there has been a surge in studies using high altitude balloons, termed heliotropes, to mitigate this problem. Since the balloon drifts with the wind there is minimal relative speed between the balloon and the local air. To resolve the direction of arrival from a signal, there must be sufficient separation between sensors. However, this separation increases the relative motion between the sensors and the local air. On Earth this has shown to have a weak dependence, but the use of such measurements are currently being considered as a means of exploring the interior of Venus. Vertical shear on Venus is between 5 and 10 m/s per km, which will create approximately 1 m/s of relative wind speed per 100 m of separation. Thus, the current work aims to identify wind noise reduction schemes for balloon based infrasound sensing on Venus using an Earth analog. This presentation will report on a variety of windscreen configurations examined, ground-based testing of the designs, and heliotrope flight testing of windscreens.

TUESDAY AFTERNOON, 9 MAY 2023 NORTHWESTERN/OHIO STATE, 1:00 P.M. TO 4:00 P.M.

Session 2pEA

Engineering Acoustics, Signal Processing in Acoustics, Architectural Acoustics, Underwater Acoustics, Physical Acoustics, and Noise: Microphones: Design, Development, and Characterization II

Thomas E. Blanford, Cochair
The Pennsylvania State University, State College, PA 16804

Zane T. Rusk, Cochair
The Pennsylvania State University, 104 Engineering Unit A, University Park, PA 16802

Invited Papers

1:00

2pEA1. A new sixth-order Eigenmike® spherical microphone array for spatial sound field recording. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com), Jens Meyer (mh Acoust., Fairfax, VT), and Steven Backer (mh Acoust., Truckee, CA)

A new spherical microphone array capable of up to 6-th order spatial sound field recording is described. Spherical microphone array beamforming is normally accomplished by judicially combining multiple acoustic pressure microphones appropriately mounted on the surface of a rigid sphere to form a set of orthonormal spherical harmonic modal beams or “Eigenbeams” (also referred to as Higher-Order-Ambisonics). Eigenbeamformer processing exploits the diffraction and scattering of the incident acoustic waves onto the rigid sphere. General beamformers are then created by properly combining the orthonormal Eigenbeams to form and steer a desired set of beampatterns up to the decomposition order limit. One advantage of using the modal decomposition approach to beamforming is that it is computationally compact, and results in an elegant scalable beamformer architecture. Steering beams formed by the spherical array is accomplished by a computationally simple matrix multiply. Another benefit is that beampatterns formed using spherical array processing maintain their design spatial responses independent of the steering direction. Beampatterns can also be frequency independent. In this talk, we will describe a new 64-element spherical microphone array and associated software capable of up to 6-th order performance, as well as present some real-world measurements.

1:20

2pEA2. Miniature optical MEMS microphone with 14dBA noise floor. Jakob Vennerod (sensiBel AS, Gaustadalléen 21, Oslo 0349, Norway, jakob.vennerod@sensibel.com) and Matthieu Lacolle (sensiBel AS, Oslo, Norway)

This paper explains the fundamental technology used to create an optical microphone transducer. In recent years, microelectromechanical system (MEMS) capacitive microphones have demonstrated improved performance. State-of-the-art capacitive MEMS microphones can achieve SNR in the order of 73 dBA (21 dBA noise floor) with overall dynamic range in the order of 101 dB. There are fundamental challenges to driving the performance of capacitive MEMS microphone technology in very small packages to new heights. Piezoelectric MEMS microphones have not demonstrated SNR performance >65 dBA. The next breakthrough in miniature microphone technology will come from optical MEMS microphone technology. 80dB SNR (14 dBA noise floor) with 132 dB dynamic range (146 dB maximum sound pressure level) has been achieved in a very small package. This paper will review the fundamentals of optical acoustic transduction and describe some of the approaches to miniaturization of the technology.

1:40

2pEA3. The design of an optomechanical microphone using a photonic waveguide interferometer. Xiaoyu Niu (Chandra Family, Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, 2501 Speedway, EER 4.822, Austin, TX 78712, xyniu@utexas.edu), Yuqi Meng, Ehsan Vatankhah, and Neal A. Hall (Chandra Family, Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Most commercial microphones rely on capacitive or piezoelectric readout of a vibrating diaphragm. Optical methods have also been used to read the motion of a vibrating microphone diaphragm. Most optical microphone demonstrations to date use free-space optics. We present the design of an optical interferometric microphone that uses a photonic waveguide embedded within the diaphragm. The deflection of the moving diaphragm causes a strain field within the diaphragm, and this strain in turn changes the optical path length of the waveguide. Light traversing the sensing arm of the interferometer combines with light traversing an on-chip reference arm to yield an output light power that depends on the instantaneous displacement of the diaphragm. The output of the combined beams is coupled from the chip to a photodetector to complete the interferometer. An advantage of such transduction method is the absence of a microphone backplate and its associated thermal-mechanical noise. Electrical noise associated with high input-impedance amplification, as required for small capacitive and piezoelectric sensors, is also circumvented. This embodiment does, however, introduce several implementation challenges including the routing of light onto and off the microphone diaphragm.

Contributed Papers

2:00

2pEA4. Optimized directionality in acoustically coupled, dual motor, uniphase, moving coil microphones. Roger Grinnip (Shure, Inc., 5800 W. Touhy Ave., Niles, IL 60714, grinnir@shure.com)

In uniphase, moving coil microphones, pneumatic vibration cancellation is an effective means to minimize structural excitation. However, pneumatic vibration cancellation requires precise control of the transfer function between the capsule and the microphone body, and the internal volumes are large. Therefore, the uniphase, moving coil capsule with pneumatic vibration cancellation is not well suited as a portable platform in applications that require a compact size. The addition of a secondary transduction mechanism is commonly used to mitigate a variety of unwanted signal sources in microphones, and is an effective alternative to the pneumatic vibration cancellation. Because the secondary transduction mechanism is integral to the capsule, precise control of the capsule/body transfer function is not needed, resulting in a more portable solution. However, the current strategy for vibration cancellation is to isolate the secondary transduction mechanism from the acoustical excitation, which does not minimize the internal volume and is not compact. A new strategy proposes an alternative architecture where the two transduction mechanisms share an internal volume. When certain constraints are met, acoustically coupling the motors results both a minimized internal volume and an optimized microphone directionality at low frequencies. Simulated and measured results are shown to illustrate the implementation.

2:15

2pEA5. Parylene as an electret for condenser microphones. Benjamin J. Grigg (Shure Inc., 5800 W Touhy Ave., Niles, IL 60714, griggb@shure.com)

Electrets in condenser microphones are typically made from varieties of Teflon. This work evaluates different formulations and thicknesses of vapor deposited Parylene electrets as potential alternatives to Teflon. Parylene-coated condenser backplates used in this study were obtained from two different suppliers, as a third variable in addition to formulation and thickness. Long-term charge stability of the electrets was tested with a thermally stimulated discharge (TSD) method. Charge stability was also evaluated by building the charged backplates into condenser microphones and running them through environmental accelerated-life testing. Standard PTFE and FEP Teflon backplates were used as controls in these tests. The experiments showed clear differences in the long-term charge stability of the Parylene electrets due to thickness, formulation, and supplier. Several combinations of thickness, formulation, and supplier were found to perform comparably to traditional Teflon electrets and appear suitable for use in high-end condenser microphones.

2:30–2:45 Break

2:45

2pEA6. Novel design method for sub-miniature MEMS microphones with enhanced acoustic SNR. Michael Pedersen (Knowles LLC, 1151 Maplewood Dr., Itasca, IL 60143, michael.pedersen@knowles.com), Vahid Naderyan, and Peter Loeppert (Knowles LLC, Itasca, IL)

A well understood performance limitation in the conventional design of MEMS microphones relates to the dimensions of the back volume, which, along with the barometric vent, sets the ultimate limit of acoustic noise in the system, regardless of the microphone transduction principle. The acoustic noise in the back volume is generated from the heat flow within the thermal boundary layer near the isothermal walls. In this paper, a new and previously unused part of the back volume design space is explored, in which the primary loss mechanism is thermal (as opposed to viscous), and the effective thermal conductivity and heat capacitance of the gas in the volume is designed to achieve low acoustic noise in extremely small enclosures. In this new regime, the acoustic SNR in frequency bands of interest can be optimized to achieve values comparable or even larger than those in the conventional design space with much larger enclosures. The primary trade-off in design relates to achievable acoustic SNR versus necessary acoustic compliance with the selected transduction principle. Underlying principles and design examples are presented with supporting measurements from prototype devices, showing acoustic SNR in excess of 70 dB with a back volume of less than 0.2 mm^3.

3:00

2pEA7. A compact high performance micro-electro-mechanical systems microphone. Yunfei Ma (Knowles Corp., 1151 Maplewood Dr., Itasca, IL 60143, yunfei.ma@knowles.com), Shubham Shubham, Michael Kuntzman, Jen-I Cheng, and Jing Ouyang (Knowles Corp., Itasca, IL)

This paper reports on the development of a MEMS capacitive microphone design with 72 dBA signal-to-noise-ratio (SNR) in a compact $3.4 \times 2.3 \times 0.7\,mm^3$package. The design incorporates a circular diaphragm disc suspended on one end of the cantilever beam. The diaphragm, under the bias conditions, is supported using peripheral and center protrusions extended from the back plate. The design optimization is targeted to achieve high sensitivity and low damping noise to achieve maximum SNR possible in the mentioned footprint. Finite element modeling (FEM) combined with the lumped element circuit modeling have been implemented to realize the microphone performance. The results have been validated against the measurement with very good correlation of sensitivity, noise and total harmonic distortion (THD). With the sensitivity of −35 dBV (ref. 1 V/Pa at 1 kHz) and acoustic overload point of 134 dBSPL, this is one of the highest performing MEMS analog microphone reported today. Therefore, it is very well suited for audio applications such as mobile phones, true wireless stereo (TWS) earphones, hearing aids and automotive, which demand miniaturized size, low cost and high performance.

3:15

2pEA8. Bio-inspired ultra-sensitive directional acoustic flow sensing using electro-spun nanofiber mesh over a close-back cavity. Junpeng Lai (Mech. Eng., Binghamton Univ., 3205 Burris Rd., Apt. C12, Vestal, NY 13850, jlai16@binghamton.edu), Mahdi M. Farahikia (Div. of Eng. Programs, SUNY New Paltz, New Paltz, NY), Morteza Karimi (Mech. Eng., Binghamton Univ., Vestal, NY), Zihan Liu, Changhong Ke, and Ronald Miles (Mech. Eng., Binghamton Univ., Binghamton, NY)

Nearly two decades ago, it is shown that the mechanically coupled tympanic membrane ear of the *Ormia fly* can detect directional sound from a distance [1]. It achieves that by sensing the pressure gradient across the two coupled membranes. The recently discovered acoustic flow sensing using spider silk [2], opens new perspectives in acoustic sensing of particle velocity. Here we demonstrate an ultra-sensitive directional sensing scheme using a nanofiber mesh that is uniformly placed over the surface of a close back micro cavity. The nanofiber mesh can capture sound-induced velocity fluctuation well enough to represent the acoustic air particle velocity. Our result shows that the uniform sound field, which travels parallel to the fiber mesh plane, can drive the fiber mesh in and out of the cavity underneath. Since the small size of the cavity makes the air inside incompressible, the front and rear sides of the fiber mesh will move out of phase, much like the directional ear of the *Ormia fly*. By sensing acoustic air particle velocity and adopting the practical design of a back volume, this work provides a new approach for ultra-sensitive directional acoustic sensing. [1] R. N. Miles, D. Robert, and R. R. Hoy, "Mechanically coupled ears for directional hearing in the parasitoid fly Ormia ochracea," *JASA*, **98**(6), 3059–3070 (1995). [2] J. Zhou and R. N. Miles, "Sensing fluctuating airflow with spider silk," *Proc. Natl. Acad. Sci. U. S. A.* **114**(46), 12120–12125 (2017).

3:30

2pEA9. Design and testing of ultraminiature microelectromechanical systems middle ear accelerometers. Panagiota Kitsopoulos (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, pkitsop@umich.edu), Alison E. Hake (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Emily Z. Stucken, Christopher M. Welch (Otolaryngology-Head and Neck Surgery, Univ. of Michigan, Ann Arbor, MI), and Karl Grosh (Mechanical Eng., Univ. of Michigan, Ann Arbor, MI)

Moderate to severe hearing loss is a debilitating condition that affects over 5% of the world's population. Hearing aids and cochlear implants positively impact the lives of those who suffer from sensorineural hearing loss. However, both display numerous limitations that affect their adoption and use rates, notably including those associated with the external elements of these devices (e.g., microphone and signal processor). These external components impact device safety, appearance, acoustic performance, and ease-of-use. A totally implantable auditory prosthesis would help to address these issues by eliminating external components. A major barrier to progress toward this goal is the lack of a completely implantable acoustic sensor capable of matching or exceeding the performance of commercial external microphones. Our previous studies have indicated that piezoelectric microelectromechanical systems (MEMS) accelerometers have the potential to function as implantable sensors within the middle ear meeting a 20-phon noise floor over a 100–8 kHz range. In this paper, we describe a process to design and fabricate dual and tri-resonance devices that are comprised of piezoelectric cantilever bimorph beams tip-loaded by a proof mass, can meet or exceed the 20-phon noise floor, and can produce a measurable voltage output when deflected by sound-induced ossicular vibrations.

3:45

2pEA10. Speakers—As a sensor and actuator for ear condition monitoring. Noori Kim (School of Eng. Technol., Purdue Univ., 401 N. Grant St., Rm. 115, West Lafayette, IN 47907, kim4147@purdue.edu) and Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Acoustic devices such as microphones and speakers are critical tools in modern hearing care research. When utilizing acoustic data for such devices, most approaches drive sound signals from speakers and obtain corresponding electrical signals from microphones. However, significant opportunities and valuable information are missed if researchers conclude that a speaker is just a sounding device. It should be noted that traditional acoustic pressure measurement with microphones requires complex signal processing, such as noise filtering, to overcome crosstalk and noises from various sources. This approach engineers a speaker as an actuator and a sensor simultaneously. Electrical input impedance from an earphone's speaker will be monitored and measured to detect acoustic load statuses such as ear conditions such as blocked or opened. Speakers are two-port electroacoustic transducers, based on our previous work, and they can be used as more powerful tools to provide important physiological information related to hearing sensitivities and acoustic load changes. The electrical input impedance data contain static information of acoustic loads. It is feasible that this additional information captured from the speaker may supplement the acoustic pressure measurement to enhance the microphone's data when an acoustic probe is used for clinical use.

Session 2pED

Education in Acoustics, Physical Acoustics, Engineering Acoustics, Architectural Acoustics, and Noise: Assessment of Acoustics Education

Andrew A. Piacsek, Cochair
Dept. of Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926-7422

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—1:00

Invited Papers

1:05

2pED1. Program assessment in undergraduate acoustics at Kettering University. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu) and Ronald E. Kumon (Kettering Univ., Flint, MI)

Kettering University has been teaching acoustics as part of undergraduate engineering for decades, and the senior-level acoustics laboratory course has assumed a critical role in the continuous improvement process for our ABET-accredited programs in Applied Physics and Engineering Physics. This project-based course addresses learning objectives focused on transferable skills in test and measurement, literature and theoretical foundations, and computational modeling. These are set in the context of two modules, on sound source directivity and modal testing. The assessment structure is built around rubrics, used for both formative and summative evaluation of student performance on authentic tasks including recording activity in lab notebooks and scientific writing. A 32-question multiple choice instrument is deployed as a pre-test and post-test to assess gains in content knowledge from the course. Finally, we discuss specific examples of assessment for effective teamwork (ABET EAC Outcome 5) and developing and conducting experiments and analyzing and interpreting data (ABET EAC Outcome 6).

1:25

2pED2. Using standards-based assessment and reporting in a general education acoustics course. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

Standards-based assessment and reporting is a method of grading in which each student is responsible for demonstrating mastery of a set of course objectives called standards. In this course, students are assessed on each standard using short assessments that resemble a quiz, usually one or two pages long. Each standard is assigned a rating on a 5-step scale. The student's ratings throughout the semester determine the final grade the student will earn. If a student is not satisfied with a rating on an assessment, they are allowed to reassess the standard to demonstrate they have developed a fuller understanding of the topic. This method of student assessment is intended to encourage students to develop intrinsic motivation for learning the course material. Students are also not penalized for having a bad exam or quiz score due to unforeseen circumstances. Students often report that they need time at the start of the semester to get used to the grading system but that they ultimately see the benefit to their learning.

1:45

2pED3. Using the mechanical waves concept survey in a non-majors acoustics class. Kimberly A. Riegel (Phys., Farmingdale State College, 222-05 56th Ave., Bayside, NY 11364, kriegel@qcc.cuny.edu)

The Mechanical Waves Concept Survey (MWCS) is a concept inventory designed to identify common misconceptions specifically related to Waves and Optics. It is a 22 question multiple choice assessment designed to be given pre and post instruction. The development of this inventory involved many revisions and administering the test to hundreds of students ranging from high school to university students in several different countries including Mexico, Thailand, and Australia. The use and effectiveness of the MWCS in other settings such as America, community colleges and non-majors courses have not been thoroughly examined. This presentation will outline results for an American community college specifically for students who are not intending to be STEM majors. Several semesters of assessment data will be presented. Due to the change in instruction modality during COVID, a comparison between delivery methods, in person and online was examined. Future work regarding improvements to the concept inventory will be discussed.

2:05

2pED4. Early development of a vibrations concept inventory for undergraduate acoustics education. Christopher M. Jasinski (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, jasinski@hartford.edu)

Vibrations I is a required course for all mechanical engineering majors at the University of Hartford (including those in the Acoustics concentration). This builds upon a mechanics sequence from the second-year curriculum including Statics and Dynamics. While the current course layout has proven successful in achieving the program's intended outcomes for accreditation, some non-Acoustics majors have been observed to be missing some important concepts that are not routinely evaluated in exam questions. In order to understand these issues more clearly, a concept inventory for vibrations is being developed. A concept inventory is a multiple-choice assessment designed to evaluate students' understanding of the underlying concepts regarding course material. The assessment is designed in such a way that the student is challenged to determine the outcome of a scenario related to course material, and many questions will specifically target common misconceptions held by students in mechanical engineering programs. This type of assessment has been used frequently since the original Force Concept Inventory from Hestenes *et al*. (1985). This presentation will describe some of the initial work towards developing this inventory, discuss alternative assessments, and will welcome input from the Education in Acoustics community on how to best move forward with this idea.

2:25

2pED5. Development of a waves and vibration concept inventory: Challenges and applications. Andrew A. Piacsek (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

A useful tool for measuring learning gains and for assessing the effectiveness of instruction in the physical sciences is the "concept inventory." This is a set of questions administered at the beginning of a course, or prior to a unit of instruction, as well as at the conclusion. Students may told their scores, but are not provided with the answers. An early example is the Force Concept Inventory (FCI), published in 1992, which has been widely adopted for use in introductory physics courses and has served as the model for concept inventories. This presentation describes the ongoing development of an inventory designed to test conceptual understanding of basic principles of vibration and wave propagation that are typically covered in a first-year physics course or introductory or intermediate courses on acoustics-related topics. The current version builds on previous work by the author to develop a more focused inventory for an introductory course in musical acoustics (https://asa.scitation.org/doi/10.1121/1.5067886). Results from the trial use of this inventory in multiple courses at Central Washington University will be discussed.

2:45–3:00 Break

3:00

2pED6. Longitudinal standing waves, mechanical waves, and diagnostics. Jack A. Dostal (Dept. of Phys., Wake Forest Univ., PO Box 7507, Winston Salem, NC 27109, dostalja@wfu.edu)

The Standing Wave Diagnostic Test is a 22-item multiple choice survey designed to assess student understanding of longitudinal standing waves in air. I created it to measure my students' difficulties with the subject. I also created the Longitudinal Standing Wave Tutorial to address these difficulties. The tutorial uses marked springs, tuning forks, pipes, and plenty of discussion to develop students' understanding. Both the diagnostic and tutorial are based in physics education research and developed using survey and interview data from college physics classes. I often use the tutorial as a laboratory exercise in my Physics of Music class, which generates some interesting differences from the college physics classes. Some are a result of students' prior knowledge about sound, while others are influenced by the course material covered prior to the tutorial. I will discuss some of my own results with these materials and put them in context with other physics education research based mechanical wave surveys.

Contributed Papers

3:20

2pED7. Strategies for assessment of student learning outcomes via the final project in a general education acoustics course. Anne C. Balant (Dept. of Commun. Disord., State Univ. of NY at New Paltz, 1 Hawk Dr. HUM09, New Paltz, NY 12561, balanta@newpaltz.edu) and Heather L. Lai (Eng. Programs, State Univ. of NY at New Paltz, New Paltz, NY)

Learning outcomes of the final project in an undergraduate general education acoustics course were assessed via three short essay questions on an open book final exam. For the project, students construct a simple sound-making device or musical instrument that produces at least four different sounds using ordinary household objects, recyclables, etc. They are required to develop a hypothesis about how changes in the device will change the properties of the sounds. They then test their hypotheses using spectrograms, FFT measurements, and/or sound level measurements. The written responses of 60 students were downloaded and stripped of identifiers. The first two essay questions were designed to assess whether students could connect the steps of their project with the scientific method and explain the rationale for their choice of measurement methods based on their hypothesis. These results were assessed via rubrics. The third question was an open-ended reflection on what they learned about acoustics by completing the project, and how they learned it. These responses were assessed via qualitative data analysis. Results, key findings, and implications for the design of the assignment and future assessments will be discussed.

3:35

2pED8. Engaging students with team based and open-ended response projects in a descriptive acoustics course. Bonnie Andersen (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84058, bonniem@uvu.edu)

Helping non-science majors feel comfortable in a physics class can be a challenge. Inclusive pedagogy techniques encourage creating learning communities and providing more opportunities for students to engage with the content. Elements of Team Based Learning (TBL) and open-ended response projects were applied to a descriptive acoustics course. TBL allows students to work with permanent, strategically placed teams during the semester. Teams are selected to help spread the abilities uniformly for all teams to be successful. In this course, group activities, a group component of exams, and the peer evaluation were implemented. The evaluation holds students

accountable for their effects with their teams, rewarding students who work hard and minimize freeloading, an often-challenging aspect of group work. In addition to TBL, open-ended response projects were used in place of a research paper. Students can choose to research an acoustics topic or perform their own experiment. The students can also decide how they would like to present their work. The flexibility allowed students to display their results in a variety of ways from traditional written work to podcasts.

3:50

2pED9. Wind farm acoustics course for engineering students. Heather L. Lai (Eng. Programs, State Univ. of NY at New Paltz, 208 Eng. Innovation Hub, New Paltz, NY 12561, laih@newpaltz.edu) and Anne C. Balant (Commun. Disord., State Univ. of NY at New Paltz, New Paltz, NY)

The Accreditation Board for Engineering and Technology (ABET) has two learning outcomes for all engineering students that relate to "public health, safety, and welfare" and the need to consider "the impact of engineering solutions in global, economic, environmental, and societal contexts." We describe the development of a one-credit hybrid course on wind energy acoustics that meets these learning outcomes. This course was one of five one-credit modular courses on different aspects of wind energy developed via a SUNY-funded initiative. Topics included human and environmental impacts of wind farm noise, sound perception, noise regulations, wind farm noise generation and propagation, and noise control. Students gained the acoustics background needed to understand the necessary technological aspects of wind farm noise. They used simple models to predict community noise levels and assess noise control strategies and then completed a case study project in which they proposed and debated strategies for reducing the negative impacts of a controversial wind farm. Upon completion of the course, the students expressed that they had not previously considered the possible impacts of wind farm noise. They seemed to have grown in their ability to think holistically about the potential societal and environmental impacts of wind farm noise.

4:05–4:35
Panel Discussion

TUESDAY AFTERNOON, 9 MAY 2023 — CHICAGO A/B, 4:30 P.M. TO 5:30 P.M.

Session 2pMU

Musical Acoustics: Concert Session: Harp Duo Julie Spring and Ellie Kirk

James P. Cottingham, Cochair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

Gordon P. Ramsey, Cochair
Physics Department, Loyola University Chicago, Chicago, IL 60660

A concert by the harp duo Two Columns

The harp duo Two Columns consists of Chicago harpists Julie Spring and Ellie Kirk. They have played together since 2017 and always welcome exceptional challenges, achieving artistry with musical depth. Julie and Ellie have each played across the world on numerous international stages and have a passion for harp education. Both members work extensively with harp students of all ages in the Chicago area. Julie attended the Eastman School of Music for her undergraduate education and Ellie attended Columbia University. Both Julie and Ellie earned a Master of Music from Roosevelt University and have played with numerous professional orchestras including the Lyric Opera and Chicago Symphony. Ellie is the principal harpist with the Illinois Symphony Orchestra, and Julie has been principal harpist with both the Toronto and Hartford Symphony Orchestras.

Session 2pNS

Noise: Incorporating Tones in Noise Criteria

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

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

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Response variations in psychoacoustic tests focused on the assessment of tonal office noise. Guochenhao Song (Ray W. Herrick Labs., Purdue Univ., 177 S, Russell St., West Lafayette, IN 47907, song520@purdue.edu), Patricia Davies (Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN), and Yangfan Liu (Purdue Univ., West Lafayette, IN)

Tonal noise from rotating machinery is known to be a problem in buildings. To guide the design of a comfortable office environment, a proper understanding of people's annoyance due to exposure to tonal office noise is valuable. Earlier, authors have proposed and verified two annoyance prediction models based on the psychoacoustic subjective tests conducted in an office mock-up, and further developed a software to implement the metric models. In the present work, the focus is shifted to some unexpected response variations observed while analyzing the responses from subjective tests: (1) the learning pattern that occurs at the beginning of the test; (2) the significant differences in the averaged annoyance ratings with respect to the gender and the Noise-Sensitivity-Questionnaire (NoiSeQ) score; (3) the acclimation to sounds: i.e., the annoyance ratings for the 2-min sounds were lower than the ratings for the corresponding 5-s sounds; and (4) the inconsistencies in ratings: i.e., subjects tend to use the annoyance scale differently in different parts of the test and to rate sounds relative to the background level in the room. In the end, several approaches to deal with these response variations and a few recommendations on the psychoacoustic test design are summarized.

1:25

2pNS2. Applications of the psychoacoustic tonality method. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, Roland.Sottek@head-acoustics.com) and Thiago Henrique Gomes Lobato (HEAD Acoust. GmbH, Herzogenrath, Germany)

Psychoacoustic tonality calculated using the Sottek Hearing Model is an established method, standardized in ECMA 418-2 that can be used in various product assessments to identify and quantify prominent tonal components. The perception and evaluation of sound events containing such components has become increasingly important, e.g., in the field of vehicle acoustics to evaluate tonality due to alternative powertrains, or in information technology (IT) equipment due to hard disk noise. In addition, many products contain fans, e.g., IT equipment, household appliances, and air conditioning systems in buildings. Noticeable noise can emanate from these fans. However, there is no central publication describing most applications of the method, so its capabilities are not widely known. The goal of this article is to fill this gap by presenting the method clearly and concisely, and highlighting its various applications in industry. By providing a comprehensive overview of the method and its applications, we aim to help practitioners improve the quality of their audio designs by making them aware of the benefits of psychoacoustic tonality.

1:45

2pNS3. Comparison of annoyance of tones predicted from sound power versus sound pressure measurements. Gregory Meeuwsen (Trane Technologies, La Crosse, WI), Derrick P. Knight (Trane Technologies, 2313 20th St. South, La Crosse, WI 54601, Derrick.Knight@TraneTechnologies.com), and Dave Edmonds (Trane Technologies, La Crosse, WI)

ASHRAE 1707-TRP provides a means by which to predict the annoyance of tones in noise as related to building services equipment. The scope is limited to sound pressure measurements taken in a classroom, office, or other occupied space. However, product manufactures rely on sound power measurements taken in reverberation chambers to quantify noise emissions. This case study will show how sound power data taken in accordance with AHRI 260 performs in predicting tonal annoyance.

2:05

2pNS4. Evaluation of tone annoyance metrics on reverberation room measurements of HVAC equipment. Paul F. Bauch (Appl. Equipment, Johnson Controls, 100 JCI Way, York, PA 17406, paul.f.bauch@jci.com) and Rajavel Balaguru (Appl. Equipment, Johnson Controls, York, PA)

Reverberation rooms are a primary tool for HVAC equipment manufacturers to calculate equipment sound power levels, which lead directly to equipment sound ratings. While sound quality has been included in standards, namely, the withdrawn AHRI 1140, the adoption and value of those standards has been minimal leaving specifiers to construct their own tone quality metrics. This paper will evaluate the application of common sound quality metrics, specifically tonal metrics, to reverberation room measurements. Room response is a critical factor in measuring tone prominence, especially at lower frequencies where adjacent frequency tones can vary by 6dB or more. Metrics evaluated include TTNR, Prominence, Tonality and a new metric developed in ASHRAE 1707-TRP.

2:25

2pNS5. An air handling equipment manufacturer's perspective on tonal noise measurement and application. Mark W. Fly (NAIC Lab, AAON, Inc., 2425 S YT, Tulsa, OK 74011, markwfly@gmail.com)

In order to reliably predict sound from HVAC equipment, including tonal noise, sound power data needs to be determined for each noise producing component in the equipment. Though there are accepted standards for determining broad band sound in resolutions no finer than one-third octaves, there is a general lack of component data regarding tonal characteristics. This noise produced by the HVAC system in indoor environments is often an important source of masking noise improving speech privacy and the overall acoustical environment. The presence of prominent tones that are propagated into the space is counter to creating comfortable acoustical environments. The primary HVAC noise source is the fan and all fans produce tones at the blade pass frequency and harmonics. There is currently no accepted standard method of test for the tonality of fan noise. This paper proposes a standard method that can consistently be used to determine the tonality of fan noise, which can in turn be used to determine if when combined with other self gen noise generated by the air distribution system to determine the prominence of tones that will propagate to the occupied space.

2:45

2pNS6. Predicting tonal noise problems in HVAC systems. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Current indoor noise criteria for occupied spaces in buildings can be found in Table 1 of the *2019 ASHRAE Applications Handbook*. Table 1 of this handbook provides design guidelines for recommended NC, RC, dBA, and dBC ratings for a wide variety of interior spaces, but there is nothing in the table or the table notes that mentions the issue of audible tones in the background noise spectrum. There is a comment in the handbook text that essentially states that ideally, there should be no audible tones such as hum or whine. However, no guidance is provided as to how to predict or even measure (after the fact) if an audible tone is present. This presentation provides some guidance as to how a person might go about predicting the presence of an audible tone in the background noise of the HVAC system, including an example of a fully installed system.

3:05–3:25 Break

3:25

2pNS7. Optimizing building designs for mechanical equipment tones. Roman Wowk (Papadimos Group, 4302 Redwood Hwy., Ste. 100, San Rafael, CA 94947, rwowk@papadimosgroup.com)

Poor noise conditions in buildings often involve tones from mechanical equipment. However, noise criteria generally neglect tones due to (1) lack of agreement on what constitutes a tone; (2) lack of acoustical data from equipment and noise treatment manufacturers in ⅓-octave or better resolution; and (3) limited design tools and resources to act on detailed information even if it were available. Since building designs also cannot be tested and optimized prior to production like cars, appliances and other mass-produced products, problems can remain hidden until final occupancy. Therefore, design strategies usually involve a variety of conservative and/or safe "best practice" approaches that avoid certain space adjacencies or equipment types altogether, often at significant cost. This is unsurprising when considering the complexity and risks associated with many tonal noise control problems, often including sound-structure interaction. However, a path for innovation exists through wider use of frequency-response-function (FRF) measurements (or measurements with otherwise known inputs) and the ability to analyze, share and incorporate the findings into source-path-receiver designs. This presentation will propose steps to gather, standardize and distribute such information and use case studies to illustrate how this is an enormous opportunity for optimizing building designs for cost and performance.

3:45

2pNS8. Tonal noise case study: Locating the source of tones on HVAC equipment. Roderick Mackenzie (Soft Db, 250 Ave Dunbar, Ste. 203, Montreal, QC H3P 3E5, Canada, r.mackenzie@softdb.com)

The overall noise level measured from chiller units is controlled by primarily tonal noises from (1) the fans, (2) the compressors, and (3) the associated pipework. Three large chiller units were measured using AHRI 370 methodology to quantify the overall sound level spectra and A-weighted sound level from different chiller units in a range of environmental conditions. Whilst this method can identify a general area of tonal emission to within a couple of meters across a unit's surface that is emitting discrete tonal noises, the method is limited when there are multiple potential tonal sources. Sound intensity imagery was used on two units to quantify the overall sound power emanating from the source, but also to offer a more advanced method of localizing the source of discrete tones to aid in the application of mitigation treatments. Results presented include visually displaying and quantifying the location of noise emission from fan blade pass frequencies and associated harmonics, localization of problematic tones emanating from individual pipework junctions and valves, and the reductions in sound power emissions achieved from lagging compressors and pipework.

4:05

2pNS9. Ambient noise definitions in municipal codes and the application of acoustical engineering. Jordan L. Roberts (Veneklasen Assoc., 1416 ACTON St., Berkeley, CA 94702, jordanlennonroberts@gmail.com), Wayland Dong, and John Lo Verde (Veneklasen Assoc., Santa Monica, CA)

The duration of a noise measurement period that is required to establish an ambient or background noise level can vary drastically, from minutes to years. The descriptor of ambient noise levels can also vary whether it is a time-average sound level (L_{eq}), or an x-percentile-exceeded sound level (L_{50}, L_{90}) for a specified duration. None of the above methods are necessarily right or wrong, however some may be more or less appropriate depending on the sources of sounds that are sought to be regulated, the context of potential human impact, and the goals of the municipality. This session will present examples of ambient noise measurement procedures as informed by municipal code language, discuss the noise sources that are likely to exceed the respective ambient, and determine if the limits are appropriate, can be left to interpretation, or would require additional definition to properly represent conditions.

4:25

2pNS10. Evaluating tonal noise in the outdoor environment. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Most outdoor noise criteria are based on the overall A-weighted sound pressure level evaluated at one or more location(s) on the receiving property. The maximum allowable sound pressure level is usually based on the zoning or actual use of the source and receiving properties. The referenced levels are often the Leq and/or the Lmax measured over a defined time interval, although the L(n) percentile statistical levels are sometimes specified. It is a well-known fact in the acoustics community that broadband sound at a given level is less annoying than sound with strong tonal components at the same sound pressure level. As a result, many state and local governments have included tonal penalties in their environmental noise ordinances to help compensate for this tonal annoyance factor. This presentation identifies several state and prominent local governments that have imposed such a penalty and how the government has elected to assess the presence of a tone that would warrant such a penalty.

Contributed Paper

4:45

2pNS11. Conversion of tones to soothing noise. Xue Han, Ying Hu (Mech. Eng., Univ. of Hong Kong, Hong Kong, Hong Kong), Yumin Zhang (Foshan Univ., Foshan, Guangdong, China), and Lixi Huang (Dept. of Mech. Eng., Univ. of Hong Kong, University of Hong Kong, Hong Kong, Hong Kong, lixi@hku.hk)

When pure-tone sound signal of frequency f_s interacts with a boundary whose acoustic property varies at a frequency of f_m, part of the incident sound energy is scattered to the sum and difference frequencies, $f_s \pm n\ f_m$, where n is any integer but $n = 1$ dominates. When the modulation frequency f_m has a bandwidth, spanning from f_{m1} to f_{m2}, the resulting scattered sound carries the same bandwidth of f_{m2}-f_{m1}. In this manner, pure tone is partly converted to broadband noise and will be less annoying to human ears. This talk first introduces the initial experimental evidence of such frequency scattering using a realistic design (see reference paper with doi:10.1038/s42005-021-00721-1). Then the latest results of numerical study on the maximization of the energy scattering is presented. An attempt is also made to examine the question of what room mode or eigen frequencies will be when part of the room boundaries has a time-dependent wall properties. Note that the time-varying wall remains passive in the sense that the property variation is independent of the incident sound and there is no energy input required.

5:00–5:20
Panel Discussion

Session 2pPA

Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration: Interaction of Electromagnetic Waves with Acoustic Waves

Chu Ma, Cochair
Electrical and Computer Engineering, University of Wisconsin-Madison, 1415 Engineering Drive, Room 3436, Madison, WI 53706

Parag V. Chitnis, Cochair
Bioengineering, George Mason University, 4400 University Dr., 1J7, Fairfax, VA 22030

Invited Papers

1:00

2pPA1. Interaction of acoustic and electromagnetic waves for thermoacoustic imaging and vibration sensing. Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Rm. 3436, Madison, WI 53706, chu.ma@wisc.edu), Audrey L. Evans, and Dajun Zhang (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

Acoustic wave and electromagnetic wave can interact and lead to rich physical phenomenon to be used for applications such as medical diagnosis, non-destructive material characterization, and remote sensing. In this talk, I will present our research studies on two ways acoustic and electromagnetic waves interact with each other: (1) When pulsed microwave is absorbed by tissue, thermoacoustic waves are generated through thermal expansion. We explored the sensing and processing of thermoacoustic waves generated by a pulsed microwave ablation antenna as a potential way for monitoring the ablation process in real time. (2) Electromagnetic waves such as Wi-Fi signals and light can be modulated by acoustic/mechanical vibrations in the environment. We developed non-contact vibration sensing systems based on such modulations, and also developed a metamaterial-based vibration amplification device that can enhance the vibration magnitude sensitivity over a broad range of vibration frequencies.

1:30

2pPA2. Deep-tissue photoacoustic imaging of photoswitchable transgenic mouse models. Junjie Yao (Duke Univ., 100 Sci. Dr., Hudson Hall Annex 261, Durham, NC 27708, junjie.yao@duke.edu)

Compared with visible light, near-infrared (NIR) light (700–1300 nm) is capable of deep-tissue penetration because of low tissue attenuation, and thus holds great promise for deep-tissue optical manipulation and imaging. In this work, we take advantages of the NIR Rhodopseudomonas palustris bacterial phytochrome photoreceptors (BphP1) and generate a loxP-BphP1 photoswitchable transgenic mouse model that enables Cre-dependent temporal and spatial targeting of BphP1 expression at the whole-body scale *in vivo*. This BphP1 phytochrome incorporates biliverdin chromophore and reversibly switches between the ground state and activated state, which enabled the light-controlled manipulation of gene expression and differential detection in photoacoustic tomography (PAT). We first validated the optogenetic performance of BphP1 that binds its engineered protein partner QPAS1 to trigger gene transcription in primary cells and living mice, and demonstrated its superior capability of deep-tissue optogenetics over the existing visible-light proteins. Then, by taking advantage of PAT's deep-tissue non-invasive imaging ability and the BphP1's photoswitching properties, we can suppress the non-switching signals from background blood and improve the molecular detection sensitivity by three orders of magnitude. This study shows the BphP1-encoded photoswitchable transgenic mouse model can be used for both NIR optogenetic manipulation and deep-tissue PAT.

2:00

2pPA3. Quartz tuning fork based photoacoustic spectroscopy and sensing. Andrea Zifarelli (PolySense Lab - Dipartimento di Fisica, Univ. and Politecnico of Bari, Via Giovanni Amendola, 173, Bari 70100, Italy, andrea.zifarelli@uniba.it), Pietro Patimisco, Angelo Sampaolo, Marilena Giglio, Michele Di Gioia (PolySense Lab - Dipartimento di Fisica, Univ. and Politecnico of Bari, Bari, Italy), Lei Dong, Hongpeng Wu (State Key Lab. of Quantum Optics and Quantum Optics Devices, Inst. of Laser Spectroscopy, Shanxi Univ., Taiyuan, China), and Vincenzo Spagnolo (PolySense Lab - Dipartimento di Fisica, Univ. and Politecnico of Bari, Bari, Italy)

Quartz crystal tuning forks (QTFs) resonators are key components for timing and frequency measurements, due to their high stability, high-quality factors and low power consumptions. Thanks to their piezoelectric properties, QTFs are employed as sensitive element in many fields and systems such as atomic force microscopy, near-field and microwave microscopy, and mass/viscosity sensor. Since 2002, QTFs are also widely used as a sharply resonant acoustic transducer to detect weak photoacoustic excitation for Quartz-Enhanced Photoacoustic Spectroscopy (QEPAS). Among most sensitive optical techniques, QEPAS was demonstrated as the leading-edge technology for addressing these application requirements, providing also modularity, ruggedness, portability and allowing the use of extremely small volumes. QEPAS technique does not require an optical detector, it is wavelength independent, it is immune to environmental noise

and can operate in a wide range of temperature and pressure. These factors, together with its proven reliability and ruggedness, represent the main distinct advantages with respect to other laser-based techniques for environmental monitoring and *in situ* detection. Starting from the basic physical principles governing the QTF physics, I will review the main results achieved by exploiting custom QTFs for QEPAS sensing and as photodetector in LITES setup, with a main focus on real-world applications.

Contributed Papers

2:30

2pPA4. *In situ* monitoring of electromigration in a single nano wire by picosecond ultrasonics. Akira Nagakubo (Graduate School of Eng., Osaka Univ., M1-523, 2-1, Yamada-Oka, Suita, Osaka 565-0871, Japan, nagakubo@prec.eng.osaka-u.ac.jp), Eriko Asanuma, and Hirotsugu Ogi (Graduate School of Eng., Osaka Univ., Suita, Japan)

Recent nano-scale fabrication has realized the progress of integrated circuits, where current density increases with the downscaling of each electronic component, leading to failure due to electromigration. Electromigration is the transport of materials via momentum exchange from high-density conducting electrons to metal atoms. Defects of atoms result in the disconnection of the circuits and the accumulation of atoms causes the short circuit. Therefore, electromigration due to highly dense current flow has been studied for long years. However, it is difficult to non-destructively evaluate the structural and physical property changes due to the electromigration in an *in-situ* and microscopic measurement. In this study, we develop a real-time monitoring method of structural and mechanical property changes of a single nanowire using picosecond ultrasonics. Picosecond ultrasonics is ultrafast pump-probe laser ultrasonics using femtosecond pulse laser to excite and detect sub-THz ultrasound. We can focus the laser light within 1-micrometer diameter, enabling us to evaluate the mechanical property changes in a single nanowire from the ultrasound propagation. By applying direct current and observing ultrasound, we discuss the relationship among mechanical, electrical, and structural changes due to electromigration.

2:45

2pPA5. All-optical, non-contact, local measurement of both Young's modulus and Poisson's ratio in metals using a combination of Rayleigh and leaky surface acoustic waves. Ivan Pelivanov (Bioengineering, Univ. of Washington, 616 NE Northlake Pl, Benjamin Hall Bld, Rm. 363, Seattle, WA 98105, ivanp3@uw.edu), Ryan Canfield (Bioengineering, Univ. of Washington, Seattle, WA), Aleksandra Ziaja-Sujdak (Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), John Pitre, Matthew O'Donnell (Bioengineering, Univ. of Washington, Seattle, WA), and Lukasz Ambrozinski (Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland)

In non-destructive evaluation (NDE), measuring ultrasound (US) longitudinal and shear wave speeds is the main method to determine two independent mechanical moduli (Young's modulus and Poisson's ratio). Most US techniques use time-of-flight measurements. However, when the local sample thickness is unknown or the sample geometry is complex, bulk-wave propagation speeds cannot be accurately defined. Here we show that a properly shaped beam of nanosecond laser pulses can be used to efficiently excite (without material ablation) two types of surface acoustic waves. In addition to a conventional surface, or Rayleigh, wave, a leaky surface wave (LSAW) can be launched in the near field of the excitation region. We present the theoretical background, numerical simulations, and experimental results clearly showing that both elastic constants can be reconstructed locally by tracking Rayleigh and LSAW waves. Spatially resolved elastic properties can be obtained using local values of wave speeds obtained by sample scanning. Non-contact optical detection of propagating waves at the sample surface, in which a fiber-optic Sagnac interferometer was used, is a key piece of the method. It does not require acoustic coupling and allows remote measurements in the near field of the laser source with micron-scale resolution.

3:00

2pPA6. Quartz enhanced photoacoustic spectroscopy gas sensing exploiting beat frequency approach. Giansergio Menduni (PolySense Lab - Dipartimento Interateneo di Fisica, Univ. and Politecnico of Bari, Via Giovanni Amendola, 173, Bari 70125, Italy, giansergio.menduni@poliba.it), Biao Li (State Key Lab. of Quantum Optics and Quantum Optics Devices, Inst. of Laser Spectroscopy, Shanxi Univ., Taiyuan, China), Andrea Zifarelli, Marilena Giglio, Angelo Sampaolo, Pietro Patimisco (PolySense Lab - Dipartimento Interateneo di Fisica, Univ. and Politecnico of Bari, Bari, Italy), Tingting Wei, Hongpeng Wu, Lei Dong (State Key Lab. of Quantum Optics and Quantum Optics Devices, Inst. of Laser Spectroscopy, Shanxi Univ., Taiyuan, China), and Vincenzo Spagnolo (PolySense Lab - Dipartimento Interateneo di Fisica, Univ. and Politecnico of Bari, Bari, Italy)

Quartz-enhanced photoacoustic spectroscopy (QEPAS) is a highly sensitive optical technique, suitable for real-time and *in situ* trace gas detection. In QEPAS, Quartz tuning forks (QTF) are employed as piezoelectric transducers of sound waves induced by gas non-radiative energy relaxation as consequence of infrared modulated light absorption. The generated electric signal depends on the gas concentration. An accurate and reliable QEPAS measurement requires the QTF characterization, in terms of resonance frequency and quality factor. Furthermore, tens of seconds are required to complete the tuning range scan of the laser employed to detect the selected gas. Beat frequency QEPAS (BF-QEPAS) is an alternative approach to standard QEPAS. In BF-QEPAS, a fast scan of the laser tuning range is employed to generate an acoustic pulse. Gas concentration, QTF resonance frequency, and quality factor can be measured acquiring and analyzing the transient response of the QTF to the acoustic pulse. In this work, a custom T-shaped QTF was employed to detect nitrogen monoxide (NO), targeting its absorption feature at 1900.07 cm^{-1} with an interband cascade laser. A minimum detection limit as low as 166 ppb of NO, and a highly accurate measurement of the QTF resonance frequency and quality factor were demonstrated using BF-QEPAS.

3:15–3:30 Break

Invited Papers

3:30

2pPA7. Transcranial acoustoelectric imaging: Towards noninvasive mapping of current densities in the human brain. Russell S. Witte (Medical Imaging, Univ. of Arizona, 1230 N Cherry Ave., BSRL, #248, Tucson, AZ 85721, rwitte@arizona.edu), Margaret Allard (Optical Sci., Univ. of Arizona, Tucson, AZ), Teodoro Trujillo, Alex Alvarez (Biomedical Eng., Univ. of Arizona, Tucson, AZ), Chet Preston (Medical Imaging, Univ. of Arizona, Tucson, AZ), Jinbum Kang, and Matthew O'Donnell (Biomedical Eng., Univ. of Washington, Seattle, WA)

Acoustoelectric Imaging (AEI) is a disruptive technology that exploits an ultrasound (US) beam to transiently interact with physiologic or artificial currents, producing a radiofrequency signature detected by one or more surface electrodes. By rapidly sweeping the US beam and simultaneously detecting the acoustoelectric modulations, 4D current density images are generated at high spatial resolution determined by the ultrasound beam focus. The principle has been used for *in vivo* mapping of currents in the swine heart during the cardiac activation wave. When applied to the brain, transcranial acoustoelectric imaging (tABI) overcomes limitations with electroencephalography (EEG), which suffers from poor spatial resolution and inaccuracies due to blurring of electrical signals as they pass through the brain and skull, and, unlike fMRI and PET that measure slow metabolic or hemodynamic signals, tABI directly maps fast time-varying current within a defined brain volume at the mm and ms scales. This invited presentation will describe the underlying physics and mathematics of tABI, recent progress and challenges using numerical simulations and bench-top models, and its potential impact as a cutting-edge noninvasive modality for fast and accurate electrical brain mapping in humans.

4:00

2pPA8. Radiation-induced acoustic imaging. Liangzhong S. Xiang (Radiology + BME, UC Irvine, 825 Health Sci. Rd., Irvine, CA 92697-5000, liangzhx@hs.uci.edu)

We explore new ways to generate ultrasound for medical diagnosis and treatment monitoring. Specifically, we use varies radiations (X-ray, proton, and electrical field) to treat diseases, simultaneously it will produce ultrasound waves for image-guided interventions. This talk will cover three primary research areas: (1) x-ray-induced acoustic computed tomography (XACT) for precision radiotherapy and radiology; (2) proton-induced acoustic imaging for proton therapy guidance; and (3) electroacoustic tomography (EAT) for irreversible electroporation (IRE) monitoring. Prospects and challenges for the clinical implementation of these techniques will be discussed. The successful development of these technologies will expand the current clinical paradigm towards precision medicine.

4:30

2pPA9. Thermoacoustic signals generated from pulsed microwave ablation in liver tissue. Audrey L. Evans (Elec. and Comput. Eng., Univ. of Wisconsin-Madison College of Eng., 1415 Eng. Dr. RM 3415, Madison, WI 53706, alevans3@wisc.edu), Susan C. Hagness, and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

Microwave-induced thermoacoustic signals are generated when pulsed microwave energy is absorbed by tissue. Thermoacoustic signals contain information about the electromagnetic, thermal, and acoustic material properties of the tissue. Thus, we propose using microwave-induced thermoacoustic signals to monitor microwave ablation, a thermal therapy used to heat and kill malignant tissue, in real-time. Conventional microwave-induced thermoacoustic signals are excited using an external waveguide. Our approach excites thermoacoustic signals via pulsed microwave energy delivered by the already present interstitial microwave ablation antenna. In this talk, we will present the experimental investigation of the evolution of microwave-induced thermoacoustic signals generated during microwave ablation in bovine liver tissue. Pulsed microwave energy was delivered via a narrow-diameter coaxial ablation antenna to simultaneously ablate tissue and excite thermoacoustic signals. Trials were conducted in room-temperature fresh liver tissue as well as room-temperature boiled liver tissue to investigate the influence of ablative temperature rise and tissue coagulation on thermoacoustic signals. Our experimental results show thermoacoustic signal characteristics change throughout the course of microwave ablation that could potentially be used for monitoring the microwave ablation process.

Contributed Papers

5:00

2pPA10. Strongly focused nonlinear surface acoustic wave beams in isotropic solids. Brittany A. McCollom (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78713, bmccollom@utexas.edu), John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Edoardo Baldini (Dept. of Phys., 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)

A promising direction in the tailored control of quantum materials is the modification of their structure-function relationship via engineered laser-driven shock waves. In order to inform how the laser excitation should be tailored to optimize the effect of the shock waves, a model is desired for strongly focused nonlinear surface waves in crystals. A model was developed previously in the paraxial approximation for weakly diffracting nonlinear surface wave beams in isotropic solids [Shull *et al*., JASA **97**, 2126 (1995)]. Harmonic generation and shock formation were investigated numerically in that work for sources with uniform amplitude distributions, with analytical solutions for the fundamental and second harmonic obtained for weakly focused Gaussian beams. Here, numerical results are reported for shock formation in focused nonlinear surface waves obtained using a model unrestricted by the paraxial approximation in order to accommodate strong focusing in isotropic solids. Removal of the restriction to isotropic solids is work in progress based on representation of the wave field in terms of its angular spectrum and a model for planar nonlinear surface waves in crystals [Hamilton *et al*., JASA **105**, 639 (1999)]. [Work supported by the IR&D Program at Applied Research Laboratories.]

5:15

2pPA11. Virtual imaging trials to investigate impact of skin color on three-dimensional optoacoustic tomography of the breast. Seonyeong Park (Bioeng., Univ. of Illinois Urbana-Champaign, 1406 West Green St., Urbana, IL 61801, sp33@illinois.edu), Umberto Villa (Oden Inst. for Computational Eng. & Sci., The Univ. of Texas at Austin, Austin, TX), Alexander A. Oraevsky (TomoWave Labs., Houston, TX), and Mark Anastasio (Bioengineering, Univ. of Illinois Urbana-Champaign, Urbana, IL)

Optoacoustic tomography (OAT), also known as photoacoustic computed tomography, is being actively developed for breast imaging applications. The endogenous optical contrast in OAT images is associated with oxygen saturation and concentrations of chromophores, such as hemoglobin, melanin, fat, and water, within the tissue. In OAT breast imaging, near-infrared light propagates through the skin, where the optical energy is absorbed primarily by melanin. The photoacoustic effect results in the generation of a pressure wavefield, and the propagated pressure wavefield is measured by ultrasonic transducers located on a measurement aperture surrounding the breast. Thus, the melanin concentration influences lesion contrast in OAT images. However, the extent to which skin color affects lesion detectability in OAT breast imaging remains unexplored. To address this, we generate realistic optoacoustic 3D numerical breast phantoms containing a lesion at different locations (three depths and two polar angles) with five skin colors and virtually acquire optoacoustic data employing them. To assess the skin color impact on lesion detectability, we quantify numerical observer performance for a signal-known-exactly and background-known-exactly detection task. The results confirm that the signal-to-noise ratio of the test statistic is degraded in darker skin, and the extent depends on lesion locations and the light delivery system design.

TUESDAY AFTERNOON, 9 MAY 2023 SALON I, 1:00 P.M. TO 5:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Best Student Paper Award: Psychological and Physiological Acoustics (Poster Session)

Kelly L. Whiteford, Chair

Psychology, University of Minnesota, N218 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55414

All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd numbered abstracts will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even numbered abstracts will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

2pPP1. Validation of a clinical test of auditory emotion recognition. Shae D. Morgan (Otolaryngol., Univ. of Louisville, Louisville, KY), Bailey LaPaugh (Otolaryngol., Univ. of Louisville, 500 S. Preston St., 4th Fl., Louisville, KY, bailey.lapaugh@louisville.edu), Emily Cullum, Camryn McClure, and Anna Grace O'Donell (Otolaryngol., Univ. of Louisville, Louisville, KY)

Currently, there are no accessible clinical tests that can be used to measure auditory emotion recognition. Experimental emotion recognition tasks usually involve many stimuli and take a lot of time to complete, making them unfeasible for use in a clinical setting. Prior research from our laboratory has measured the impact of various test factors on auditory emotion recognition, including the number of emotion categories, the number of stimuli per emotion category, the number of talkers, and using sentence versus word materials. In this study, we present a preliminary validation of a proposed audiotry emotion recognition test. Based on the results from our laboratory, the test consists of forty sentences (1 sentence $\times$ 10 talkers $\times$ 4 emotions) in order to maintain an appropriate test duration for a clinical setting. Listeners with normal hearing will complete the testing in quiet. Results will be compared with expectations informed by prior research. Performance and duration of the task will be measured, and standard deviation of scores among all participants will be assessed. Implications will be discussed, including the percentage correct of emotion recognition by normal hearing listeners.

2pPP2. Usability and perceived benefit of hearing assistive features in Apple AirPods Pro. Emily A. Hammond (Commun. Sci. & Disord., Western Washington Univ., 516 High St., Bellingham, WA 98225, hammone2@wwu.edu) and Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., Bellingham, WA)

Individuals with hearing loss face high personal costs for hearing aids, making over-the-counter hearing aids, or headphones featuring custom audio control, alternative options. Apple Inc.'s AirPods Pro contain a variety of headphone accommodations designed to help people hear better. Recent research has evaluated electroacoustic properties of AirPods Pro demonstrating good ability to fit prescriptive targets for different hearing losses. Here, we examine the usability and perceived benefit from engaging custom audiogram-driven features in AirPods Pro. Listeners with self-assigned hearing profiles ranging from normal hearing to a moderate degree of hearing loss take a hearing screener (Mimi Hearing Test). AzBio sentences are presented in quiet and in noise while enabling and disabling the custom features in Apple's Accommodation settings. Listeners then complete a questionnaire targeting usability of the features and perceived benefit. Overall, listeners found the features easy to use and to be at least a little helpful, regardless of hearing status. Listeners with suspected or known hearing impairment reported slightly greater benefit than normal hearing listeners when listening in noise with the custom audio features. Headphone Accommodations in AirPods Pro may be an affordable option for an improved listening experience. [Work supported by Western Washington University.]

2pPP3. Addressing auditory processing challenges and accessibility in live music settings. Kai White (Computational and Cognit. Musicology Lab, Georgia Inst. of Technol., North Ave., Atlanta, GA 30332, mwhite76@gatech.edu)

With the social media boom connecting people worldwide and easing the sharing of experiences, the everyday lives and struggles of people who are considered neurodivergent as well as those with invisible disabilities suddenly became widely known and the accessibility of even casual excursions such as grocery shopping and movie viewings came into question. As understanding of sensory processing disorders increases and disability advocacy moves to the forefront of conversations about public spaces, it's not surprising that live music and the entertainment industry would become the next target of scrutiny. While the creation of sensory-friendly concerts is beneficial and highly valuable as a stepping stone to a greater and more widespread understanding of accessibility for those whose needs are often ignored by society, the limited scope and availability of these events largely excludes a multitude of music enthusiasts. This study aims to highlight the experiences and desires of concertgoers who have auditory sensitivities while also providing potential solutions to these barriers to entry.

2pPP4. Loudness in rooms and the sound strength parameter: Linking perception and physical acoustics. Ann Holmes (Psychol. & Brain Sci., Univ. of Louisville, 2082 Douglass Blvd., Apt. 5, Louisville, KY 40205, ann.holmes@louisville.edu), Matthew Neal, and Pavel Zahorik (Otolaryngol. and Comm. Disord., Univ. of Louisville and Heuser Hearing Inst., Louisville, KY)

Loudness as a perceptual phenomenon has been extensively studied within psychoacoustics. In room acoustics, however, the psychoacoustical factors influencing loudness have yet to be incorporated into standard parameters such as sound strength (G). G is the decibel ratio of the integrated energy of a room impulse response (RIR) compared to the same source located 10m away in a free field. The purpose of this study was to determine how the physical RIR properties measured by G relate to perceived loudness in the same room. Listeners were asked to match pink noise presented anechoically to the perceived loudness of the same noise in a simulated room. Listeners could continuously adjust the level of the anechoic reference. Room simulations were rendered over a 36-channel loudspeaker array in an anechoic chamber. Loudness matches were performed for different room types, source-receiver distances, and source powers, which produced received sound levels between 35 and 85 dBA. Results show a relatively linear relationship between loudness and G. However, it is clear that G underpredicts loudness ratings at higher sound levels. This demonstrates a more complex relationship between G and loudness. Effects of room type and source-receiver distance appear to introduce additional complexity not represented by G.

2pPP5. Measuring the timeline of retroactive sentence repair. Steven P. Gianakas (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, giana009@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Individuals with hearing loss are more likely to miss a word during speech perception, but can use context to retroactively repair the missing word. When sentences need to be repaired, listening effort remains elevated even after the entire sentence is heard; suggesting the repair process continues beyond the stimulus presentation. Importantly, if a listener is stuck processing a previous sentence, they may miss an upcoming utterance, making everyday conversation difficult. By measuring the amount of time that is needed to repair a missing word, we will better understand if processing would interfere with the perception of a subsequent sentence. Individuals with normal hearing and hearing loss heard sentences that did or did not require retroactive repair while performing a secondary case-judgement task. Listeners decided whether a visually presented letter was upper- or lower-case either during the sentence, 0.5 s after, or 2 s after the sentence. Preliminary results indicate that reaction times were faster for intact sentences and decrease (improve) as more time has passed (as the missing word has been resolved) for both sentence types. Our results suggest that the timeline of the repair process can be understood using behavioral measures that complement intelligibility scores.

2pPP6. Cognitive performance on a neuropsychological battery with accommodations for hearing loss by listeners with cochlear implants. Rebecca S. Kelly (Hearing and Speech Sci., Univ. of Maryland, College Park, 9741 Summer Park Court, Columbia, MD 21046, rhiggin2@umd.edu), Miranda Cleary (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Anjeli Inscore, Moira Dux (Neuropsychology, Departmental of Veterans Affairs, Baltimore, MD), Nicole Nguyen (Hearing and Speech Sci., Univ. of Maryland, College Park, College Park, MD), Jacob Blumenthal (Geriatrics, Dept. of Veterans Affairs, Baltimore, MD), Anna Tinnemore, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

A relationship between hearing loss and cognitive decline has emerged across multiple studies. Hearing aids and cochlear implants (CIs) may slow down the progression of cognitive decline. Cognitive factors may also influence performance with a CI. However, cognitive tests often assume perfect sensory perception, underestimating performance in individuals with hearing loss. The purpose of this study is to determine how CI listeners perform on a cognitive battery when accommodations for hearing loss are provided. Twenty-eight older adults with CIs completed a neuropsychological test battery created by an interdisciplinary team which assessed six cognitive domains and included accommodations for hearing loss. Participants also completed sentence recognition tests in quiet and in noise, and several subjective questionnaires. Preliminary data revealed that the performance of CI listeners with accommodations on tests of processing speed ($p < 0.001$) and visuospatial ability ($p = 0.002$) was significantly poorer compared to the normative sample. Performance on measures assessing memory, executive function, attention, and language was not significantly different. In conclusion, when neuropsychological test instructions and stimuli are adapted for hearing loss and functional speech performance is accounted for, individuals with CIs perform similarly to normal-hearing peers but not in all domains, suggesting a more accurate assessment of cognitive functioning.

2pPP7. Broad phonetic feature classifier for real-time hearing aid processing. Prasad Madhava Kamath (Elec. and Comput. Eng., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, pkamath@ucsd.edu), Martin Hunt (Nadi Inc., West Lafeyette, IN), Bhaskar D Rao (Elec. and Comput. Eng., UC San Diego, La Jolla, CA), and Harinath Garudadri (Qualcomm Inst., UC San Diego, San Diego, CA)

Current Hearing Aid (HA) devices compensate for hearing loss by amplifying sounds based on instantaneous energy estimates in specific frequency bands. The dynamics of speech are incorporated using attack and release times—a crude approximation of the rich temporal dynamics in natural speech. Recent advances in silicon enables one to consider more complex sound processing approaches with minimal impact on the HA battery life. We envisage a broad phonetic feature classifier, followed by amplification strategies optimized for specific temporal dynamics across each segment of natural speech. In this work we present a broad phonetic classifier for 'vocalic', 'non-vocalic' and 'mixed' speech segments. The classifier is based on (i) Interpretable models- unsupervised learning methods such as Gaussian Mixture Models followed by supervised learning based Radial Basis Function networks; and (ii) Deep Neural Networks with supervised learning. The frontend feature extractor consists of MFCC, delta MFCC features, spectral flatness measure and segmental energy. The training and test sets are generated using speech files and the corresponding phone labels from the TIMIT dataset. We present our findings with respect to the classifier's performance and deployability for real time HA processing.

2pPP8. The influence of musical abilities on the processing of second language prosody: An eye-tracking study. Nelleke Jansen (Appl. Linguist., Univ. of Groningen, Oude Kijk in 't Jatstraat 26, Groningen 9712 EK, the Netherlands, n.g.jansen@rug.nl), Hanneke Loerts (Appl. Linguist., Univ. of Groningen, Groningen, the Netherlands), Eleanor Harding, Deniz Baskent (Dept. of Otorhinolaryngology/Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, the Netherlands), and Wander Lowie (Appl. Linguist., Univ. of Groningen, Groningen, the Netherlands)

The perception of speech prosody in a second language (L2) remains challenging even for proficient L2 users. Eye-tracking evidence indicates that Dutch listeners show difficulty in the processing of pitch accents that signal a contrast (i.e., contrastive focus) in English, whereas native English listeners use this cue in perception to anticipate upcoming information [Ge *et al.*, *Appl. Psycholinguist.* **42**, 1057–1088 (2021)]. Prosody perception abilities in foreign languages have been associated with individual differences in musical abilities [Jansen *et al.*, *Speech Prosody*, 713–717 (2022)]. We investigated whether musical abilities influenced the processing of prosodic cues by 45 Dutch adult L2 English users, using a visual-world eye-tracking paradigm. Participants listened to sentences with pitch accent cues to contrastive focus on different words, while viewing pictures showing objects and characters mentioned. We investigated to what extent participants showed anticipatory fixations on the image reflecting the alternative of the contrast, after hearing the accented word. We measured participants' music perception abilities with a standardised test and analysed its influence on anticipatory fixations. Initial analyses indicate that individuals with stronger musical abilities show more anticipation. This suggests that having stronger perceptual resources underlying both music and speech processing [Patel, *Front. Psychol.* **2**, 142 (2011)] may benefit L2 prosody perception.

2pPP9. Acoustic feedback control in hearing aids with frequency warping. Alice Sokolova (Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, aasokolo@ucsd.edu), Varsha H. Rallapalli (Commun. Sci. & Disord., Northwestern Univ., Evanston, IL), Anusha Yellamsetty (Dept. of Audiol., San Jose State Univ., San Jose, CA), Martin Hunt (Nadi Inc., West Lafeyette, IN), Baris Aksanli (San Diego State Univ., San Diego, CA), Fredric Harris, and Harinath Garudadri (Univ. of California San Diego, San Diego, CA)

Acoustic Feedback Control continues to be a challenging problem due to the emerging form factors in advanced hearing aids and hearables, especially for severe and profound losses. In order to mitigate the effects of feedback in hearing aids, we present a novel use of frequency warping called "Freping," for manipulating signals in the frequency domain. Freping is a non-linear transformation which breaks the similarity between the input and acoustic feedback signals of a hearing aid to prevent feedback. In this work, we implemented Freping in realtime for feedback management, as well as novel techniques to preserve the naturalness of sound. Our prior work has shown that Freping improves Added Stable Gain (ASG) in the presence of feedback while maintaining the same Hearing Aid Speech Quality Index (HASQI). We present new results for feedback management with Freping by performing "on-ear" measurements on an acoustic KEMAR manikin using standard hearing aid verification systems. We conducted feedback tests to measure the ASG with various configurations of Freping for various hearing loss levels using the NAL-NL2 prescriptions on a realtime open-source speech processing system for hearing aids. We also compare Freping feedback performance with that of commercial hearing aids.

2pPP10. Relating monaural and binaural envelope processing abilities to spatial release from masking. Morgan Barkhouse (Speech-Lang. Pathol. and Audiol., Towson Univ., Speech-Lang. Pathol. and Audiol., Towson University, Towson, MD 21252, mbarkh1@students.towson.edu), Sadie O'Neill, Chhayakanta Patro, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

The difference in speech identification thresholds between colocated condition (when the target and the maskers are placed at the same location) and spatially separated condition (when target and the maskers are symmetrically separated) is quantified as Spatial Release from Masking (SRM). SRM is thought to be a combination of monaural and binaural advantages arising from spatially separating the target from the maskers. Monaural contributions are from the head shadow effect whereas the binaural contributions are from processing the interaural differences between the two ears. However, the exact contributions of the monaural and binaural advantage to SRM is unknown. Here, we present data on monaural and binaural envelope processing abilities and their relationship to SRM on a large cohort of young normal hearing listeners. Monaural envelope processing ability was measured using the envelope regularity discrimination task (Moore *et al.*, 2019). Sensitivity to ITD-envelopes was used to quantify binaural envelope processing ability. SRM was measured using coordinate response measure sentences. The relationship between SRM, envelope regularity index, and ITD thresholds along with the monaural and binaural envelope processing abilities to SRM will be discussed.

2pPP11. Interplay between attention, working memory, and cognition to speech understanding in noise. 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, and Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Speech understanding in noisy environments depends not only on hearing acuity but also on a host of other cognitive skills including attention, working memory, and executive function that support listeners' ability to segregate, track and attend to a "target" signal while tuning out other unwanted signals. Identifying the cognitive aspects that affect listening outcomes in complex listening environment is critical for interpreting individual variability and understanding the challenges listeners with different cognitive profiles might face during listening in such environments. The latter is important for characterizing the listening deficits in older adults who, in addition to impaired peripheral auditory processing, also tend to exhibit a decline in various cognitive abilities that might affect listening. Here, we present data from a large cohort of younger listeners on various working memory (Reading Span Test), attention (auditory and visual single and dual task), processing (trail making task), executive control (flanker task), inhibition (Stroop task), and speech in noise tests (spatial release from masking using coordinate Response Measure sentences). The relationship between these various attention, working memory, and cognitive components to speech understanding in complex listening environments will be discussed.

2pPP12. Assessing the challenge of accented speech using effort discounting. Grace E. Teuscher (Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, grace.teuscher@gmail.com), Drew J. McLaughlin (Basque Ctr. on Cognition, Brain, and Lang., St. Louis, MO), and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

The subjective ease of understanding accents that differ from a listener's has typically been assessed using self-reports. This approach, however, relies on metacognitive judgments that are difficult to interpret and may not converge with objective measures of effort. To address this challenge, this study utilizes effort discounting, a paradigm borrowed from behavioral economics. In the experiment, participants are familiarized with one L1 English speaker and three L2-accented speakers. During the task, participants choose between listening to the L1-accented speaker for a smaller monetary reward or to one of the L2-accented speakers for a larger reward. By varying the reward offered for the easier option based on previous choices, the subjective value of the effort expended for each L2 speaker can be determined. Data collection is ongoing. We expect participants will remain willing to listen to highly intelligible L2-accented speakers. However, as speakers become less intelligible, participants will be less willing to expend the required additional effort and will choose the easier speaker. We also predict that participants who rate L2-accented speakers lower on an affect and attitudes questionnaire will be more likely to discount their reward to avoid the effort required by the L2-accented speech.

2pPP13. Decoding speech envelopes from electroencephalographic recordings: A comparison of regularized linear regression and long short-term memory deep neural network. Zhe-chen Guo (Dept. of Linguist., Univ. of Texas at Austin, 307 E 31st St. Apt. 105, Austin, TX 78705, y9024131@gmail.com), Kevin Pangottil (Dept. of Comput. Sci., Univ. of Texas at Austin, Austin, TX), Bharath Chandrasekaran (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA), and Fernando Llanos (Dept. of Linguist, Univ. of Texas at Austin, Austin, TX)

The speech envelope provides enough acoustic information to accurately recognize consonants and vowels (Shannon *et al*., 1995). The neural representation of speech envelopes is often assessed by reconstructing the envelopes from neural oscillations in the electroencephalogram (EEG) using linear decoders. One such approach is the multivariate temporal response function (mTRF), which achieves envelope reconstruction through regularized linear regression. Here, we compared the envelope reconstructions achieved by the mTRF and a non-linear alternative derived from a long-short term memory (LSTM) deep network. EEGs were collected from 15 native English speakers listening to an English audiobook (Reetzke *et al*., 2021). We trained a different decoder for each consonant and vowel in each listener. Reconstruction accuracy was measured as the Pearson coefficient (r) between observed and reconstructed envelopes. Preliminary results for the reconstruction of all vowels revealed that speech envelopes were more accurately reconstructed by the LSTM decoder (r: M = 0.247, SEM = 0.0024) than the mTRF (r: M = 0.074, SEM = 0.0025). Reconstruction accuracy was equally high and less variable across subjects for the LSTM approach. Additionally, high vowels showed lower decoding performance potentially due to their lower amplitude. These findings demonstrate the potential of non-linear approaches to investigating the neural representation of speech envelope cues.

2pPP14. Associations of acoustic parameters and gender dysphoria in transmasculine individuals. Megan M. Lee (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT) and Brett R. Myers (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT, brett.myers@hsc.utah.edu)

Gender dysphoria refers to a feeling of discomfort due to a mismatch between a person's biological sex and gender identity. Voice is a secondary sex marker, and it may contribute to feelings of unease or dissatisfaction if the voice does not match a person's gender identity. The Utrecht Gender Dysphoria Scale—Gender Spectrum (UGDS-GS) was designed to measure a person's degree of gender dysphoria. Transmasculine individuals were assigned female at birth and identify as male or masculine non-binary, and acoustic studies focusing on transmasculine voices are notably sparse. In a study including 20 transmasculine individuals, we compared UGDS-GS results to the following acoustic parameters: vocal tract length (VTL) estimates, fundamental frequency (*f0*), standard deviation of *f0*, and cepstral peak prominence smoothed (CPPS). We found a significant positive correlation between each acoustic measure and gender dysphoria. We also found robust associations between each of these measures and a voice-related quality of life measure. These findings indicate that the sound of one's voice may be a strong predictor of gender dysphoria.

2pPP15. The influence of non-canonical vowels on phonetic parsing. Yuka Tashiro (Linguist, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 504, Bloomington, IN 47405, yutash@iu.edu)

Speech segments are highly context sensitive due to coarticulation. This paper investigates phonetic context effects and how listeners *parse* the acoustic signal (Fowler, 1984). The previous literature reported that listeners could parse adjacent segments in a dissimilative manner (e.g., Mann, 1980), or in an assimilative manner (e.g., Fujimura *et al*., 1978). More recently, Rysling *et al*. (2019) reported that spectrally ambiguous or discontinuous transition can lead *dissimilative parsing* than *assimilative parsing*. This paper examines how non-canonical vowels affect the perception of adjacent consonant. When listeners hear non-canonical vowels, they may parse the incoming phonetic signal matching canonical exemplars, and assign any deviance in the spectral frequencies to the adjacent consonant. Hence, listeners may do assimilative parsing even with or without appropriate vowel transitions. American English listeners identified synthetic CV and VC syllables with an ambiguous fricative noise between [s] and [ʃ], and vowels including a good exemplar and poor exemplars of [i] and [u], each vowel with or without vowel transitions. Analyses are currently underway, but will illuminate the cause of assimilative parsing with respect to exemplar matching in the exemplar-based model (e.g., Johnson, 1997; Pierrehumbert, 2001).

2pPP16. A phonetic contrast function to improve sound processing in hearing aids. Prasad Madhava Kamath (Elec. and Comput. Eng., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, pkamath@ucsd.edu), Alice Sokolova (Elec. and Comput. Eng., UC San Diego, San Diego, CA), Martin Hunt (Nadi Inc., West Lafeyette, IN), Bhaskar D. Rao (Elec. and Comput. Eng., UC San Diego, La Jolla, CA), and Harinath Garudadri (Qualcomm Inst., UC San Diego, San Diego, CA)

An objective of a hearing aid (HA) is to improve speech communication for the hearing-impaired listeners. The current best practices for dispensing HAs involve pure tone audiometry (PTA), a functional test that characterizes ability to detect pure tones, but not perception of complex time-frequency patterns in natural speech. We present a phonetic-contrast function (PCF) that takes a test-reference pair from the Minimal Contrast Set (MCS) and generates a time-frequency-energy matrix of contrast regions. The test word is first time-aligned to the reference word using modified dynamic time warping. The frequency axis is composed of 11 bands defined by an audiometric filterbank presented in a companion work. The energy is in a pseudo dB scale in the ±120 dBSPL range. We will demonstrate the operation of PCF for English MCS spoken by a male and a female speaker. In a companion work, we developed a functional test for listening competence based on MCS that include substitutions, deletions, and insertions of acoustic-phonetic features in the stimuli, as in [lock, log, locks]. Future work includes PCF-based investigations with hearing impaired listeners to improve intelligibility of fricatives, stops, etc. by refining the HA parameters in the 11 audiometric frequencies used in sound processing.

2pPP17. Verification of interaural cues using OpenSoundNavigator. Hannah Tsai (Commun. Sci. & Disord., Western Washington Univ., 13519 SE 24th St., Bellevue, WA 98005, tsaih98@gmail.com) and Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., Bellingham, WA)

Advancements in hearing aid technology have led to increasingly complex digital signal processing features. Processing strategies in Oticon Opn hearing aids have features including complex noise reduction and directional microphone technology that allows for amplification of speech located behind the listener. These features are meant to assist listeners in complex acoustic scenes, but it is currently unknown whether there are unintentional effects on the spatial location of auditory objects in the environment. Additionally, do open-fit versus occluding earmolds interact with the complex digital signal processing. This study examined effects of interaural cues for Oticon Opn hearing aids. Devices were placed on a Bruel & Kjaer Head and Torso Simulator (HATS) in an anechoic chamber. Spondees were played from 11 speakers (±75°) separated by 15° azimuth in quiet and in the presence of a speech-shaped noise. Two-channel recordings were analyzed using binaural models to evaluate frequency specific interaural cues for each speaker location. Hearing aids were coupled to HATS using open-fit and occluded domes and compared to unaided recordings. All features available in the OpnSoundNavigator fitting software were systematically evaluated. Enhancements, reductions, and distortions of interaural cues were found across conditions in comparison to unaided recordings.

2pPP18. The effects of aging on the across-frequency processing of interaural time differences. Anhelina Bilokon (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., College Park, MD 20742, abilokon@umd.edu) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Aging and hearing loss cause temporal processing deficits, including poorer interaural-time-difference (ITD) processing, for simple stimuli like tones. However, there is little understanding of how aging and hearing loss affect across-frequency ITD processing. Binaural models suggest that lateralization depends on the alignment of peaks of the interaural cross-

correlation function across frequencies ("straightness") and considerable weighting around 700 Hz ("dominant region"). We hypothesized that aging and hearing loss would reduce across-frequency ITD processing and/or shift the dominant region to lower frequencies. Sixteen younger normal-hearing (YNH, <45 yrs), ten older normal-hearing (ONH, >65 yrs), and ten older hearing-impaired (OHI) listeners performed an intracranial lateralization task. They were presented complex tones and narrowband noises with fixed upper-frequency boundaries, varying bandwidths, and ITDs. For complex tones, the range of lateralization was smaller for older compared to younger listeners, but there were no effects of hearing loss. Some conditions showed less efficient across-frequency ITD processing with age, and others showed the dominant region shifted to lower frequencies. For narrowband noises, there were no effects of age or hearing loss, suggesting that envelope ITDs can affect lateralization in this task. These results suggest that accounting for aging in across-frequency binaural processing models is important.

2pPP19. Objectivization of the occlusion effect induced by earplugs in laboratory conditions—Effect of earplug type, insertion depth and background noise levels. Hugo Saint-Gaudens (Mech. Eng., ETS (Ecole de technologie superieure), 1100 Notre-Dame West, Montreal, QC H3C 1K3, Canada, hugo.saint-gaudens.1@ens.etsmtl.ca), Hugues Nélisse, Franck Sgard (Institut de Recherche Robert-Sauvé en Santé et en Sécurité du Travail (IRSST), Montreal, QC, Canada), and Olivier Doutres (Mech. Eng., ETS (Ecole de Technologie Superieure), Montreal, QC, Canada)

Blocking the earcanal's entrance with an earplug can lead users to experience discomforts, one being the occlusion effect (OE), typically described as a distorted perception of one's own voice. This discomfort sometimes causes users to misuse or remove their earplugs which significantly lower their efficiency. Reducing the OE generated by earplugs is therefore critical to make them more comfortable. However, assessing the OE is cumbersome and time-consuming as participants' feedback is required. Moreover, the influence of factors on the OE, namely, the type or earplug, the insertion depth, and the background noise level, remains to be understood. Hence, this ongoing research aims at objectivizing the OE induced by earplugs during speech. To do so, the OE is assessed in laboratory conditions with 30 normal hearing participants using a questionnaire and by using surrogate earplugs for in-ear microphonic measurements. Various sound pressure level-based indicators are proposed and correlated to the (dis)comfort during the objectivization step. Multiple combinations of earplugs, insertion depths and background noise levels are tested to obtain a ranking of the conditions generating more or less OE. The objectivization of the OE will be a useful tool for manufacturers developing new earplugs without requiring participants' feedback.

2pPP20. Event-related potential responses to differences in vibrotactile frequency: Evidence for continuous encoding of tactile information during early perception. M. Ryan Henderson (Villanova Univ., 800 Lancaster Ave., Villanova, PA 19085, mhende13@villanova.edu) and Joseph C. Toscano (Villanova Univ., Villanova, PA)

Perception of vibrotactile frequency is driven by activation of different types of mechanoreceptors embedded in the skin, which respond to vibrations in a frequency range of approximately 0.4 to 500 Hz. While previous behavioral work has measured participants' sensitivity to differences in vibration frequency, it is unclear how this information is encoded in neural representations during perception. This issue was addressed in an event-related potential (ERP) experiment investigating responses to vibrotactile stimuli. EEG data were recorded while participants rested their hand on a tactile transducer. On each trial, the transducer delivered a sinusoidal vibration with a duration of 1000 ms. Frequency of vibration ranged from 30 to 70 Hz in 10 Hz steps. Results revealed that frequency is encoded in early somatosensory ERP responses recorded at Cz. Specifically, the amplitude of a negative-going ERP component, centered at approximately 50 ms post-stimulus onset, increased as stimulus frequency increased from 40 to 70 Hz. These results demonstrate that vibrotactile frequency is encoded continuously during perception and that these responses can be measured using scalp-recorded EEG.

2pPP21. Evaluating remote post-fitting support services for over-the-counter hearing devices. Harley J. Wheeler (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, wheel488@umn.edu), Peggy Nelson (Ctr for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), John Ellison, and Erin O'Neill (GN Adv. Sci., Minneapolis, MN)

Over-the-counter (OTC) hearing aids are a more affordable and accessible option for many individuals with mild-to-moderate hearing loss. However, many clinicians and industry partners are apprehensive about how successful these new hearing aid (HA) users will be without the traditional support of audiologists. Much effort has gone into making the fitting and personalization of hearing aids accessible for OTC customers, but other post-fitting support services have not yet been translated to a remote format. Specifically, the individualized counseling audiologists traditionally provide is currently inaccessible to new HA users. The aim of this study is to explore ways of translating important aspects of audiologic counseling that contribute to satisfaction, use, and retention of HAs, to a format that is not dependent upon services provided within the clinic. Individuals with mild-to-moderate hearing loss were recruited and randomly assigned to one of three treatment groups: (1) a traditional clinical treatment model; (2) a model reflective of the typical "hands off" OTC experience; or (3) a non-clinical model with orientation and personalized counseling materials provided in a remote format. We hypothesize that individuals receiving personalized counseling remotely will be as successful as those receiving traditional clinical counseling, and more successful than typical OTC customers.

2pPP22. Effect of target word position on use of sentence context in acoustic-hearing listeners to spectrally degraded speech. Anna R. Tinnemore (Neurosci. and Cognit. Sci. Program, Univ. of Maryland, 0100 LeFrak Hall, College Park, MD 20742, annat@umd.edu), Sandra Gordon-Salant, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Understanding of degraded speech can be greatly improved by using sentence context cues. While sentence context effects for sentence-final words are well known, relatively little is known about sentence context effects for sentence-initial words. It is possible that subsequent sentence context can be used to improve understanding of a missed initial word. This study measured phoneme classification of a voice-onset time continuum. The target words from the continuum were presented in one of two sentence locations (beginning, end), and the sentence context cued one target word interpretation (based on phoneme classification) or the other. Acoustic-hearing listeners were presented sentences in clear and spectrally degraded (vocoded) conditions. Preliminary results suggest that sentence context effects are greater when the target word is at the beginning rather than the end of the sentence. We hypothesize that listeners may revise their initial acoustics-based interpretation of the degraded target word based on the sentence context when the target word is first in the sentence, but not when it is last. These location-dependent context effects may interact with individual cognitive skills, such as working memory and processing speed. The data supporting this hypothesis and an analysis of contributing factors (e.g., cognitive skills, age) will be presented.

2pPP23. Age-related rollover for spectrally degraded temporal speech contrasts. Anna R. Tinnemore (Neurosci. and Cognit. Sci. Program, Univ. of Maryland, 0100 LeFrak Hall, College Park, MD 20742, annat@umd.edu), Erin Doyle (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), Pallavi Atluri (Univ. of Maryland, College Park, College Park, MD), Miranda Cleary, Bobby E. Gibbs, Chengjie G. Huang, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

Some speech sounds, or phonemes, are primarily distinguished acoustically from other phonemes by temporal cues. A decline in temporal-processing abilities is a common consequence of aging that might cause a listener to have difficulty differentiating between sounds, especially when the speech is degraded. One type of signal degradation occurs in a cochlear-

implant vocoder simulation, which eliminates most of the spectral information in speech while maintaining the temporal envelope. In addition, temporal processing abilities may be challenged at higher presentation levels causing a decrease in performance (i.e., rollover), particularly in older listeners. This study measures the phoneme classification of younger and older listeners of four spectro-temporal contrasts when presented in quiet with and without vocoding at a range of presentation levels. We hypothesized that, due to declines in temporal processing abilities, older listeners would be more susceptible to rollover, measured by a decrease in phoneme classification performance for degraded temporal contrasts as presentation levels increase. We further hypothesized that performance rollover would occur more often for vocoded speech than for unprocessed speech, especially in older listeners. Preliminary data show interactions between speech contrasts, presentation levels, and vocoding. Our results will help facilitate understanding speech perception difficulties in older cochlear-implant users.

2pPP24. Hourly sound level exposures for preterm infants in the neonatal intensive care unit. Jana Khudr (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL, jkhudr2@illinois.edu), Rohit M. Ananthanarayana, and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Infants born preterm are at greater risk for auditory dysfunction than full-term infants. To better understand and characterize the neonatal intensive care unit (NICU) auditory experience, we sought to examine the sound pressure levels (SPLs) in the NICU and the presence of a circadian pattern of sound level exposure. Data were collected for very preterm infants (born $\leq$ 32 weeks' gestation; n = 36) during NICU stay. Audio recordings were collected over 24-hour intervals, three times per week for each subject using a LENA recorder that was adhered to the inside wall of the infant's incubator or crib. Average hourly SPL values were calculated from the raw recordings. Preliminary analysis indicates that the highest hourly exposures occurred during the hours of 8–9 AM and 8–9 PM, presumably corresponding to a shift change of the NICU nursing staff. Ongoing analyses are examining whether 24-hour patterns of exposure are affected by bed type and location in the NICU. It is hoped that this line of study will lead to interventions designed to prevent audiological impairments associated with preterm birth and NICU environmental exposures. [Work supported by NIH Grant R21-DC017820.]

2pPP25. Daily sound level exposure for preterm infants in the neonatal intensive care unit. Lauren A. Vicencio (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, lav3@illinois.edu), Rohit M. Ananthanarayana, Jana Khudr, and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

The neonatal intensive care unit (NICU) provides lifesaving care to premature neonates. During this vulnerable time of life, infants are exposed to a variety of sounds uncharacteristic of the womb. Within the NICU, sound pressure levels have the potential to vary across days. The American Academy of Pediatrics has recommended an hourly noise exposure limit for NICUs of less than 45 dBA. The aim of this study is to assess daily sound pressure level exposures for preterm infants in the NICU. Sound pressure levels were measured using 24-hour audio recording devices (LENA). The device was placed inside the crib or incubator with the infant or moved to the area of care if the infant was removed from the crib or incubator. We collected 672 24-hour recordings from 36 very preterm infants ($\leq$32 weeks gestational age). Preliminary analysis indicates the average daily exposures ranged from 53.4 to 68.6 dB SPL across subjects and days. Ongoing analyses are examining potential environmental and medical factors contributing to high levels of noise exposure for these infants. [Work supported by NIH Grant R21-DC017820.]

2pPP26. Band importance for speech-in-speech recognition in the presence of extended high-frequency cues. Rohit M. Ananthanarayana (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, rohitma2@illinois.edu), Emily Buss (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

Extended high-frequency (EHF; 8–20 kHz) cues support speech recognition in noisy backgrounds, particularly when the masker has reduced EHF levels relative to the target. This scenario can occur in natural auditory scenes when the target talker is facing the listener, but the masker talkers are not. The EHF benefit stands in contrast to past studies that have focused on lower frequencies and presumed that EHFs play no role in speech intelligibility. Although EHF cues improve speech recognition, it is unclear how the magnitude of benefit compares to that of other portions of the speech spectrum. In this ongoing study, we measure band importance functions (BIFs) for a female target and two-talker masker by notch filtering individual contiguous bands from 40 to 20 000 Hz. With the target facing the listener, two masking conditions were tested: (1) masker facing the listener; (2) masker facing 56.25°. Preliminary data indicate an interaction between the filtered band and masker head orientation. For the facing condition, the BIF shows a peak between 0.4 and 3 kHz and drops sharply at higher frequencies, resembling previous data. When the masker faces away, however, the benefit of EHFs increases relative to the lower bands, somewhat flattening the BIF. [Work supported by NIH grant R01-DC019745].

2pPP27. Speech in noise performance in individuals with misophonia and hyperacusis using behavioral and auditory brainstem response. Gibbeum Kim (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 605 E White St., D39, Champaign, IL 61820, gibbeum2@illinois.edu), Ragnar Lindberg, Namitha Jain, and Fatima T. Husain (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Hyperacusis is reduced tolerance to everyday environmental sounds, whereas misophonia is a relatively unknown disorder in which characterized by intense and excessive emotional responses to specific "trigger" sounds. Few studies have investigated ability of speech in noise (SiN) and examined them using auditory brainstem responses (ABR)—which is used to measure early auditory processing and overall hearing function in human listeners, in particular for the sound sensitivity disorders of misophonia and hyperacusis. The primary aim of this study was to investigate the relationships between SiN performance and ABR components, specifically for group differences. A total of 60 participants were categorized into three groups: 13 with misophonia, 12 with hyperacusis, and 35 control group. Our findings indicated that (1) the misophonia group showed poor speech in noise performance than control and hyperacusis groups at 20 and 5 SNR; (2) both hyperacusis and misophonia groups showed enhanced wave I amplitude compared with the control group; and (3) the misophonia group showed prolonged wave V latency compared to the other two groups. Additionally, delayed ABR wave V latency in misophonia group was correlated with decreased speech-in-noise performance score, suggesting possible delays in processing of speech sounds in this group leading to poorer performance.

2pPP28. Investigating the parameters affecting spectro-temporal integration in pitch. J. C. Kooistra (Univ. of Michigan, Kresge Hearing Res. Inst., 4605 Medical Sci. Unit II, Ann Arbor, MI 48109, kooipond@umich.edu) and Anahita H. Mehta (Univ. of Michigan, Ann Arbor, MI)

Previous research has shown that a virtual pitch percept of the fundamental frequency (F0) can be extracted from a sequence of non-overlapping, harmonically related tones in the presence of background noise. We have established that the F0 percept is strongest for low numbered harmonics and

absent when only high-numbered harmonics (>10) are presented as well as when the tones are inharmonically related. The present series of experiments explores the effect of dichotic presentation, notched noise, and varying degrees of inharmonicity on this virtual pitch percept. F0 difference limens (F0DLs) for synchronous and sequentially presented components in noise were measured in normal-hearing participants. For the dichotic conditions, each ear received a random subset of components to investigate if this percept was dependent on peripheral interactions. The inharmonic conditions varied the extent of spectral jitter on each of the components (by 10%, 20%, and 30%) to understand if F0DLs worsened with increasing amounts of jitter. Finally, the notched noise conditions investigated whether background noise within the harmonic frequencies was essential for the F0 percept. Preliminary results indicate that the virtual pitch can be perceived dichotically and the characteristics of the background noise and harmonic relations play key roles in this illusory percept.

2pPP29. Investigating the neural correlates of the emergence of pitch using harmonic and inharmonic stimuli. Anjelica Ferguson (Univ. of Michigan, Kresge Hearing Res. Inst., 4605 Medical Sci. Unit II, Ann Arbor, MI 48109, anjelife@umich.edu) and Anahita H. Mehta (Univ. of Michigan, Ann Arbor, MI)

The emergence of a pitch percept, referred to as the pitch onset response (POR), has predominantly been studied using transitions from noise to regular interval noise (RIN), click trains and more recently, harmonic complex tones. The POR is typically an asymmetrical change response which shows a larger response for stimuli that go from having no salient pitch to a salient pitch but not vice versa. However, the spectro-temporal differences between the pitch inducing stimulus and noise make it difficult to disentangle whether the response reflects a change in regularity or a true difference in pitch salience. The aim of this experiment is to use spectrally similar stimuli that differ in pitch salience (harmonic versus inharmonic tones) without changes in temporal regularity. We recorded EEG responses in normal-hearing participants to investigate the cortical correlates of pitch emergence in humans. We used the noise to RIN transitions as a baseline measure to replicate the asymmetry in the POR. We also used transitions from harmonic to inharmonic stimuli and vice versa to investigate if the POR asymmetry would hold for stimuli with minimal spectro-temporal differences. Results from this study will help us further understand the parameters affecting this response.

2pPP30. Cross-species characterization of joint otoacoustic emission profiles in sensorineural hearing loss. Samantha Hauser (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, samantha.n.hauser@gmail.com), Michael G. Heinz (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN), and Hari Bharadwaj (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Current clinical assessment of otoacoustic emissions (OAEs) is typically limited to presence or absence of emissions to detect hearing loss, but recent work suggests advanced analyses and data collection methods have the potential to improve the diagnostic utility of OAEs. OAEs arise from two distinct mechanisms, nonlinear distortion and linear reflection, which are both sensitive to the health of outer hair cells but may reflect separate aspects of cochlear mechanical function. Joint distortion-reflection OAE profiles offer a non-invasive characterization of peripheral auditory physiology and dysfunction (Abdala and Kalluri, 2017). In this study, swept-tone distortion-product OAEs were separated into distortion and reflection components. Swept-tone stimulus-frequency OAEs, reflection-type emissions, were measured using a suppressor paradigm. Joint OAE profiles will be estimated for multiple pre-clinical chinchilla models of sensorineural hearing loss (SNHL) and in humans with either normal hearing or sensorineural hearing loss. In chinchillas, we expect SNHL models with distinct profiles of cochlear damage to yield separable joint OAE profiles. Findings in chinchillas will provide mechanistic insight into OAE profile clusters in humans. These data test the sensitivity of joint OAE profiles to specific cochlear dysfunctions and contribute to development of precision diagnostics for SNHL.

2pPP31. Tinnitus profile and speech in noise performance in military and non-military individuals: A comparative study. Namitha Jain (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth, Champaign, IL 61820, namitha3@illinois.edu), Gibbeum Kim (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Yihsin Tai (Dept. of Speech Pathol. and Audiol., Ball State Univ., Muncie, IN), and Fatima T. Husain (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Military-affiliated-individuals (MI) are more likely to develop hearing loss and tinnitus during training and deployment due to noise, ototoxic chemicals, and other events. There have been extensive studies about the incidence, prevalence, and functional impact of tinnitus in MI, but careful comparison of its impact between MI and non-military-individuals (NMI) is understudied. We aimed to compile and compare audiological and tinnitus profiles in MI and NMI. A total of 62 MI and 158 NMI were classified into four sub-groups; (1) normal hearing without tinnitus (NH_{CON}); (2) normal hearing with tinnitus (NH_{TIN}); (3) hearing loss without tinnitus (HL_{CON}); and (4) hearing loss with tinnitus (HL_{TIN}). Each participant underwent complete audiological, speech-in-noise assessments (QuickSIN) and tinnitus profiling. Our overall observations revealed that MI_{HL_TIN} had higher tinnitus-related distress than NMI_{HL_TIN}in the relaxation subscale of tinnitus functional index. In MI_{NH_TIN} group perception of tinnitus was much louder than the NMI_{NH_TIN}. These findings highlight the importance of considering military status when developing assessment and management techniques. Military status had no effect on speech-in-noise scores, in both populations TIN_{NH} group outperformed the CON_{NH}, while the TIN_{HL} group performed worse than CON_{HL}. Better speech-in-noise performance in NH_{TIN} may suggest heightened selective auditory attention or a potential spotlight phenomenon in this group.

2pPP32. Quantifying the impact of hearing-aid microphone placement and head-mounted devices on binaural cues from head-related transfer functions. Gabriel S. Weeldreyer (Durham School of Architectural Eng. and Construction, Univ. of Nebraska–Lincoln, 1110 S 67th St., Omaha, NE 68182, gweeldreyer2@unl.edu), Z. Ellen Peng (Boys Town National Res. Hospital, Omaha, NE), Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska–Lincoln, Omaha, NE), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Emerging techniques for assessment and training of spatial hearing use virtual reality (VR) devices to present auditory-visual cues. Sounds may be processed using head-related transfer functions (HRTFs), which capture directional acoustic cues, and visuals may be presented using a head-mounted display (HMD). For users of hearing devices such as hearing aids or cochlear implants, microphone placement may alter the binaural cues carried by the HRTF. Cues may also be distorted in presentations that combine HMD visuals with loudspeaker audio. This work seeks to understand these impacts by measuring and creating a database of HRTFs in the presence of hearing aids and HMDs. HRTFs were recorded within an anechoic chamber, using a standard manikin in a loudspeaker array located on the horizontal plane. Recordings were made with real sound sources of 5.625° resolution and virtual sources of 1° resolution. We will compare interaural time and level differences (ITDs and ILDs) from HRTFs measured through hearing-aid microphones with varied microphone placement (i.e., behind-the-ear and in-the-ear) and the presence versus absence of an HMD during the HRTF recording. Results will provide insights regarding HRTFs for individuals with hearing devices, as well as characterization of the HMD impact on VR-guided individual HRTF measurement.

2pPP33. Modeling the nonlinear mechanics and dynamics of Cochlear Outer Hair Cell Stereocilia. 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)

Sound waves vibrating the eardrum excite the ossicles in the middle ear ultimately driving waves in the cochlea. Cochlear vibrations are processed by inner hair cells and outer hair cells (OHCs). Our focus is on the OHCs that nonlinearly amplify the sound converting a time-varying motion of its apically adorned hair bundle (HB) to an alternating current. The OHC HB consists of roughly three rows of stereocilia arranged according to their heights. Understanding how the bundle stiffness, sensitivity, and transduction current depend on the physiology and anatomy of the stereocilia is crucial and open question. Therefore, we are developing a three-row model of an isolated HB to quantify each row's contribution to the passive and active mechanics of the HB. The derived equations of motion include the nonlinear kinematics, viscoelastic HB mechanics, and the nonlinear response of the mechano-electric transducer channels coupled to an adaptation mechanism. We also linearize the model to conduct stability analysis and determine the dependence of the responses on the rate constants. Our preliminary results show a higher current influx through the middle row than the shortest row and a more significant stiffness contribution from the middle-to-tallest row pair compared to the shortest-to-middle row pair connection.

2pPP34. Typical sound level in environmental sound representations. Urszula Oszczapinska (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Baker 342C, Pittsburgh, PA 15213, uoszczap@andrew.cmu.edu), Bridget Nance, Seojun Jang, and Laurie M. Heller (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Although the sound level reaching a listener's ear depends upon the sound source level and the environment, a stable source level can be perceived (McDermott *et al*., 2021). Nonetheless, variation in sound level can disrupt recognition in a short-term old/new task (Susini *et al*., 2019). We asked whether there is evidence of long-term memory of the typical level of everyday sounds. First, we found that listeners can report the level at which they typically hear a sound. Next, we compared sound judgements over headphones (ESC-50 dataset) across two conditions: (1) "typical": levels set to produce the loudness experienced as "typical" for each sound (as determined by pilot studies); and (2) "equal": levels at 70 dB SPL. Recognition, familiarity, and pleasantness were judged. There was no significant difference in recognition accuracy between level conditions and no interaction with whether sounds were louder or softer than their typical levels. In addition, recognition increased as sound familiarity increased, but this did not interact with level condition. Furthermore, consistent with past findings, sound pleasantness decreased as loudness increased, but this effect did not depend upon the condition. [Work supported by REAM.]

Session 2pSA

Structural Acoustics and Vibration, Engineering Acoustics, Computational Acoustics, and Physical Acoustics: Acoustic Metamaterials II

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Chair's Introduction—1:00

Invited Papers

1:05

2pSA1. Reaching new levels of wave scattering via piezoelectric metamaterials and electro-momentum coupling. Jeong-Ho Lee (Mech. Eng., Univ. of California, Berkeley, 2521 Hearst Ave., Berkeley, CA 94709, lee.jh@berkeley.edu), Zhizhou Zhang, and Grace X. Gu (Mech. Eng., Univ. of California, Berkeley, Berkeley, CA)

Piezoelectric materials are materials that can convert mechanical energy into electrical energy (and vice versa). In addition to the well-known electromechanical interaction between mechanical deformation and electric fields, a recent discovery revealed that the macroscopic linear momentum of piezoelectric materials is also coupled with the electric field. This means that their kinetic movement can be controlled by the electric field. This effect is termed electro-momentum (EM) coupling and provides a new design degree of freedom for piezoelectric devices with better properties and dynamic performance. In this talk, theoretical bounds of EM coupling in wave scattering will be discussed, which can provide valuable information on estimating the performance space of piezoelectric metamaterials for various applications such as acoustic sensing and energy harvesting. The results show that scattering performance from EM coupling can be of the same order of magnitude as that of the scatterer's geometric features, such as shape and size, which are considered major factors for scattering performance. Furthermore, its capability of promising wave manipulation will be demonstrated, i.e., tunable scattering-cloaking piezoelectric devices via EM coupling. Moreover, how material properties and geometrical microstructures affect EM coupling will be investigated, which allows researchers to precisely control EM coupling in piezoelectric materials.

1:25

2pSA2. Dissipation-induced acoustic nonreciprocity. Arkadii Krokhin (Phys., Univ. of North Texas, 1155 Union Circle # 311427, Denton, TX 76203, arkady@unt.edu)

Linear acoustic transmission through a stationary elastic medium possesses Rayleigh reciprocal symmetry with respect to pressure produced by a source at point $A(B)$ and measured at point $B(A)$, $p_A(r_B) = p_B(r_A)$. In ideal (inviscid) fluids, pressure plays the role of the scalar potential for velocity, $\mathbf{v} \sim \nabla p$. The Rayleigh reciprocity theorem is formulated for velocity potentials but not for their gradients. Any function of the absolute value of velocity, $v(\mathbf{r})$, lacks reciprocal symmetry. In particular, acoustic intencity $I = pv$ is not symmetric, $I_A(r_B) \neq I_B(r_A)$. Dynamics of ideal fluid is time-reversible, and lack of reciprocal symmetry is attributed to asymmetry in scattering. However, in a viscous environment dissipation breaks T symmetry making fluid dynamics irreversible. We report that acoustic transmission is not only irreversible but also becomes nonreciprocal due to different amount of dissipated energy for forward and backward propagation, provided that mirror symmetry (P symmetry) is broken. Since dissipation occurs within a narrow viscous layer at solid–fluid interface, the amount of nonreciprocity in transmission strongly depends on the quality of the surface of scatterers. Experiments performed with scatterers of different surface quality show how surface roughness affects the level of nonreciprocity. In a series of experiments with asymmetric phononic crystal, Tesla valve, and acoustic cavity, we demonstrate dissipation-induced nonreciprocity in transmission. [This work is supported by the NSF under EFRI Grant No. 1741677.]

1:45

2pSA3. Aerospace grade acoustic metamaterial design, modeling, and experimental validation. Ali Sheikh (Mech. and Mater. Eng., Queen's Univ., 8 Davidson Dr., Acton, ON L7J0A4, Canada, 15ahs2@queensu.ca), Chris Mechefske (Mech. and Mater. Eng., Queen's Univ., Kingston, ON, Canada), Charly Kone (National Res. Council Canada, Ottawa, ON, Canada), Zacharie Laly (Université de Sherbrooke, Sherbrooke, QC, Canada), and Sebastian Ghinet (National Res. Council Canada, Ottawa, ON, Canada)

Vertical Take-Off and Landing Urban Aerial Mobility vehicle propulsion systems use rotors and blades to achieve lift, which results in heightened noise at the blade passing frequency. These frequencies must be attenuated to reduce cabin discomfort and comply with noise exposure limits. The design must absorb 350 Hz noise, be fire retardant to comply with aviation standards, and be maximally one inch thick. This paper outlines the design, computational simulation, and experimental validation of a novel aerospace-grade acoustic metamaterial. It contributes to the field of computational analysis of new metamaterial structures and experimental validation using prototype Helmholtz unit cells. Achieving maximum sound absorption with the aerospace requirements of minimal mass and fire safety led to the consideration of Nomex honeycomb Helmholtz resonator acoustic panels. Their sub-wavelength attenuation, tunability, lightweight and fire resistance make them an optimal fit for this application. A novel phenomenon was observed in experimental impedance tube testing where multiple Nomex honeycomb cells acted as one resonator cavity. This discovery allowed for samples of only one inch in thickness (cell depth) to achieve 83% sound absorption at 348 Hz. Experimental values match well with the calculated analytical solution using the Helmholtz equation and simulated sound absorption in COMSOL MultiPhysics. The acoustic metamaterial achieves tunable noise attenuation with minimal thickness, weight, and fire suppression using a novel design.

2:00

2pSA4. Controlling elastic wave propagation in plates using magic squares. Marupong Vongsiri (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, mxvongsiri@shockers.wichita.edu), Maria Carrillo-Munoz, and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

A magic square is defined as a square array of distinct positive numbers arranged such that the sum along the horizontal, vertical, and main diagonal directions are the same, which is called a magic number. Previously, we leveraged the magic square concept to create a new class of phononic structures whose dispersion behavior can be tuned without altering their global mass and stiffness. In this study, we study the dispersion behavior of a thin plate with non-structural mass arranged in a magic square pattern. The plate unit cell is partitioned into nine portions and individual point masses are embedded at the center of each portion. Our results show that the magic square mass-embedded plate provides a low-frequency out-of-plane bandgap whose widths can be controlled while maintaining the global mass of the plate structure. Additionally, we investigate a magic plate concept by applying different densities at each portion associated with the magic square patterns. Our preliminary results show that the magic density distribution results in the emergence of polarized bandgaps and provides a possibility of altering the plate modal frequency without altering the mode shape or the global mass.

2:15

2pSA5. Roton- and maxon-like dispersion relations in elastic metamaterial. Peng Zhang (Civil and Environ. Eng., Univ. of Utah, Salt Lake City, UT), Fei Chen (Mech. Eng., Univ. of Utah, Salt Lake City, UT), Xuan Zhu (Civil and Environ. Eng., Univ. of Utah, 110 Central Campus Dr., Salt Lake City, UT 84112, xuan.peter.zhu@utah.edu), and Pai Wang (Mech. Eng., Univ. of Utah, Salt Lake City, UT)

Rotons and maxons are local minima and maxima in the frequency-wavenumber spectrum, respectively. They represent wave modes with a vanishing group velocity and a finite phase velocity. Recent advancement showed that roton dispersion relations are not restricted to low-temperature quantum systems but can also be engineered in 3D elastic metamaterials. Here, we report that zero-group-velocity (ZGV) wave modes, similar to rotons and maxons, can be found in elastic waveguide structures with resonators. In addition, we perform parametric study on the parameters of the resonator unit. By tuning the parameters, the roton and maxon may coalesce into a second-order critical point (undulation point or stationary inflection point), where the second derivative of frequency with respect to wavenumber vanishes. This introduces a new non-spreading localized wave mode. Not only is such wave mode non-propogating, but it is also non-diffusing. In this case, the highly localized vibration can support energy trapping and enhance the signal-to-noise ratio in wave-based structural health monitoring.

2:30

2pSA6. Effect of cone angles on the acoustical properties of spinodoid structures. Brittany Wojciechowski (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, brwojciechowski@shockers.wichita.edu) and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Our recent investigations [1] show that the sound absorption behavior of spinodoid structures is highly dependent on their underlying spinodal-like cellular architecture. This architecture is controlled using a parameterized Gaussian random function, wherein a non-uniform orientation distribution function restricts the sampled wave vectors to lie within user-defined cone angles. While our previous work was restricted to the four different spinodoid topologies proposed by Kumar *et al*. [2], in this work, we investigate the feasibility of altering the cone angle limits with the goal of generating topologies that can provide increased acoustical benefits. We systematically alter the cone angles with respect to the conventional orthogonal Cartesian basis and investigate their effect on the resultant cellular topologies. A select set of spinodoid structures are then 3D printed and tested using a normal incidence impedance tube. Our results show that spinodoid structures provide a rich design space for the development of porous noise reduction materials that can be tuned to achieve application-specific acoustical performance. [1] B. Wojciechowski and B. Sharma, "Acoustical properties of spinodoid porous structures," *J. Acoust. Soc. Am.* **150**(4), A148 (2021). [2] S. Kumar, S. Tan, L. Zheng, and D. M. Kochmann, "Inverse-designed spinodoid metamaterials," *npj Comput. Mater.* **6**(1), 1 (2020).

2:45

2pSA7. Dispersion behaviour of a non-resonant elastic metamaterial. Claudia L. Clarke (DSTL, DSTL Porton Down, Salisbury SP4 0JQ, United Kingdom, cclarke4@dstl.gov.uk)

Metamaterials are of great interest on account of their tunable material proper-ties, arising from an internal microstructure, which can often be chosen to give a desired behaviour. The behaviours of acoustic metamaterials are often derived via analogy with electromagnetic metamaterials, and typically described using Willis relations. Here an elastic metamaterial consisting of a series of elastic plates interspaced by fluid is considered. It is found that the effective constitutive equations contain higher derivative terms than would normally be present in the Willis (or elastic) constitutive equations for an anisotropic material. This leads to behaviour that is highly dispersive due to an "emergent scale" associated with bending on the constitutive equations for an anisotropic material. This leads to behaviour that is highly dispersive due to an "emergent scale" associated with bending on the constituent plates. The existence of this scale is due to the boundary conditions on the plate-fluid interfaces and appears to be a purely elastic phenomena. The metamaterial in this case has no resonant inclusions; nevertheless, the ef-fective behaviour leads to broadband dispersion and is different from anisotropic elasticity. The dependence of the dispersion curves on the material properties of the constituents is examined and the connection with fractional derivative theories and rotons considered.

3:00–3:15 Break

3:15

2pSA8. Resonant Willis metamaterials based on fractal geometry. 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), Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH), Cody M. Best (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A class of metamaterials that exhibit mass-momentum coupling are known as Willis materials. The mass-momentum coupling arises from hidden degrees of freedom that are asymmetrical with respect to the overall material. Resonant Willis metamaterials incorporate resonators, e.g., a Helmholtz resonator, along with material asymmetry to effectively obtain Willis coupling over a narrow frequency band. By incorporating fractal geometry into the resonator shapes Willis coupling can be obtained over a broader frequency range. This is accomplished by self-similar resonator shapes, which translates into resonance frequencies that exhibit a power-law dependence. Furthermore, manipulation of the internal asymmetry allows for tuning of the Willis coupling. This study reports on the measurement and modeling of resonant Willis metamaterials based on fractal geometry. Measurements are obtained with a transmission impedance tube, and predictions based on a lumped-element model. With resonator volume shapes based on the Sierpiński triangle, Willis coupling is obtained over a frequency range of 500 to 1500 Hz. Furthermore, for a subset of the resonators, Willis coupling is tuned by inverting the asymmetric location of the resonator necks.

3:30

2pSA9. Modeling and characterizing the dynamic response of Octet-Truss lattice structures using elastic parameters determined from Resonant Ultrasound Spectroscopy. Karl A. Fisher (Lawrence Livermore National Lab., 7000 E. Ave., Livermore, CA 94551, fisher34@llnl.gov), Jenny Wang, and Brian Tran (Lawrence Livermore National Lab., Livermore, CA)

Modeling and characterization of additively manufactured materials and structures is a critical stage in the design process. This presentation will present an approach using Resonant Ultrasound Spectroscopy to determine the elastic tensor of an additively manufactured Octet Truss lattice structure. The lattice structures are fabricated from an alloy of Titanium using a laser powder bed fusion process. Material characterization begins by applying resonance techniques on simple regular parallelepiped shapes. Solid and lattice samples are measured to obtain estimates for the elastic properties of the bulk material and the *effective* elastic properties of the lattice structure. The measured elastic properties are then incorporated into structural models representing continuum approximations of several test parts. Comparisons between the continuum models and the experimental measurements indicate agreement over a wide frequency range within a long wavelength approximation defined for the lattice. The measurements also suggest that there is additional physics and geometrical effects accounted for in the RUS continuum approximations of the lattice that are not completely captured by the full finite model utilizing only base material properties.

3:45

2pSA10. Wave propagation in a phononic material with periodic bilinear elastic layers. Elizabeth Smith (Mech. Sci. and Eng., Univ. of Illinois at Urbana Champaign, Mech. Eng. Bldg., 1206 W Green St., Urbana, IL 61801, esmith19@illinois.edu) and Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

This presentation discusses how bilinear elastic layers influence wave propagation in phononic materials. Bilinear elastic layers have been shown to produce wave propagation phenomenon such as shock waves, non-reciprocity, phase-reversal effects, and harmonic generation. Here, we examine how a propagating wave through a periodic array of linear elastic layers coupled with springs that exhibit bilinear stiffness can be tuned with the stress state of the excited wave. Using time-dependent finite element simulations, we study the interactions of bilinear springs and phononic materials, analyzing how the tensile or compressive stress state influences dispersion, band gaps, and energy transfer between frequencies. We select excitation waveforms to exploit the bilinear stiffness to fundamentally change the shape of the propagating wave. Results demonstrate phase-reversal effects, attenuation of tensile strain, and cumulative displacement offsets. To experimentally probe these results, bilinear stiffness couplings are physically realized by fabricating samples with prescribed delaminations between stiff and soft materials so the samples exhibit higher stiffness in compression than tension. Single layered samples are experimentally studied using both propagating waves and resonant methods. The nonlinear response from delaminations may be used to control frequency energy transfer and wave transformation for applications such as blast mitigation and passive mechanical sensing.

4:00

2pSA11. Computational and experimental analysis of lattices with modulated nonlinearity. Matthew D. Fronk (Phys., U.S. Naval Acad., 572C Holloway Rd., Annapolis, MD 21402, fronk@usna.edu), Caleb F. Sieck, Alec K. Ikei, and Matthew D. Guild (Code 7160, U.S. Naval Res. Lab., Washington, DC)

There has been growing interest in wave propagation in nonlinear lattices due to their amplitude-dependent dispersion. While previous studies have primarily considered lattices with spatially-uniform stiffness nonlinearity, this work explores lattices with nonlinear stiffness spatially-modulated in both sign and strength. A multiple scales perturbation analysis reveals exotic dispersion behavior, including the coexistence of hardening and softening within the same branch, the merging of two branches near the band edge, and the suppression of amplitude-dependent dispersion shifting across entire branches. Moreover, this study presents details of the computational and experimental validation of this newly-discovered nonlinear dispersion behavior. For numerical validation, direct simulation of the lattice equations of motion is carried-out. Distinctions are drawn between harmonic excitation of the lattice as compared to prescription of the lattice's initial displacement and velocity fields. An experimental realization of the lattice is proposed whereby the sign and strength of the nonlinearity is spatially modulated using springs grounded at different angles. Experimentally-feasible lattice parameters are detailed and the experimental set-up is explored computationally using motion simulation and finite element software. [Work supported by the Office of Naval Research.]

4:15

2pSA12. Highly-efficiency low-frequency acoustic energy harvesting with PDMS-modified loudspeakers. Xiuyuan Peng (Elec. and Comput. Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27705, xp21@duke.edu), Junfei Li, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

Noise is everywhere in our lives. Low-frequency noise in particular has a conspicuous if not disturbing presence since thicker materials are required to absorb or block it. Meanwhile, the very ubiquitousness of noise in our environment makes it a good candidate as a potential power source for micro-devices. Here we present an acoustic energy harvester (AEH) capable of achieving 99% sound absorption coefficient and 67% of energy conversion ratio at 58 Hz, with a fractional bandwidth of more than 34 %. The peak power conversion efficiency is nearly three times the previous state of the art. The harvester is made from an electrodynamic loudspeaker driver retrofitted with a custom-made PDMS surround for lower mechanical loss on the diaphragm. The low cost (<$50) makes our AEH suitable for large-scale deployment.

4:30

2pSA13. Ultra-broadband acoustic metasurface for speaker directivity control. Junfei Li (Elec. and Comput. Eng., Duke Univ., 101 Sci. Dr., Rm 3417, FCIEMAS bldg, Durham, NC 27708, junfei.li@duke.edu), Xiuyuan Peng, Gregory Hernandez, and Steven Cummer (Elec. and Comput. Eng., Duke Univ., Durham, NC)

Acoustic metasurfaces have shown remarkable capabilities for shaping acoustic wavefronts. However, most thin metamaterials can only work well for a narrow range of acoustic frequencies. This limitation has thus far dramatically limited their potential for human-based audio and acoustic

applications. Here we present a generic acoustic metasurface design approach to reshape acoustic wavefronts in a broad bandwidth. As a demonstration, we designed two acoustic lenses. The first acoustic lens converts a 4-in. speaker into an omnidirectional source, covering a frequency range from 1 to 10 kHz. The second acoustic lens delivers super-directionality for a 1-in. speaker from 1 to 20 kHz. Experimental measurements showed great agreement with the simulations. Our design approach enables the design of ultra-broadband metasurfaces useful for audio applications.

4:45

2pSA14. Unit cell of three nonlinear layers for the design of tunable nonlinear metamaterials. Pravinkumar R. Ghodake (Dept. of Mech. Eng., Indian Inst. of Technol. Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Tang *et al.* (2012) and Kube (2017–2018) discussed the harmonic scattering of elastic waves from nonlinear inclusions through a theoretical perspective modeling early-stage damages as quadratic and cubic nonlinear elastic material models. Achenbach and Wang (2017–2018) studied the harmonic scattering of a wave from a single nonlinear layer embedded in a linear elastic material using the reciprocity theorem in elastodynamics and the concept of compensatory waves. They demonstrated sensitivity of the back and forward scattered waves towards the dimension and the intensity. Based on this understanding, a new unit cell is proposed in this study that can act as a tunable nonlinear metamaterial, where we can tune the amplitudes of the back and forward-scattered nonlinear waves. The unit cell contains three nonlinear layers of equal widths; the intensity of the central layer, along with the total width of the unit cell, is varied to tune the harmonic responses of scattered waves. Computational studies are carried out to demonstrate the presence and absence of harmonically backscattered waves from a unit cell. In some case studies, parameters are tuned so that the amplitudes of 2nd harmonics (*2f*) of forward, backscattered waves and waves recorded at the interfaces of the three nonlinear layers.

TUESDAY AFTERNOON, 9 MAY 2023 — SALON I, 1:00 P.M. TO 5:00 P.M.

Session 2pSCa

Speech Communication and Psychological and Physiological Acoustics: Clear Speech(es) Across People, Places, and Time II (Poster Session)

Rajka Smiljanic, Cochair
Univ. of Texas at Austin, 305 E. 23rd St., B5100, Austin, TX 78712

Georgia Zellou, Cochair
Department of Linguistics, UC Davis, 1 Shields Ave., Davis, CA 95616-3086

All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd numbered abstracts will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even numbered abstracts will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

2pSCa1. Comparing intelligibility and recognition memory of human and text-to-speech voices. Nicholas Aoki (Linguist., Univ. of California, Davis, 469 Kerr Hall, One Shields Ave., Davis, CA 95616, nbaoki@ucdavis.edu) and Georgia Zellou (Linguist., Univ. of California, Davis, Davis, CA)

Clear speech can be more intelligible and better remembered than casual speech. Intelligibility and memory are assumed to be related—under the effortfulness hypothesis, the intelligibility-enhancing properties of clear speech (e.g., slower speech rate) facilitate processing, which frees up working memory resources and confers a memory advantage. However, prior work has only investigated naturally produced clear speech. How does clear speech influence memory for computer voices? The current study examines intelligibility and recognition memory of clear and casual speech for a text-to-speech (TTS) voice and a human voice, and compares older and younger adult listeners. In a speech-perception-in-noise task, clear speech was more intelligible than casual speech for both voices, the human was more intelligible than the TTS voice, and the younger adults performed better than the older adults. In an old-new recognition task, memory was better for the human voice relative to the TTS voice, consistent with the intelligibility results and the effortfulness hypothesis. However, inconsistent with the effortfulness hypothesis, there was no clear speech memory benefit, and the older adults performed better than the younger adults. The current findings are relevant for our understanding of human-computer interaction and theories about the relationship between speaking style and memory.

2pSCa2. Noise-induced change of tone dispersion in Cantonese speech. Ting Zhang (Dept. of Linguist and Translation, City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon, Hong Kong, tzhang343-c@my.cityu.edu.hk) and Bin Li (Dept. of Linguist. and Translation, City Univ. of Hong Kong, Kowloon, Hong Kong)

Noise effect on speaking non-tonal languages is reported on acoustic correlates including increased intensity, enlarged vowel space and raised fundamental frequency. Adverse influence may also be present in realization of tones, which however remains under-explored. Tone-space size and the according tone dispersion offer key indicators to the nature of a tonal system. The present study investigates whether and how Cantonese tones are modified in speaking under noise. Adult native speakers of Cantonese read

tri-syllabic words embedded in carrier sentences in quiet and under multi-talker babbles. The speech materials cover six tones (T55, T25, T33, T21, T23, T22) and are balanced between high-low and low-high tone combinations to avoid biased articulatory effects. Results confirm Lombard speech effect in Cantonese speaking. Moreover, under the noise condition, tone spaces of all six tones are enlarged and tones are more dispersed, especially in realization of T55 and T21 that mark the upper and lower ranges of the tonal space. Changes in tone production by female speakers are more significant than those in male speech. This study provides new evidence of phonetic accommodation under noise and in particular noise effects on sizes of tone space, which is not found in the production of monosyllabic words in Cantonese. Our speakers may have used labored speech production to compensate for signal degradation under babble noise, leading to higher dispersion of tones than clear speech in quiet.

2pSCa3. Clear speech and phonological typology: A case study of Tashlhiyt Berber. Georgia Zellou (UC Davis, Davis, CA, gzellou@ucdavis.edu), Mohamed Lahrouchi (Paris 8 & CNRS, Paris, France), and Karim Bensoukas (Univ. Mohammed 5, Rabat, Morocco)

Clear speech is a type of listener-oriented, intelligibility-enhancing mode of speaking. It has been shown to enhance the perceptibility of many different types of phonological contrasts, cross-linguistically. An open question is whether all phonological contrasts are enhanced to an equivalent extent in clear speech. In the current study, we ask whether rarer phonological patterns receive less of a clear speech intelligibility boost, relative to more common phoneme contrasts. Tashlhiyt Berber is an Afroasiatic language spoken in Morocco. Tashlhiyt has been well studied for having typologically uncommon phonotactic properties. This study examines the effect of clear speech on the discrimination of rarer lexical contrasts in Tashlhiyt Berber. We predict that the more typologically uncommon contrasts (e.g., word pairs containing complex and geminate initial onsets) will have a smaller increase in perceptibility from casual to clear speech than more common contrasts (e.g., singleton contrasts). Furthermore, native and naive listeners' (here, American English speakers) discrimination of these contrasts across speech styles is also compared. Cross-language perception of clear speech provides a window to understanding the phonetic bases for cross-linguistic typological patterns.

2pSCa4. Acoustic characteristics of clear speech and noise-adapted speech in preschoolers. Hoyoung Yi (Speech, Lang., and Hearing Sci., Texas Tech Univ. Health Sci. Ctr., 3601 4th St. Stop 6073, Lubbock, TX 79430-6073, hoyoung.yi@ttuhsc.edu) and Delaney DiCristofaro (Speech, Lang., and Hearing Sci., Texas Tech Univ. Health Sci. Ctr., Lubbock, TX)

It is well-known that adults can change their speaking styles to make them more intelligible for their listeners. However, we do not know how young children can achieve the ability to alter their speech. The study investigated if preschoolers can modify their speech responding to challenging communicative situations. Children performed "spot the difference" tasks (Baker and Hazan, 2011) with a communication partner to elicit conversational speech (CO), noise-adapted speech (NAS), and listener-oriented clear speech (CL) production. We examined the acoustic features of the three speaking styles in 28 preschoolers. Suprasegmental features of CO, NAS, and CL were analyzed, including F0 (Hz), F0 range (Hz), energy in the 1–3 kHz range(dB), and speech rate(syllables per second). Results showed conversation-to-NAS modifications for all four acoustic measures in all age groups. For conversation-to-CL modifications, all age groups decreased their speaking rate. The 5-year-olds increased their energy and pitch for CL. The 4-year-olds also increased the pitch for CL. The 3-year-olds did not change pitch and decreased energy for CL, which displays the opposite tendencies of adults. Preschoolers appear to be aware on some level of the challenging nature of communicative environments. They undergo ongoing development of speech modification for their listeners.

2pSCa5. Acoustic prominence of verbal instructions in Parkinson's disease. Thea Knowles (Communicative Sci. & Disord., Michigan State Univ., 1026 Red Cedar Rd., Oyer Speech & Hearing Bldg., Rm. 205, East Lansing, MI 48824-1212, theaknow@buffalo.edu), Meghan Clayards (Linguistics/Commun. Sci. & Disord., McGill Univ., Montreal, QC, Canada), and Nathan Cline (Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY)

Talkers signal important information to listeners in part by manipulating prosodic prominence. In English, this is typically achieved by producing more prominent words with greater durations, intensity, f0, and hyperarticulation, though this tendency may be reduced in talkers with motor speech disorders. In the present study, older adults with and without Parkinson's disease (PD) and dysarthria engaged in a simple instructional game activity in which they directed a research confederate to move around pictures on a board. Utterances were designed to manipulate prominence on stop-initial monosyllabic target words that differed in onset voicing and following vowel quality. Non-prominent words were repeated in the spoken instructions ("Move the pot above the chair, now move the [pot] above the box"), while prominent words contrasted with a minimal pair differing in onset voicing ("Move the pot above the chair, now move the [bot] above the chair"). Eleven people with PD and 11 age- and gender-matched controls participated. Overall, prominent words were longer, louder, and higher pitched, though control speakers modulated f0 to a greater extent than talkers with PD. Both groups modulated the amount of voicing during closure but not VOT as a means of increasing stop voicing contrasts in prominent words.

2pSCa6. Influence of noise on perception of different types of dysarthric speech. Emily Q. Wang (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina St., Ste. 1017, Chicago, IL 60612, emily_wang@rush.edu) and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chiago, IL)

Disturbances in the central nervous system may lead to changes in the speech muscle system, which impact muscle strength, tone, and coordination as well as movement speed, resulting in different types of dysarthria, each impacting respiration, phonation, articulation, resonance, and prosody differentially. Bilateral upper motor neuron damage leads to spastic dysarthria, cerebellar damage ataxic dysarthria, and damage to basal ganglia hypokinetic dysarthria. In estimating the severity of speech impairment, and measuring speech intelligibility, both reading and spontaneous speech samples are routinely used by speech-language pathologists. However, noise levels in acute and rehabilitation hospitals often exceed the World Health Organization guidelines (Busch-Vishniac *et al.*, 2005) which may impact the clinical outcomes. In this study, 100 graduate student clinicians in Speech-Language Pathology measured speech intelligibility and estimated the severity of speech impairment of speech samples produced by three types of dysarthric speakers: spastic, ataxic, and hypokinetic, in two different noise conditions (quiet, and hospital noise). The differential impact of noise on different types of dysarthria is examined, and its implication on clinical outcomes is discussed.

2pSCa7. Medically related speech: Impacts of age, familiarity, and noise on word recognition. Brian J. Puckett (Architectural Eng., Univ. of Nebraska–Lincoln, 1110 S 67th St., Omaha, NE 68182, puckett.brian@huskers.unl.edu), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska–Lincoln, Omaha, NE), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR), and Natalie Manley (Univ. of Nebraska Medical Ctr., Omaha, NE)

Verbal communication between patients and healthcare providers is an important piece of successful healthcare delivery. Less familiar medical terminology coupled with noisy hospital environments can lead to speech intelligibility challenges between healthcare providers and patients. Prior results

showed lower word recognition accuracy for lower familiarity and frequency words and in the presence of noise in younger adults (Bent *et al.*, 2022). The current study expands on the previous work with the inclusion of older adults and an additional noise masking condition. Online listening tests were conducted to study the impacts of word familiarity and background noise on speech perception in younger (age 18–35) and older adults (age 60–85). Participants were presented with a corpus of 160 medically related sentences with varying word familiarity and frequency characteristics, and word recognition accuracy was recorded. Sentences were presented in quiet, hospital noise, speech-shaped noise, and speech-shaped noise modulated by the hospital noise envelope, a novel condition from previous work. Results highlight a comparison of speech recognition performance between the two age groups. Effects of sentence familiarity/frequency rating and type of noise masker on speech intelligibility are discussed. [Work supported by the James S. McDonnell Foundation https://doi.org/10.37717/2021-3028.]

2pSCa8. Acoustic correlates of perceived anger in the Ferguson clear speech database. Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, 390 S 1530 E, Rm. 1218, Salt Lake City, UT 84112, liz.d.young@utah.edu), Shae D. Morgan (Otolaryngol., Univ. of Louisville, Louisville, KY), and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Previous work has shown that clear speech is perceived as sounding angry significantly more often than conversational speech (Morgan and Ferguson, 2017; Young, 2021). The current study examined possible acoustic correlates of perceived anger in clear speech. A principal component analysis was completed to guide selection of acoustic variables for analysis. The remaining six acoustic measures, fundamental frequency median, fundamental frequency variability, relative mid-frequency energy from 1000 to 3150 Hz, speaking rate, and degree of amplitude modulation at 2 Hz and 16 Hz envelopes, were analyzed using mixed effects regression. Results indicated that after controlling for speaking style, fundamental frequency approached significance but did not significantly predict perceived anger. However, speaking rate significantly predicted perceived anger after controlling for speaking style, with slower rates leading to an increase in perceived anger. Further, there was a significant interaction between speaking rate and speaking style, where slower speech was only perceived as angry in the clear speech style. Finally, a decrease in the degree of low-frequency amplitude modulation (i.e., equal emphasis on both content and function words) led to an increase in perceived anger after controlling for speaking style. The results have implications for counseling individuals in the use of clear speech.

2pSCa9. Contrast enhancement in clearly spoken Mandarin sibilants. Ivy Hauser (Linguist., Univ. of Texas Arlington, 701 Planetarium Pl., Box 19559—132 Hammond Hall, Arlington, TX 76019, ivy.hauser@uta.edu) and Xinwen Zhang (Linguist., Univ. of Texas Arlington, Arlington, TX)

Clearly spoken sibilants have been shown to exhibit temporal, spectral, and amplitudinal enhancement, though most existing work focuses on English. This study presents acoustics of clearly spoken sibilants in Mandarin, which exhibits a three-way place contrast between alveolar, retroflex, and alveopalatal sibilant fricatives. Sibilant-initial stimuli were elicited in a carrier phrase and were crossed according to: sibilant (alveolar, retroflex, alveopalatal), vowel context (a u), word frequency, number of syllables, and tone. The participants were eight native speakers of Mandarin with origins in mainland China. The first block was a baseline with no instruction on speech style. In the second block they were instructed to speak clearly ’as if talking to someone hard of hearing’. In the third block they were instructed to speak conversationally ’as if casually talking to a friend’. As expected, speakers enhanced duration and amplitude in the clear speech block and reduced duration and amplitude in the conversational block. Departing from previous work, spectral moments were not significantly enhanced in clear speech, which could be due to differences in type of task. However, second formants of vowels following alveopalatal sibilants were significantly higher in the clear speech block. This would be expected if clear speech is generally contrast enhancing, as the second formant of the following vowel is known to be the primary cue distinguishing the alveopalatal in perception and production.

2pSCa10. Impact of clear face masks on audiovisual speech intelligibility and subjective listening effort with normal hearing young adults. Kate E. Kocins (Psychol. & Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., Saint Louis, MO 63130, kkocins@wustl.edu), Kristin J. Van Engen, Violet A. Brown (Psychol. & Brain Sci., Washington Univ. in St. Louis, St. Louis, MO), Kate McClannahan (Dept. of Otolaryngol., Washington Univ. School of Medicine, St. Louis, MO), and Jonathan Peelle (Ctr. for Cognit. and Brain Health, Northeastern Univ., Boston, MA)

During the COVID-19 pandemic, facemask use made people suddenly aware of the importance of both visual information and acoustic clarity in speech perception. Masks with transparent panels can provide listeners with visual speech information, but such masks tend to introduce more acoustic distortion than other opaque masks. Thus, their value for improving speech recognition (at least for listeners with normal hearing) is not clear. In the current study we investigated the ability of three different clear face masks to improve speech intelligibility relative to an opaque surgical mask. Normal hearing young adult participants ($N = 180$) were presented with audiovisual recordings of 150 sentences from a female speaker in 5 different mask conditions (no mask, surgical mask, and three different clear masks) and at three different signal-to-noise ratios (no noise, –5 dB SNR, and −9 dB SNR). After each condition, participants also completed the NASA-TLX survey for subjective effort. Participants also completed an independent lipreading task. Preliminary data suggest decreased subjective effort and improved performance of the clear masks relative to the opaque mask at high SNR. This implies a potential for visual information to make up for decreased acoustic clarity, making clear masks a better choice in difficult listening conditions.

2pSCa11. Crosslinguistic influence in bilingual production of clear speech. Ye-Jee Jung (Purdue Univ., 215 Nimitz Dr., Apt. 4, West Lafayette, IN 47906, jung292@purdue.edu) and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

Clear speech, a listener-oriented speaking style adopted to mitigate communicative barriers, is often characterized by language-specific acoustic enhancement of phonological contrasts (Smiljanic and Bradlow, 2008; Cho *et al*., 2011). Previous work (e.g., Smiljanic and Bradlow, 2009) suggested a possibility that bilingual speakers’ L1 and L2 clear speech strategies could affect each other. The current study examined English and Korean clear speech produced by late Korean-English bilinguals ($n = 30$) living in the United States. Their clear speech produced in each of the two languages was compared with clear speech produced by either English ($n = 20$) or Korean ($n = 20$) monolinguals. Of specific interest was how English and Korean stop contrasts were realized and enhanced in clear speech with respect to the use of VOT and onset F0. The bilinguals enhanced the English voicing contrast in the same manner as English monolinguals, via asymmetrical lengthening of voiceless stops’ VOT. In Korean clear speech, bilinguals also lengthened VOT of the aspirated and lenis (long-lag) stops, whereas Korean monolinguals did not exhibit the same pattern. These findings suggest that Korean clear speech strategies produced by the bilinguals could be affected by their L2, due to their extensive exposure to the language.

2pSCa12. Clear speech improves word segmentation in quiet and in noise: Evidence from visual-world eye-tracking. Zhe-chen Guo (Dept. of Linguist., Univ. of Texas at Austin, 307 E 31st St. Apt. 105, Austin, TX 78705, y9024131@gmail.com) and Rajka Smiljanic (Dept. of Linguist., Univ. of Texas at Austin, Austin, TX)

Listener-oriented hyperarticulated clear speech facilitates linguistic processing and cognitive functioning associated with speech perception under various listening conditions. Using the visual-world eye-tracking paradigm, we investigated whether clear speech also aids speech segmentation, or the discovery of word boundaries, and examined the dynamic time course of its effect. Native American English speakers ($N = 77$) heard sentences in which the target word (e.g., *ham*) was temporarily ambiguous with a longer

unintended competitor (e.g., *hamster*) across a word boundary (e.g., *She saw the ham starting…*) while viewing images depicting the target, competitor, and unrelated distractors. Clear and conversational sentences were presented in quiet or in speech-shaped noise at +3 dB signal-to-noise ratio. Analysis of eye fixations to the images over time revealed that compared with conversational speech, clear speech facilitated the disambiguation of the target from the competitor even before the disambiguation point was reached. The facilitation was found in both listening conditions but was relatively delayed in noise. These findings suggest that speaking clearly improves word segmentation and reduces lexical competition especially in optimal listening conditions. The speech segmentation facilitation may partly underlie the clear speech benefits observed for other signal-dependent and relatively signal-independent linguistic and cognitive processes.

2pSCa13. High spectral covariation between frequency channels contributes to clear speech intelligibility. Fernando Llanos (Linguist., Univ. of Texas at Austin, Austin, TX, fllanos@utexas.edu), Kirsten Meemann (Linguist., Univ. of Texas at Austin, Austin, TX), Rajka Smiljanic (Univ. of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Speech signals are acoustically redundant, which could explain why sentence intelligibility is fairly robust even when sentences are acoustically degraded. We investigated the contributions to sentence intelligibility of clear speech redundancy encoded as patterns of spectrotemporal covariation between frequency channels. Participants ($N = 16$) transcribed 120 clear-speech English sentences acoustically degraded to 5, 8, or 15 frequency bands derived from an ERB-scaled filter bank. Before the acoustic degradation, each sentence was expressed as a linear combination of principal component eigenvectors representing different patterns of covariation between channels. Half of the sentences preserved the channels providing larger score magnitudes for the eigenvector accounting for *more* spectral covariance (*high-covariance* condition). These channels represented the spectral covariation patterns that were more dominant in each sentence. The other half of the sentences preserved the bands conveying larger score magnitudes for the eigenvector accounting for*less* spectral covariance (*low-covariance* condition). These bands represented the spectral covariation patterns that were less dominant. Participants yielded significantly better transcription accuracy in the high-covariance condition (mixed-effects, $ps < 0.0021$). Critically, accuracy in this condition was higher than 56% on average for as few as 5 bands. These findings indicate that clear speech intelligibility is supported by patterns of spectral covariation between frequency bands.

2pSCa14. Abstract withdrawn.

2pSCa15. Speech intelligibility and listening effort: A dual-task paradigm for accented speech perception. Mel Mallard (Psychol. and Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, m.mallard.wustl@gmail.com) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Unfamiliar accents can make speech communication difficult, both by reducing speech intelligibility and by increasing the effort listeners must put forth to understand speech. In general, intelligibility refers to the proportion of words that a listener can correctly identify (100%, 50%, etc.), while listening effort refers to the cognitive resources that must be devoted to a difficult listening task. Generally, lower intelligibility is related to higher effort. However, it is now apparent that these two constructs, while related, are independent: for example, two 100% intelligible speakers may elicit different amounts of effort. To better characterize the relationship between intelligibility and effort, this study presents speakers of four intelligibility levels (one L1 English speaker, and three Mandarin-accented English speakers) within a dual-task paradigm (featuring a vibrotactile secondary task) to measure listening effort. We hypothesize an inverse relationship between intelligibility and effort, but the shape and slope of this relationship are of primary interest. These results will begin to illuminate the relationship between intelligibility and effort across a more continuous range of accent intelligibility. Data collection is ongoing. Results for the Mandarin Chinese accent will be presented. Preliminary results for the study's second iteration (using a Korean accent) will also be presented.

2pSCa16. Effect of a portable sound-booth on teachers' voice acoustic parameters. Lady Catherine Cantor-Cutiva (Michigan State Univ., Carrera 30 Calle 45, Ciudadela Universitaria, Bogota 110110, Colombia, lccantorc@unal.edu.co) and Eric J. Hunter (Com Sci. & Disord., Michigan State Univ., East Lansing, MI)

Background: College professors expose themselves to different vocal demands during teaching. During the COVID-19 pandemic, college professors were exposed to extra working loads associated with working from home, with less-than-optimal acoustic conditions. This is the reason why it was needed to think of easy-to-implement and low-cost solutions for online teaching from home. Objective: To evaluate the effect of an insulating cabin prototype on the voice acoustic parameters of college professors during homeworking. Methodology: Longitudinal study with the participation of 32 (thirty-two) full-time college professors. Participants were asked to fill out a questionnaire before reading a paragraph of around 30 seconds of duration with and without the portable sound booth. A type I sound level meter was used to measure the background noise in professors' teaching environments with and without the cabin prototype. Results: Voice acoustic parameters changed when voice production was performed with the portable sound booth. Therefore, this low-cost solution is a good alternative to prevent vocal effort and vocal fatigue among teachers.

Session 2pSCb

Speech Communication: Speech in Different Listening Conditions (Poster Session)

Sheyenne Fishero, Chair
Department of Linguistics, University of Kansas, 1541 Lilac Ln, Blake Hall Room 427, Lawrence, KS 66045

All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd numbered abstracts will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even numbered abstracts will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

2pSCb1. Perceptual effects of formant enhancement with the factors of phonetic type, listening conditions, and language experience of listeners. Mingshuang Li (Dept. of Commun. Disord. and Sci., California State Univ., Northridge, Northridge, CA 91330, mingshuang.li@csun.edu) and Chang Liu (Univ. of Texas at Austin, Austin, TX)

The second formant (F2) enhancement is a technique that aims to improve speech perception in adverse noise by amplifying the F2 of speech signals. The current study was to investigate whether F2 enhancement would improve speech identification with the factors of phonetic type (e.g., vowel and consonant), listening conditions (e.g., speech and nonspeech noise at moderately and challenging SNRs), and language experience of listeners (e.g., native and nonnative listeners), and whether the amount of perceptual benefit was dependent on these factors. Two groups of participants, English native and nonnative listeners, were recruited in this study. Identification of English vowels and consonants with and without F2 enhancement were measured in quiet, long-term speech shaped noise (LTSSN) and six-talker babble (6-TB) at the signal-to-noise ratios (SNRs) of −10 dB and −15 dB. Overall, significant improvements from F2 enhancement were found in both vowel and consonant identification for both native and nonnative listeners in various listening conditions. Furthermore, greater improvement was found at the SNR of -15 dB than at the SNR of −10 dB, as well as for nonnative listeners than native listeners in vowel identification. Meanwhile, the amount of benefit was generally comparable speech and nonspeech noise. These results indicate that F2 enhancement could improve phonetic identification in noise for native and nonnative listeners, showing a potential as a speech enhancement algorithm in challenging noise.

2pSCb2. Comparing the effects of target and masker voice familiarity on children's speech-in-speech recognition. Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, maryflah@illinois.edu)

Familiarity with a target voice has been shown to aid spoken word recognition for both children and adults. Familiarity effects of a target voice are not well understood, and even less is known about the impacts of masker voice familiarity. The present study investigated long-term target and masker voice familiarity effects on children's speech-in-speech recognition by using the child's own mother's voice as stimuli. Open-set sentence recognition thresholds (SRTs) were measured adaptively in a two-talker speech masker. Twenty children were tested in three conditions, hearing their mother's voice and two unfamiliar female voices: (1) Familiar Target/Unfamiliar Masker; (2) Familiar Masker/Unfamiliar Target; and (3) Unfamiliar Target/Unfamiliar Masker. Condition 1 measured effects of target familiarity; Condition 2 examined effects of masker familiarity; Condition 3 served as baseline. Results showed SRTs were significantly better when target speech was spoken by a familiar talker (the mother). When background speech was spoken by the familiar talker, performance was worse relative to an unfamiliar masker. This suggests that children do benefit from long-term voice familiarity with the target but not the masker speech. Thus, voice familiarity may impact children's ability to attend to the target speech but does not always improve segregation.

2pSCb3. The perception of acoustically distorted speech produced with face masks in multilingual multi-talker environments. Faith Chiu (Univ. of Glasgow, 12 University Gardens, University Ave., Glasgow G12 8QQ, United Kingdom, chiufaith@gmail.com), Laura Bartoseviciute (Univ. of Essex, Colchester, United Kingdom), Albert Lee (The Education Univ. of Hong Kong, Tai Po, Hong Kong), and Yujia Yao (Univ. of Essex, Colchester, United Kingdom)

This paper examines the perception of speech produced with face masks in multilingual multi-talker environments. Three groups of participants varying in language background listened to and reported English target sentences produced with or without a face mask in the presence of a competing English or Lithuanian talker. Listeners were monolingual native speakers of English, second language (L2) English speakers with Lithuanian as first language (L1), and L2 English speakers with L1 Mandarin Chinese. In addition, Lithuanian speakers also completed the same experiment but with Lithuanian targets. Results indicate that participants were more accurate with perceiving target sentences in their L1. Targets produced with a face mask were less accurately perceived across all groups regardless of listening in L1 or L2. In general, a competing talker in a language which matches the target (English distractor on English target) had a more detrimental effect on perception accuracy than a mismatched one (Lithuanian distractor on English target). Exceptionally, only when Lithuanian participants—with both English and Lithuanian knowledge – listened in their L1 was there no added challenge from matching distractor and target language. We conclude that acoustic distortions from face masks present an across-the-board difficulty while linguistic knowledge can reduce distraction from competing talkers.

2pSCb4. Acoustical impacts of blindness on speech production strategies linked to Lombard speech and audiovisual interaction. Pamela Trudeau-Fisette, Camille Vidou, Cristina Uribe (Linguist., UQAM, Montreal, QC, Canada), and Lucie MENARD (Linguist., UQAM, Pavillon J.-A. De sève, DS-4423, 320, Sainte-Catherine Est, Montréal, QC H2X 1L7, Canada, menard.lucie@uqam.ca)

This paper investigates the acoustical correlates of Lombard speech and audiovisual interaction, in blind and sighted adults. Ten blind and nine sighted adults were recorded while producing repetitions of the French vowels /i/, /u/, /a/ in a "pVp" context. Sentences were produced under four conditions, varying in terms of interaction type (whether the speaker could be heard and seen—audiovisual interaction—or only heard— auditory interaction—by the interlocutor) and presence/absence of noise. Inter-vocalic and intra-vocalic formant dispersion as well as fundamental frequency (F0), intensity and vowel duration values were measured. Results of linear mixed effects models showed that blind speakers increased their inter-vocalic

distances in noise, but decreased it when they were told they were seen by the interlocutor (audiovisual interaction). Blind speakers decreased F0 in the audiovisual conditions while sighted speakers increased F0 in noise. Finally, sighted speakers produced louder vowels in the noisy conditions and in the auditory interactive conditions. This pattern was only found for unrounded vowels in blind speakers. These results highlight the role of vision on speech production and show that sighted speakers likely use active strategies to enhance visual cues in audiovisual interactions, compared to auditory only interactions.

2pSCb5. Misperception in Northern Vietnamese Tones due to the Lombard effect. Giang Le (Linguist., Univ. of Illinois Urbana-Champaign, 707 S Mathews Ave., Urbana, IL 61801, gianghl2@illinois.edu) and Yan Tang (Linguist., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Studies examining Lombard speech have found that intelligibility often increases due to acoustic changes such as elevated intensity, increased F0 and F0 range, and elongated sonorous segments. By previously studying the production of the eight lexical tones in Northern Vietnamese in noise, we have also observed that all tones experienced differing degrees of the Lombard effect. For instance, tone B2 (mid-falling with creakiness) showed resistance to increased glottal regularity, while tone C2 (mid-rising with creakiness) exhibited a greater convergence towards regular phonation. The current study conducted a perception experiment to investigate to what extent those acoustic adaptations due to hyper-articulation impact listeners' perception of tonal identity. Native Vietnamese listeners identified the tones produced in quiet or in noise of 78 or 90 dB SPL by a male and a female speaker. The results show that the creaky tones B2 and C2 had the lowest identification accuracy while the rising modal tone B1 received the highest. Tone C2 in particular was most likely to be identified as the high tones A1 or B1 when originally uttered in noise. The findings call into question whether hyper-articulation always leads to better speech intelligibility, especially for tonal languages.

2pSCb6. Perception error variation in masking contexts. Aarya N. Menon (Linguist., Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Ecuador, anmenon@ualberta.ca) and Benjamin V. Tucker (Linguist., Univ. of AB, Edmonton, AB, Canada)

Masking noise is an important tool in speech and hearing experiments. It provides a means to simulate real world conditions and provide misperceptions that are as naturally elicited as possible. However, the exact characteristics of the variable approach to this parameter is poorly understood. The present investigation explores the types and variability of errors across different noise conditions. The English Consistent Confusion Corpus [Marxer *et al.*, *JASA* **140**, EL458–EL463 (2016)], which provides data on consistently reproducible mispercepts, was used for this investigation. The dataset contains tokens of words in multiple masking conditions: speech shaped noise; three talker babble modulated noise; and four-talker natural babble, correlated with signal to noise ratios that bias a listener towards a specific mispercept. This dataset contains 3200 individual word tokens from four different speakers with 15 listeners. The interaction between types and variability of misperception with the different types of masking noise is explored. As well, the suitability of the different noise conditions for speech perception experiments is discussed.

2pSCb7. Speech-in-speech perception: The role of F0, rate, and rhythm. Sheyenne Fishero (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall Rm. 427, Lawrence, KS 66045, sfishero@ku.edu), Allard Jongman, and Joan Sereno (Dept. of Linguist, Univ. of Kansas, Kansas City, KS)

Speech perception typically takes place against a background of other speech or noise. The present study aims to investigate the effectiveness of segregating speech streams within a competing speech signal, examining whether cues such as pitch that typically denote a difference in talker behave in the same way as cues such as speaking rate that typically do not denote the presence of a new talker. Native English speakers listened to English target speech within English two-talker babble of a similar or different pitch and/or a similar or different speaking rate to identify whether mismatched properties between target speech and masker babble improve speech segregation. Additionally, Dutch and French masker babble was tested to identify whether an unknown language masker improves speech segregation capacity and whether the rhythm patterns of the unknown language modulate the improvement. Individual differences in selective attention were also measured to determine whether they can predict speech segregation ability. This study aims to increase the understanding of speech perception in a more ecologically valid context and to identify whether there is a link between a cue's potential to denote a new speaker and its ability to aid in speech segregation during competing speech perception.

2pSCb8. Examining acoustics and intelligibility of people with dysarthria in communicative interactions. Elizabeth Krajewski (Commun. Sci. and Disord., The 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., The Penn State Univ., University Park, PA)

A comprehensive understanding of the speech capabilities of speakers with dysarthria requires an examination of speech produced in true interactions. Intelligibility in such interactions may reflect joint contributions of speakers and listeners (Olmstead *et al.*, 2020). In this study, nine people with dysarthria secondary to amyotrophic lateral sclerosis (ALS) participated in an interactive word-matching task with nine typical individuals. In experiment one, we examined speech produced by the people with dysarthria in the interactive task and compared it to speech produced in two non-interactive tasks (baseline and clear speech). Specifically, we measured F1, F2, and vowel duration of the vowels /æ, ɛ, ɪ, and i/. We found task-related differences such that acoustic contrast was greater in the interactive task than baseline. Notably, speech in the interactive task was similar to clear speech in its spectral but not temporal patterns. In experiment two, we obtained perceptual judgements of these productions from 54 naïve listeners. We found a concordant pattern: the vowels from the matching task were easier for listeners to identify than baseline. Overall, results of these studies indicate that speakers with dysarthria can make speech adjustments in communicative tasks that are beneficial for listeners.

2pSCb9. Vowel focused training improves listener transcription of dysarthric speech. Elizabeth Krajewski (Commun. Sci. and Disord., Penn State Univ., 308 Ford Bldg., University Park, PA 16802, ezr16@psu.edu) and Anne J. Olmstead (Commun. Sci. and Disord., The Penn State Univ., University Park, PA)

Typical listeners can adjust to the speech of individuals with dysarthria through the process of perceptual learning. Past research has demonstrated that listeners improve in their recognition of both segments and connected speech produced by people with dysarthria. The mechanisms underlying learning of dysarthric speech are still uncertain, though it has been suggested that exposure allows listeners to tune into the segmental characteristics of speech. In the current study, we test this hypothesis by training listeners to identify vowels spoken by an individual with mild dysarthria. We employed a pre-test/post-test design where we first tested listeners on (1) vowel recognition and (2) transcription accuracy of connected speech, trained them, and then tested them again. We found that listeners who were trained on dysarthric vowels demonstrated greater improvements in vowel identification than those in the control condition. Likewise, the listeners who underwent training showed greater improvements in transcription accuracy than those in the control condition. However, the training advantage did not generalize to unfamiliar phrases. Overall, it appears that listeners are able to tune into the segmental characteristics of dysarthric speech after a short training session, which improves their recognition of longer connected speech.

2pSCb10. Voice acoustics and effort of three different communication scenarios presented in an anechoic baseline. Mark Berardi (Univ. Hospital Bonn, Venusberg-Campus 1, Bonn 53127, Germany, mberardi@uni-bonn.de) and Eric J. Hunter (Com Sci. & Disord., Michigan State Univ., East Lansing, MI)

While vocal pathology can elevate vocal effort, communication situations also have an effect. Previous work on vocal effort and communication

situations have relied on listener ratings of effort rather than talker-centric vocal effort. Towards that end, in this study, talker-centric vocal effort was related to different communication demands. Additionally, vocal performance was measured to investigate potential relationships between effort, production, and communication demand. In an anechoic chamber, participants described to an interlocutor various maps. The communication demands were communication distance (1, 2, 4 m), loudness goal (54, 60, 66 dB), and background noise (53, 62, 71 dBA). Talkers rated their vocal effort after each task with a Borg CR100 scale and were recorded with a calibrated head-worn microphone. There were significant increases in ratings of vocal effort, fundamental frequency, intensity, and cepstral-peak prominence (CPP) from the control condition to the extremes of the distance and loudness goal vocal demands which are consistent with previously observed Lombard responses. This study shows that communication demands that exceed typical conditions (such as talking at 4 m) result in increases in vocal effort and a Lombard-like response. Therefore, considerations to potential communication barriers are important for the reduction of vocal effort.

2pSCb11. Individual differences in older adults' perception of speech in hospital noise. Shelemiah Crockett (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, tbent@iu.edu), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE), Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR), and Natalie Manley (Dept. of Internal Medicine, Div. of Geriatrics, Gerontology and Palliative Medicine, Univ. of Nebraska Medical Ctr., Omaha, NE)

Hospital noise is associated with negative patient outcomes and staff performance issues. One understudied challenge is how hospital noise impacts healthcare provider—patient communication, particularly for information exchanges including less familiar or frequent medical terminology. This issue is of particular importance for older adults who are more likely to be hospitalized and also have greater challenges with speech in noise compared to younger adults. Here, older adults' perception of sentences containing medical terminology varying in word frequency and familiarity was tested in three listening conditions: quiet, speech-shaped noise, and hospital noise. Additionally, a range of individual factors that may predict performance were assessed. Preliminary results suggest an interaction between listening condition and lexical characteristics with larger impacts of both lexical frequency and familiarity in the noise conditions than quiet. The word frequency effect was larger in hospital noise than speech-shaped noise. Follow-up analyses will assess how age, hearing thresholds, cognitive decline, vocabulary size, and experience with hospital environments and medical terminology contribute to individual differences in speech intelligibility across the listening conditions and sentence types. [Work supported by the IU Institute for Advanced Study and the James S. McDonnell Foundation https://doi.org/10.37717/2021-3028.]

2pSCb12. Japanese pitch-accent identification accuracy by children with autism spectrum disorder. Yasuaki Shinohara (Faculty of Commerce, Waseda Univ., 1-6-1 Nishiwaseda, Shinjuku-ku, Tokyo 169-8050, Japan, y.shinohara@waseda.jp), Mariko Uchida, and Tomoko Matsui (Faculty of Letters, Chuo Univ., Tokyo, Japan)

A previous study demonstrated when Japanese speakers hear sine-wave speech carrying no fundamental frequency information, they rely on the first formant frequency for their pitch-accent perception, resulting in poor identification accuracy. However, when sine-wave speech is noise-vocoded, each formant becomes less audible, and Japanese speakers use other acoustic cues (e.g., duration and intensity), leading to better identification [Y. Shinohara, "Japanese pitch-accent perception of noise-vocoded sine-wave speech," The 183rd Meeting of ASA, Nashville, TN (2022)]. The present study examined pitch-accent perception by children with autism spectrum disorder (ASD) for the same three stimulus types: natural recordings, sine-wave speech, and noise-vocoded sine-wave speech. Children with ASD often have enhanced auditory pitch perception for non-speech stimuli. However, since the present study used a pitch-accent word identification task, it was hypothesized that Japanese-speaking children with ASD would show less identification accuracy for all three stimulus types compared to age-matched typically developing children. Our preliminary results showed a significant difference in identification accuracy between the two groups. A detailed analysis of the results and the acoustic features of each stimulus type will be discussed.

2pSCb13. Voicing-, voiceless-, and non-glimpses in speech intelligibility prediction. Yinglun Sun (Linguist., Univ. of Illinois at Urbana-Champaign, 102 N Gregory St. Apt. 7, Urbana, IL 61801, yinglun2@illinois.edu) and Yan Tang (Linguist., Univ. of Illinois Urbana-Champaign, Urbana, IL)

The number of speech spectro-temporal (S-T) regions escaping from noise masking, known as "glimpses," is proportional to speech intelligibility in noise. Previous studies have demonstrated that intelligibility can be estimated by calculating the glimpse proportion (GP). More recent evidence revealed that the contribution of glimpses to intelligibility differs in the energy level of the glimpsed regions, and that even non-glimpsed regions play a non-negligible role in speech perception in noise. This study incorporated the voicing-viceless information in estimating intelligibility using glimpses. Before computing the GP, the counts of raw glimpsed regions or those with energy above the mean noise level were weighted according to the voicing-voiceless status of a frame where the glimpses were detected. Evaluated using speech signals processed to have thirteen glimpse compositions in both temporally stationary and fluctuating noise maskers, the linear correlation between model predictions and listeners' word recognition rates increased from 0.76 to 0.80 for weighted GP, and from 0.89 to 0.92 for weighted high-energy GP. Further taking the contribution from non-glimpsed regions into account in the model improved the correlation to 0.95, suggesting that intelligibility in noise can be better predicted when the contributions of different speech regions are finely modelled.

2pSCb14. Relationship between speech intelligibility and comprehension in the presence of poor acoustics and poor voice quality. Silvia Murgia (Speech and Hearing Sci., Univ. of Illinois–Urbana Champaign, 901 South Sixth St., Champaign, IL 61820, smurgia2@illinois.edu), Bisma N. Choudhry, 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)

Noisy classrooms with poor acoustics decrease children's speech understanding and comprehension. The signal quality is also often degraded by voice disorders that have a prevalence of 60% among teachers. Emerging evidence demonstrates that cognitive factors such as working memory (WM), attention, and inhibitory control (IC) also mediate children's ability to recognize speech in noise. Speech intelligibility and comprehension tests were performed in a sound-proof booth with xx normal-hearing elementary students. The vocal material was recorded by an actor with normal and mimicked dysphonic voice. Babble noise was added to obtain 2 signal-to-noise (SNR) ratios at 0 and −12 dB. The test was administered using Qualtrics to collect intelligibility and comprehension scores, and measure response time to evaluate listening effort. Children's WM and IC were also evaluated to explain individual differences. Results showed a statistically significant decrease in performance and increase in response time for the highest SNR and the dysphonic voice with a more detrimental effect on comprehension. This research determined how listening comprehension interacts with speech recognition and cognitive factors in children, providing pilot data to examine the variability of students' learning experience in classrooms when auditory input is degraded by poor acoustics and poor voice quality.

2pSCb15. Listener bias and intelligibility of unfamiliar varieties of a language. Kenzo S. Kimura (Communicative Sci. and Disord., New York Univ., 55 E 10th St., Apt. 1505, New York, NY 10003, kimura@nyu.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Speech intelligibility depends on a combination of the speech itself and of the listener. In terms of speech, unfamiliar varieties (such as foreign-accented speech or speech produced in an unfamiliar dialect) are generally less intelligible than speech produced in a variety that matches that of the

listener. In terms of the listener, studies have found that listener experience affects intelligibility of unfamiliar varieties of a language. The current study explores the extent to which a listener's biases toward globalization and immigration predict speech intelligibility. High and low predictability sentences from three men (US English, Scottish English, Mandarin-accented English) were mixed with background noise. Fifty L1 US English listeners first typed the final word of each sentence and were then asked to answer questions related to their views on globalism and nationalism. As expected, listeners were more accurate in the high compared to low predictability sentences. Surprisingly, the Mandarin-accented speaker was the most intelligible. A follow-up study with duration-normalized samples found the same result. Contrary to expectations, listener responses to the globalist-nationalist survey had no impact on intelligibility. This may suggest that difficulty understanding unfamiliar varieties of a language is not related to listener attitudes.

2pSCb16. The Children's English/Spanish Speech Recognition Test (ChEgSS): Feasibility, Reliability, and Normative data. Lori Leibold (Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., 555 N 30th St., Omaha, NE 68131, lori.leibold@boystown.org), Emily Buss (Univ. of North Carolina, Chapel Hill, NC), Margaret Miller (Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Tiana Cowan (Boys Town National Res. Hospital, Omaha, NE), Ryan McCreery (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Jacob Oleson (Univ. of Iowa, Iowa City, IA), Barbara Rodriguez (Univ. of New Mexico, Albuquerque, NM), and Lauren Calandruccio (Case Western Reserve Univ., Cleveland, OH)

The Children's English/Spanish Speech Recognition Test (ChEgSS) is a computer-based tool for assessing closed-set word recognition in either speech-shaped noise or two-talker speech. Instructions and stimuli are recorded in both languages, and listeners respond by pointing to the picture depicting the word that they heard. Preliminary clinical feasibility and reliability data will be presented, along with age norms for children (4.1–17.0 yrs) who are either English monolinguals ($n = 89$) or Spanish/English bilinguals ($n = 83$). Results obtained with the two maskers will be evaluated with respect to age, language group, and receptive vocabulary. SRTs in English were higher in two-talker speech than speech-shaped noise and were 1.7 dB lower for monolinguals than bilinguals for both maskers. SRTs were lower in English than Spanish for bilinguals in noise but not significantly different in speech. Consistent with prior research, SRTs in both languages improved with age; effects of age were more pronounced and prolonged in two-talker speech than in speech-shaped noise. The mean discrepancy between replicate estimates of SRT in English were 1.8 dB in noise and 3.1 in two-talker speech. Overall, these results suggest ChEgSS may be a valuable tool for clinical assessment of masked speech perception in children.

2pSCb17. Effects of age and musical expertise on perception of speech in speech maskers in adults. Eleanor Harding (Otorhinolaryngology, UMCG, Groningen, the Netherlands, e.e.harding@rug.nl), Laura Rachman, Ryan Gray, Stefan Smeenk (Otorhinolaryngology, UMCG, Groningen, the Netherlands), Anastasios Sarampalis (Psych., Univ. of Groningen, Groningen, the Netherlands), Etienne Gaudrain (CNRS, Lyon, France), and Deniz Başkent (Otorhinolaryngology, UMCG, Groningen, the Netherlands)

Perceiving "cocktail party" speech, or speech-on-speech (SoS), requires perceptual mechanisms such as segregating target from masking speech using voice cues, and on cognitive mechanisms such as selective attention and inhibition. Both aging and musical expertise have been shown to affect these mechanisms. Voice cues that help distinguish different speakers include mean fundamental frequency (F0), related to voice pitch, and the vocal-tract length (VTL), related to speaker size. Some studies reported older adults' decreased sensitivity to F0 differences, possibly affecting their ability to discriminate speakers. Furthermore, age-related cognitive changes may lead to difficulties in attention direction and inhibition. Compared to non-musicians, musicians are reported to show enhanced processing of acoustic features such as F0, as well as enhanced cognitive abilities such as auditory attention skills and working memory. While this intuitively could lead to a musician advantage for SoS, reports of musicians outperforming non-musicians are inconsistent across both younger and older adults. Differences across previous SoS paradigms have made comparison of musicianship advantages in SoS in younger and older adults—and underlying mechanisms—difficult. Therefore, we investigated the extent to which older compared to younger adults benefit from either musical expertise or F0 and VTL differences between target and masker voices.

2pSCb18. Effects of frequency-to-place mismatch in children listening to simualted unilateral and bilateral cochlear implant configurations. Julianna Gross (Hearing and Speech Sci., Univ. of Maryland, 0100 Samuel J. LeFrak Hall, College Park, MD 20742, jgross11@umd.edu), Margaret Cychosz (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), and Qian-Jie Fu (David Geffen School of Medicine at UCLA, Univ. of California, Los Angeles, Los Angeles, CA)

Binaural cues are important for spatial speech recognition. While binaural cues can be partly restored by bilateral cochlear implants or a unilateral cochlear implant in patients with single-sided deafness, the benefits may be limited by frequency-to-place mismatch in both cases. Previous studies reported worse recognition of spatially separated speech as frequency-to-place mismatch increased in typical-hearing adults listening to acoustic simulations of bilateral cochlear implants (Thomas *et al.*, 2022). This work also revealed that children were more sensitive to the spectral degradation and less able to use spatial cues for spatial speech recognition. The present study investigates the effects of frequency-to-place mismatch on binaural fusion in typical hearing children (7–9 years old) listening to acoustic simulations of bi- and unilateral cochlear implant configurations. The stimuli in the left ear were either unprocessed or processed by a 16-channel sinewave vocoder with a fixed insertion depth while the stimuli in the right ear were processed by 16-channel sinewave vocoders with varying insertion depth. Speech recognition threshold results from $N = 10$ children show an effect of interaural insertion depth mismatch upon speech recognition thresholds, with no differences by age. Comparisons with adult data will likewise be reported.

2pSCb19. Fearless Steps APOLLO: Challenges in keyword spotting and topic detection for naturalistic audio streams. Aditya Joglekar (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas-Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, aditya.joglekar@utdallas.edu), Ivan Lopez-Espejo (Dept. of Electron. Systems, Aalborg Univ., Richardson, TX), and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

Fearless Steps (FS) APOLLO is a +50,000 hr audio resource established by CRSS-UTDallas capturing all communications between NASA-MCC personnel, backroom staff, and Astronauts across manned Apollo Missions. Such a massive audio resource without metadata/unlabeled corpus provides limited benefit for communities outside Speech-and-Language Technology (SLT). Supplementing this audio with rich metadata developed using robust automated mechanisms to transcribe and highlight naturalistic communications can facilitate open research opportunities for SLT, speech sciences, education, and historical archival communities. In this study, we focus on customizing keyword spotting (KWS) and topic detection systems as an initial step towards conversational understanding. Extensive research in automatic speech recognition (ASR), speech activity, and speaker diarization using manually transcribed 125 h FS Challenge corpus has demonstrated the need for robust domain-specific model development. A major challenge in training KWS systems and topic detection models is the availability of word-level annotations. Forced alignment schemes evaluated using state-of-the-art ASR show significant degradation in segmentation performance. This study explores challenges in extracting accurate keyword segments using existing sentence-level transcriptions and proposes domain-specific KWS-based solutions to detect conversational topics in audio streams. [Work Sponsored by NSF via Grant No. 2016725 and EU's Horizon 2021 R&I Program under MSC Grant No. 101062614.]

2pSCb20. Portable hearing laboaratory for developping and evaluating novel hearing aid algorithms. Chaslav Pavlovic (BatAndCat Sound Labs, 602 Hawthorne Ave., Palo Alto, CA 94301, chas@batandcat.com), Reza Kassayan (BatAndCat Sound Labs, Atherton, CA), Nicholas Michael (BatAndCat Sound Labs, Manistee, MI), Hendrik Kayser, and Volker Hohmann (Univ. of Oldenbutgh, Oldenburgh, Germany)

We report here on the further development of our portable hearing aid research platform achived in Phase III NIDCD project R44DC016247. We also report on a large number of academic studies using the platform around the world. The project builds on the openMHA distribution for ARM processors included in an optimized Linux system developed in our other project R01DC015429. Thus we provide a complete wearable software-hardware fast hearing aid device needed for development of new innovative solutions for assisted hearing. We refer to this device as the PHL, short for Portable Hearing Laboratory. Throught the project we have followed the requirement that the platform should be "acoustically perfect." That is able to transmit and process signals without audible distortion or noise beyond what is required to compensate for hearing loss. Any new sophisticated algorithms (such as those based on the AI) if run on a platform of inferior quality will adapt for that inferior structire and fail to capitalize on the reach processing capabilities of the modern hearing aids. We also report on a family of devices that can be used either with PHL or with other processors to provide supperior sound; such as bunaural BTE multimicrophone headsets. Last but not least we will present new devices connecting these auxiliary devices to powefull desktop computers for laboratory studies.

2pSCb21. End-to-end child-adult speech diarization in naturalistic conditions of preschool classrooms. Prasanna V. Kothalkar (Ctr. for Robust Speech Systems, Univ. of Texas at Dallas, 800 W Campbell Rd., Erik Jonsson School of Eng., EC32 P.O. Box 830688, Richardson, TX 75080-3021, Prasanna.Kothalkar@utdallas.edu), Dwight Irvin, Jay Buzhardt (Juniper Gardens Childrens Project, Univ. of Kansas, Kansas City, KS), and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

Speech and language development are early indicators of overall analytical and learning ability in pre-school children. Early childhood researchers are interested in analyzing naturalistic versus controlled lab recordings to assess both quality and quantity of such communication interactions between children and adults/teachers. Unfortunately, present-day speech technologies are not capable of addressing the wide dynamic scenario of early childhood classroom settings. Due to diversity of acoustic events/conditions in daylong audio streams, automated speaker diarization technology is limited and must be advanced to address this challenging domain for audio segmentation and meta-data information extraction. We investigate a Deep Learning-based diarization solution for segmenting classroom interactions of 3–5 year-old children engaging with teachers. Here, the focus is on speaker-label diarization which classifies speech segments as belonging to either Adults or Children, partitioned across multiple classrooms. Our proposed ECAPA-TDNN model achieves a best F1-score of 65.5% on data from two classrooms, based on open dev and test sets for each classroom. Also, F1-scores for individual speaker labels provide a breakdown of performance across naturalistic child classroom engagement. The study demonstrates the prospects of addressing educational assessment needs through communication audio stream analysis, while maintaining both security and privacy of all children and adults.

Session 2pSP

Signal Processing in Acoustics and Computational Acoustics: Feature Extraction, Dimensionality Reduction, and Learning in Ocean Acoustics

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Chair's Introduction—1:00

Invited Paper

1:05

2pSP1. Neural acoustic fields. Paris Smaragdis (Univ. of Illinois Urbana-Champaign, 201 N. Goodwin Ave., Urbana, IL, paris@illinois.edu)

We present a way to compactly represent acoustic transfer functions using a small, yet flexible, parametric representation. We show that we can use a neural network as a "soft" table lookup and train it to produce the value of transfer functions at arbitrary points in space and time. Doing so allows us to interpolate and produce unseen data, and to represent acoustic environments using a remarkably compact representation. Due to this representation being differentiable, this opens up multiple opportunities to employ such models within more sophisticated audio processing systems.

Contributed Papers

1:25

2pSP2. Machine learning of acoustic propagation models for sound aware autonomous systems. Ryan A. McCarthy (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 103 S. Capitol St., Iowa City, IA 52242, r1mccarthy@ucsd.edu), Sophia Merrifield (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), and Eric Terrill (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Opportunities exist for autonomous underwater vehicles to optimize acoustic communications with knowledge of the sound channel, but implementation has traditionally been hindered by computationally expensive propagation models. This work addresses computational complexity by enabling machine learning to interpret environmental inputs to predict transmission loss (TL) outputs from a physics-based ray tracing model (BELLHOP). Feature representations to reduce bathymetric and sound speed profile (SSP) information through wavelet decomposition and empirical orthogonal analysis are explored, respectively. A decision tree machine learning algorithm is trained to learn the paths (depths and ranges) of acceptable transmission loss for a pair of acoustic modems within a given environment. Our test case is off the coast of Southern California, and SSPs are collected from a 3 year record sampled from an ocean mooring. Bathymetry data is based on high resolution multibeam data. TL field realizations are calculated as training and testing data for environments of varying bathymetry with frequency of 25kHz and range of 1km through the BELLHOP model. Results are shown with stationary and non-stationary emitters to demonstrate the effectiveness of this approach.

1:40

2pSP3. Can unsupervised machine learning help us understand the data collected in the ocean? Wesley Chen (Phys. and Astronomy, Brigham Young Univ., 334 North 680 East, Vineyard, UT 84059, wesleychen1997@gmail.com) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Since the ocean is not a convenient place to conduct research, data are often collected from offshore buoys. However, the data collected from the buoys usually do not have labels (e.g., source of the sound) attached to them. Manually labeling the data is time-consuming, but labeled data are necessary to use powerful supervised machine learning methods. We propose that unsupervised learning can be used to assign preliminary labels to data samples. To show the utility of this approach, we took an echosounder to Utah Lake and collected sonar images, some of which showed evidence of fish. We used Python to crop the images, similar to how regions of interest are identified in image processing. These cropped images are used in an unsupervised k-means clustering algorithm. The images are clustered based on visual features. An analysis of the resulting clusters and the feasibility of using unsupervised learning to provide preliminary labels to data is discussed.

1:55

2pSP4. Bayesian detection and tracking of odontocetes in 3D from their echolocation clicks. Junsu Jang (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9679 Caminito Del Feliz, San Diego, CA 92121, jujang@ucsd.edu), Florian Meyer, Eric Snyder, Sean Wiggins, Simone Baumann-Pickering, and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

To study behaviors of marine mammals in a nonintrusive manner, their bio-acoustical signals can be recorded by volumetric hydrophone arrays that provide time difference of arrival (TDOA) measurements for localization and tracking. Multi-target tracking (MTT) in 3-D using TDOA measurements from multiple sensors, however, must cope with non-linear measurement models and high-dimensional states. False alarms, missed detections and unknown data associations impose further challenges, often requiring human operators to annotate the data manually. We propose a data processing chain that automatically detects and tracks odontocetes from their echolocation clicks. The echolocation clicks are detected with a generalized cross-correlation that whitens the instrument noise. Two stages of tracking are performed using a tracking framework based on factor graphs and the sum-product algorithm (SPA). The odontocetes are first tracked in the TDOA domain to remove false alarms and then in the 3-D domain, fusing the tracked TDOAs across all sensors. To efficiently handle the considered non-linear and high-dimensional MTT scenario, particle flow is embedded in the SPA. According to simulation results, the proposed tracking method outperforms the existing approach using manual data annotation. Tracking of Cuvier's beaked whales (*Ziphius cavirostris*), whose echolocation clicks are recorded by two volumetric hydrophone arrays, is demonstrated.

Invited Paper

2:10

2pSP5. Information geometry for environmental inversions in ocean acoustics. Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mkt24@byu.edu), Jay C. Spendlove (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Environmental inversions in ocean acoustics require the simultaneous estimation of many unknown parameters. Broad studies of multi-parameter models from diverse fields have shown that such inference problems are universally sloppy. Sloppiness is a phenomenon in which the predictions of a model are insensitive to all but a few key combinations of parameters. Sloppy model analysis is based on information geometry, an application of differential geometry to statistics. In this talk, we give an overview of information geometry for underwater acoustics models for environmental inversion, focusing on transmission loss (TL) in range-independent normal mode models. We demonstrate that these models are sloppy and how the geometry directly relates the information content of acoustic data to the relevance of environmental parameters. In particular, the model manifold quantifies what environmental information is encoded in ocean sounds and how unidentifiable parameters can be removed from the model to give a simplified, identifiable ocean acoustics model of comparable accuracy. We summarize physical insights revealed by this information geometry analysis for ocean inversion. [Work supported by Office of Naval Research]

Contributed Papers

2:30

2pSP6. Inferences of seabed characterization in a variable ocean environment. David P. Knobles (Knobles Sci. and Anal., 5416 Tortuga Trail, Austin, TX 78731, dpknobles@kphysics.org), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Dimensional reduction methods based on feature extraction are critical for statistical inference application in variable ocean environments. Presented are analyses for selected acoustic measurements made during the Seabed Characterization Experiment in 2021-2022. Broadband sources were deployed on the New England continental slope between the 200 and 400 m depth contours. The method of analysis includes first analyzing short range (higher grazing angles) acoustic measurements provided by both a broadband towed source and explosive sources. Features utilized for short-ranged data include those associated with seabed interferometry. Then, longer range data (lower grazing angles) are utilized to provide improved estimates of such quantities as the attenuations and sound speed gradients. Features utilized for long-ranged data include those associated with the dispersion of signals generated by explosive sources. As a general guide the mid-frequency band is utilized to infer characteristics about the upper portions of the sediment whereas the lower frequencies are utilized to infer the deeper portions of the seabed. [Work supported by Office of Naval Research.]

2:45

2pSP7. Seabed classification with ResNet-18 using multi-channel ship noise spectrograms. Ginger E. Lau (Phys., Emory Univ., 400 Dowman Dr., Atlanta, GA 30322, ginger.e.lau@gmail.com), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

Transiting cargo ships in the ocean generate noise that can be used to infer the effective seabed type below. Recently this has been demonstrated with a residual convolutional neural network trained on synthetic data that displays frequency and range-dependent transmission loss generated by a range-independent normal model and empirical source spectrum for ship noise. The generalizability of the trained model was tested on ship noise measured from the middle sensor of a 16-channel vertical line array near shipping lanes on the New England continental shelf. In contrast, this current work utilizes multiple sensors and seeks to determine the optimal combination of sensor data. It was discovered that including sensors near the seabed improved model performance. Best results were obtained when the model's average sensor depth was 75%–80% of the water depth. The distance between sensors had no discernible effect on performance. Although the almost isovelocity sound speed profile used in this study indicated little difference between one-channel and two-channel model performance, future analyses will test the hypothesis that the use of multiple sensors will improve model performance for greater sound speed variation. [Work supported by the Office of Naval Research and the National Science Foundation REU program.]

3:00

2pSP8. Machine learning a surrogate model for transmission loss on a Pekeris waveguide. Jay C. Spendlove (Phys. and Astronomy, Brigham Young Univ., C341 ESC, Provo, UT 84602, jayclark24@gmail.com), Mark K. Transtrum, and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The connection between ocean environmental parameters and transmission loss (TL) can be modeled using sound propagation models. We can analyze and improve these models using information theoretic tools such as the Fisher information matrix (FIM) and methods for model reduction, where we identify parameters that can be removed from a model without sacrificing accuracy. Critical to these methods for model reduction is the ability to evaluate derivatives of the model predictions of TL with respect to model parameters. Calculating derivatives of TL models using finite difference methods with sufficient accuracy is challenging due to the complexity of sound fields in an ocean environment; instead, we train a surrogate machine learning model to which we can apply automatic differentiation (AD). We propose a general method for model reduction in which a ML model is used as a surrogate model for physical sound propagation ocean models, which is then used to find potential simplifications of the original model. We demonstrate this method using the Pekeris waveguide model of the ocean, which calculates underwater acoustic TL in an ocean environment due to seafloor characteristics. [Work supported by Office of Naval Research]

3:15–3:30 Break

3:30

2pSP9. Waveform modeling with Gaussian Processes for inversion in ocean acoustics. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

It has been shown that, in ocean acoustics, Gaussian processes can predict a densely sampled field on a receiving array, when only sparse samples of the sound in the water column are initially available. In a similar manner, a Gaussian process approach can be designed to densely sample signals in the time domain, resulting from the transmission and propagation of broadband waveforms. A mean waveform can be obtained that allows the high-resolution estimation of multipath arrival times. These can then be used for source localization and geoacoustic inversion. Uncertainty quantification in the time-series characterization, readily available from the Gaussian process modeling, facilitates uncertainty quantification in inversion, obviating the need for onerous computations. We investigate the potential of different kernels in aiding the inversion process. [Work supported by ONR.]

Invited Paper

3:45

2pSP10. On data-driven underwater acoustic direct localization: Design considerations of a deep neural network-based solution. Amir Weiss (Elec. Eng. and Comput. Sci., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, amirwei@mit.edu), Grant B. Deane (Scripps Inst. Of Oceanogr., La Jolla, CA), Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), and Greg Wornell (Elec. Eng. and Comput. Sci., MIT, Cambridge, MA)

High reverberation, depth-dependent speed of sound and intricate environmental structures are a few of the challenges that make the development of reliable underwater acoustic localization (UAL) methods hard. Since the nontrivial physics of underwater acoustics and the inherent random effects amount to a highly complex statistical model of the measured data, it is natural to consider deep neural networks (NNs) as a central tool in the development of UAL methods. Indeed, recent studies have shown that some properly designed NNs can lead to unprecedented localization capabilities and enhanced accuracy. However, out of the already myriad available possibilities, it is not immediately clear how to choose a proper NN architecture—input structure, layer types, training procedure, loss function(s), etc.—that will lead to successful operation. In this talk, we will present an analytically informed architecture that can learn (for now, in simulations) to localize in complex underwater environments. We will also provide analytical arguments that can explain the robustness of such architectures to certain types of propagation/environment model mismatches. The operation of the discussed architecture will be demonstrated via simulation results in a rich, reverberant channel. The results suggest that deep NNs could become a viable solution to the UAL problem.

Contributed Papers

4:05

2pSP11. Underwater acoustic localization via gradient-based optimization. Dariush Kari (Elec. and Comput. Eng., Univ. of Illinois Urbana Champaign, 1308 W Main St., Urbana, IL 61801, dkari2@illinois.edu), Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore), and Amir Weiss (Elec. Eng. and Comput. Sci., MIT, Cambridge, MA)

In underwater acoustic localization via matched-field-processing, given a propagation model and a suitable environmental parameterization, one searches for the location (of the transmitter or receiver) whose replica field is closest to the observed one. The high computational complexity of such non-gradient-based optimization methods renders them infeasible for many real-time scenarios, especially when an accurate solution is desired due to resolution of the search grid required, or as the search dimensionality increases (e.g., when it is necessary to optimize over uncertain environmental parameters such as sound speed or bathymetry). In this talk, we propose a ray-based, differentiable model for acoustic propagation that can be exploited in a gradient-based optimization for localization. For localization applications in which accurate times of arrival might not be available (e.g., due to the signal's relatively small bandwidth), the proposed method does not directly rely upon times of arrivals. Rather, it seeks the location (and possibly environmental parameters) that minimize the squared-error between the observed signal and its estimation via the differentiable model. We leverage the PyTorch optimization and auto-differentiation tools for the implementation and demonstrate successful localization on synthetic data in a dense multipath environment.

4:20

2pSP12. Comparison of manifold representations of sonar data. Bernice Kubicek (Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52242, bernice-kubicek@uiowa.edu), Ananya Sen Gupta (Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

There are many challenges in active sonar target recognition due to the dynamic nature of the environment, unknown target geometries, and various clutter present within the ocean. These factors combine and entangle the target features within the received response. This research assumes the informative and discriminatory acoustic target features lie on a low dimensional manifold and can be extracted using statistical and machine learning techniques. Linear techniques, such as principal component analysis and linear discriminant analysis, are useful for dimension reduction but will typically not accurately capture the underlying non-linearity within a dataset. Nonlinear manifold learning techniques, such as T-distributed stochastic neighbor embedding and uniform manifold approximation and projection, are relatively recent techniques that create low-dimensional embeddings in an unsupervised fashion and can capture the non-linearity. Previously, persistent braid features have been reported in real sonar data[1]. Our objective here is to do comparison between these different manifold representations for both simulated data with a known ground truth and experimental active sonar data. [1] A. Sen Gupta, B. Kubicek, A. Christensen, and I. Kirsteins, "Geometric feature representation in active sonar signal processing," *OCEANS 2022—Chennai*, 2022, pp. 1–5. [Work supported by NDSEG 2021 and ONR Grant No. N00014-21-1-2420].

4:35

2pSP13. Analysis of braid manifolds using uniform manifold approximation and projection. 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 (NUWC, Newport, RI)

Autonomous sonar target recognition has been an ongoing challenge in the underwater signal processing community for many decades. Aspect-dependent target scattering and waveguide propagation causes target features to overlap, which increases the difficulty of constructing feature dictionaries. Current machine learning algorithms can suffer from an inability to track morphing target-specific features. Other "black-box" machine learning algorithms produce results that are not explainable. We seek to extend previous work on creating feature representations using braid manifolds. Specifically, we use Uniform Manifold Approximation and Projection (UMAP) as a dimensionality reduction technique to search for underlying features that lay on a manifold. UMAP is an unsupervised learning algorithm that is built upon Riemannian geometry and fuzzy simplicial sets. The low dimensional representation embedding that UMAP produces is computed using both stochastic approximate nearest neighbor search and stochastic gradient descent with negative sampling. The performance of UMAP embeddings will be compared against other common dimension reduction algorithms and evaluated using various classifiers. [The authors would like to thank the Office of Naval Research for funding under Grant No. N00014-21-1-2420 and the University of Iowa.]

Invited Paper

4:50

2pSP14. Reducing dimensionality of spectrograms using convolutional autoencoders. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wjenkins@ucsd.edu), Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA), Chih-Chieh Chien (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), and Emma Ozanich (JASCO Appl. Sci., Woods Hole, MA)

Under the "curse of dimensionality," distance-based algorithms, such as *k*-means or Gaussian mixture model clustering, can lose meaning and interpretability in high-dimensional space. Acoustic data, specifically spectrograms, are subject to such limitations due to their high dimensionality: for example, a spectrogram with 100 time- and 100 frequency-bins contains 10^4 pixels, and its vectorized form constitutes a point in 10^4-dimensional space. In this talk, we look at four papers that used autoencoding convolutional neural networks to extract salient features of real data. The convolutional autoencoder consists of an encoder which compresses spectrograms into a low-dimensional latent feature space, and a decoder which seeks to reconstruct the original spectrogram from the latent feature space. The error between the original spectrogram and reconstruction is used to train the network. Once trained, the salient features of the data are embedded in the latent space and algorithms can be applied to the lower-dimensional latent space. We demonstrate how lower-dimensional representations result in interpretable clustering of complex physical data, which can contribute to reducing errors in classification and clustering tasks and enable exploratory analysis of large data sets.

Contributed Paper

5:10

2pSP15. Sonar feature representation with autoencoders and generative adversarial networks. Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52240, tlinhardt@uiowa.edu) and Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

Feature representation in the littoral sonar space is a complicated field due to prevalent channel noise from sound reflections as well as diffraction. The response of a sound wave as it interacts with an object provides insight into the nature of the object itself such as geometry and material composition. We approach automated target recognition in this space with feature representation using autoencoders and generative adversarial networks. Through empirical analysis of learned encoding spaces, the dimensionalities of principal features in our sonar data sets are estimated. Real and complex valued training data, processed in various ways, is used to refine which features are represented, and evaluate the importance of phase information while encoding the data. [This research was funded by The Office of Naval Research with Grant No. N00174-20-1-0016.]

Exhibit

An instrument and equipment exhibition will be located in Chicago D/E on the 5th floor.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Monday, 8 May, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including a complimentary beverage.

Tuesday, 9 May, 9:00 a.m. to 5:00 p.m.: Exhibit Open Hours including a.m. coffee break and p.m. break with coffee and soft drinks.

Wednesday, 10 May, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. coffee break.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. All meetings will begin at 7:30 p.m., except for Computational Acoustics (4:30 p.m.), Engineering (4:45 p.m.), and Signal Processing in Acoustics (5:30 p.m.).

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 discussion.

Committees meeting on Tuesday:

Engineering Acoustics (4:45)	Northwestern/Ohio State
Signal Processing in Acoustics (5:30)	Purdue/Wisconsin
Acoustical Oceanography	Michigan/Michigan State
Animal Bioacoustics	Chicago FG
Architectural Acoustics	Chicago AB
Physical Acoustics	Los Angeles/Miami/Scottsdale
Psychological and Physiological Acoustics	Indiana/Iowa Acoustics

Committees meeting on Wednesday:

Biomedical Acoustics	Armitage
Structural Acoustics and Vibration	Chicago H

Committees meeting on Thursday:

Computational Acoustics (4:30 p.m.)	Lincolnshire 1/2
Musical Acoustics	Lincolnshire 1/2
Noise	Chicago C
Speech Communication	Los Angeles/Miami/Scottsdale
Underwater Acoustics	Michigan/Michigan State

Session 3aAA

Architectural Acoustics, Noise, Structural Acoustics and Vibration, Engineering Acoustics, and ASA Committee on Standards: Application and Development of Standards Used in Noise and Architectural and Structural Acoustics

Benjamin M. Shafer, Cochair
PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Evelyn Way, Cochair
Maxxon Corp, 920 Hamel Road, Hamel, MN 55340

Michael Raley, Cochair
PAC International, 2000 4th Ave., Canby, OR 97013

Invited Papers

8:00

3aAA1. Interdependence of sound standards. Derrick P. Knight (Trane Technologies, 2313 20th St. South, La Crosse, WI 54601, Derrick.Knight@TraneTechnologies.com)

Sound standards are developed by multiple organizations, which rely on one another to build complex dependencies. These interdependencies are explored through the example of using ASA/ASNI S12.60 in a hypothetical application of quiet classroom design. The dependencies are contextualized through multiple viewpoints to better understand how each user has their own hierarchy of standards prioritization while benefiting from the ability to trust the infrastructure provided by standards outside their direct area of interest. Finally, the prioritization hierarchy for each point of view is compared to demonstrate why different users might choose to engage in standards development through different organizations.

8:20

3aAA2. Quantification of individual factors impacting measurement of sound. Viken Koukounian (K.R. Moeller Assoc. Ltd., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@logison.com)

In the science of metrology, various terms—such as trueness, precision, accuracy, repeatability, reproducibility, and bias—are used to describe the statistics describing the distribution of measurement results, to establish understandings of degree of closeness of results, closeness of agreement between results of tests, and so on. Although there are many factors that affect the variability of results from a measurement method, the impact of individual factors is confounded or not sufficiently differentiated from the collective. In the following contribution, the impacts of different measurement methods (e.g., styles and procedural differences), operators, measurement instruments, and architectures (e.g., finishings, furnishings, and fit-outs) are quantified. The results from this work are critical to develop a higher level of understanding of procedural guidance in international standards. Examples of applications benefiting from these insights include the evaluation of ambient acoustic conditions (i.e., background noise and masking sound) and of sound isolation.

8:40

3aAA3. The creation of functional standards to streamline ASTM standards in committee E33 on building and environmental acoustics. Matthew Golden (Pliteq, 131 Royal Group Crecent, Woodbridge, ON L4H 1X9, Canada, mgolden@pliteq.com)

ASTM E33 standards are in wide-spread use in North America in building and environmental acoustics. Some of these standards have been in existence in one form or another for over 60 years. There are now 55 standards within the E33 committee all created by different volunteer authors at different times. Each of these standards is periodically updated by other volunteers. While efforts have been made to keep agreement among these standards, subtle and not so subtle differences creep up in similar measurements, calculations, and specifications. A small working committee has identified 21 topics that are candidates for consolidation into "functional" standards. These functional standards will be used similarly to a computer program function and can be "called" from any other standard. The hope is that this will remove inconsistences among the standards and make the standards easier to read. The current status and future plans of this project will be shared.

8:55

3aAA4. Quantifying repeatability, reproducibility, and bias in sound power level measurement methodologies via a round-robin study. 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) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

This paper presents results from an interlaboratory study to examine the repeatability, reproducibility, and bias of three different sound power measurement methods used in the heating, ventilation, and air-conditioning industry: free field method, diffuse field method, and intensity method. A loudspeaker playing both a broadband signal and a broadband signal with four discrete tones has been measured in participating laboratories according to the standardized test methods that are commonly used in each facility. Results across 1/3 and 1/1 octave bands are then compared between signals, between methods, between laboratories, and across all laboratories using the same method. Additional discussion regarding potential sources of variation between facilities is presented. Ultimately, in accordance with ISO 5725, repeatability, reproducibility, laboratory bias, and test method bias are estimated. To reduce uncertainty of these estimates for some methods, additional laboratory participants are needed. [Work supported by the Air-Conditioning, Heating, and Refrigeration Institute.]

9:15

3aAA5. E90 round-robin—Status and path forward. Evelyn Way (Res. & Development, Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, evelynway@gmail.com) and Michael Raley (PAC Int., Canby, OR)

A good understanding of the uncertainty of a test method is necessary to interpret the results of said test. The ASTM E90 test has many potential sources of variation that can lead to significant uncertainty in the test results, but this uncertainty is not well documented in the precision and bias statement within ASTM E90. The current precision and bias statement provides some limited repeatability data for a wood joist floor and a concrete slab and only provides reproducibility data for the ASTM E1289 reference specimen. The E1289 reference specimens are simple steel plates in single and double-layer configurations that yield STC ratings far below those of the wall and floor/ceiling assemblies that are commonly tested. The authors are in charge of a new round-robin that aims to provide more detailed repeatability and reproducibility data using common wall designs with STC ratings in the range more commonly seen in building code requirements and design guidelines. In this presentation, the authors will cover their proposed plan for the round-robin and will give an update on its status.

9:30

3aAA6. An overview of the proposed new ASTM standard for impulse response measurements using swept sinusoidal signals. Markus Mueller-Trapet (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, Markus.Mueller-Trapet@nrc-cnrc.gc.ca)

This contribution presents the current draft of the proposed ASTM standard for impulse response measurements using swept sinusoidal signals for use in room and building acoustics. After briefly discussing the theoretical background and main body of the proposed standard, the main focus of the presentation is the validation procedure to compare reverberation time measurements between the interrupted noise method and the impulse response method. The challenges and associated uncertainties of the validation procedure are discussed, and preliminary results for rooms of different sizes are shown.

9:50–10:00 Break

10:00

3aAA7. Floor impact noise—The history and future of ASTM E3133. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

The fundamental purpose of ASTM E3133 is to provide a test method for rating the relative noisiness of floor impacts (e.g., footfalls, dropped objects, and rolling carts) on a variety of finished flooring products. Unlike previous impact noise standards which only address the impact noise radiated to other nearby spaces, the focus of this test method is to evaluate the noise radiating into the source room where the floor impacts occur. Health care facilities and performing arts venues are an obvious application, but data of this type may also be important in other facility types including schools, churches, and courtrooms. Major flooring product manufacturers have been asking for this type of standardized test data for many years. This presentation will highlight the work conducted over the last decade in the development of this standard, culminating in the adoption of the standard in 2018, its recent withdrawal, and plans for a future revision.

10:20

3aAA8. International green construction code (IgCC)/ASHRAE 189.1: History and use of acoustical control section. Erik Miller-Klein (Tenor Eng. Group, 11514 Dayton Ave. N, Seattle, WA 98133, erik.mk@tenor-eng.com) and Viken Koukounian (K.R. Moeller Assoc. Ltd., Burlington, ON, Canada)

This seminar will provide historical context on the current Acoustical Control section 801.3.3 in the International Code Council's International Green Construction Code (IgCC) is based on the ASHRAE Standard 189.1 *Standard for the Design of High-Performance Green Buildings*. This includes a detailed review of the current language and application, and guidance on the best way to design and commission to this standard.

10:35

3aAA9. A new measurement method for floor-ceiling assembly performance with improved reproducibility. Sunit Girdhar (Mech. Eng., Michigan Technolog. Univ., Houghton, MI 49931, sgirdhar@mtu.edu), Andrew Barnard (Acoust., Penn State, University Park, PA), Jason R. Blough (Mech. Eng., Michigan Technolog. Univ., Houghton, MI), John Lo Verde, and Wayland Dong (Veneklasen Assoc., Santa Monica, CA)

ASTM standards are used to calculate a single number performance rating for floor-ceiling assembly impact noise. A higher number represents better performance, and a lower number represents poor performance. This number is calculated by placing a standard tapping machine on the floor and measuring the radiated sound pressure level (SPL) in the receiving room below. In theory, the performance of an assembly can be represented by just this calculated number. In practice, however, the same assembly tested in the same configuration can lead to a range of single-number ratings, instead of just one number. This variation primarily comes from the SPL measurement, which is highly variable in different parts of the room at low frequencies because of the existence of a non-diffuse sound field. This variation in SPL measurement for the ASTM standards leads to a high non-reproducibility which may cause confusion among the consultants and the clients. In our work, we are proposing a new measurement method with improved reproducibility in low frequencies.

10:55

3aAA10. Investigation of vibration measurements of heavy/hard impacts. Michael Raley (PAC Int., 2000 4th Ave., Canby, OR 97013, mraley@pac-intl.com) and Simon Edwards (HGC Eng., Mississauga, ON, Canada)

Further to the ASTM field test standard currently under development for heavy/hard impacts, the potential inclusion of vibration measurements warrants additional investigation. This presentation will explore measurement methodology and parameters based on a combination of laboratory and field-testing experience. Broadly speaking, the authors will address the purpose of vibration measurements of heavy/hard impacts, the current uses of vibration measurements, and a range of relevant parameters that require consideration in outlining a standardized measurement methodology. The authors intend to present initial guidance on vibration measurement parameters based on their experience in laboratory and field measurements of heavy hard/hard impacts. These preliminary discussions are intended to inform and invigorate further investigation by the acoustics community with a goal of developing a more robust measurement methodology and standard.

11:10

3aAA11. A new approach for auralization of scattering in urban spaces. Anne Heimes (IHTA, RWTH Aachen Univ., Kopernikusstraße 5, Aachen, NRW 52074, Germany, ahe@akustik.rwth-aachen.de) and Michael Vorlaender (IHTA, RWTH Aachen Univ., Aachen, Germany)

Absorption and scattering coefficients of surfaces are crucial for acoustic propagation simulations. The scattering coefficient according to ISO 17497-1, however, is the most uncertain standard metric applied in geometrical acoustics. In outdoor sound propagation simulation, the amount of scattered energy is crucial for the sound immission at the receiver. Standard random-incidence scattering coefficients are in use but with uncertain results. In contrast, specific reflection patterns are neither available nor implemented in simulation algorithms. In this paper, the scattering coefficient concept will be applied for building facades in an urban test environment. Furthermore, results from a new angle-dependent specific scattering metric, including sound steering and retro-reflection, will be compared with results from purely specular or mixed specular/random-incidence datasets.

Contributed Papers

11:30

3aAA12. Tuning the octave bands: A history of dividing the spectrum for acoustical measurements in the 20th century. Zachary Weiss (Noise Control Eng., LLC, 85 Rangeway Rd. Bldg. 2, Billerica, MA 01862, zfweiss@noise-control.com)

Standardized octave and one-third octave bands are ubiquitous in modern acoustical measurements, such that the particular way they divide the spectrum might reasonably be assumed to be self-evident. Yet when Hans Thilo was granted the patent for the first octave band analyzer in 1937, this division was anything but certain. In the ensuing years, many systems proliferated, with different center frequencies employed between airborne noise and vibration, source and absorption measurements, and even American and European practitioners. This presentation traces the multiple systems in play from Wallace Sabine's early experiments with organ pipes up through the dominant standard today based upon preferred numbers. Though the large majority of these systems have long since been abandoned, their ghosts are still occasionally encountered, haunting compliance studies beholden to legacy noise ordinances, and muddling the definitions of metrics such as speech interference level (SIL), which were originally defined using older octave bands.

11:45

3aAA13. Acoustic solutions for equipment noise: Pipe, fan coil, and ventilation duct. Pascal M. Raphoz (Armacell, 8 bis rue Quinault, Résidence Racine, Saint Germain en Laye, Yvelines 78100, France, pascal.raphoz@armacell.com)

Part 1: pipe lagging Insertion loss for drainage pipes in residential and non-residential buildings review of different standards (EN 14 366 versus ASTM E 1222) used in Americas and Europe ways of optimization: benefits of multilayers. Best practices: examples of project references. *More detail information on request* Part 2: fan coil and duct lagging Fan coil: how to reduce sound power level? Iso standard ISO 3741 New optimization path with examples Duct lagging: how to reduce breakdown noise? Frequencies spectrum optimization Best practices *More detail information on request*.

Session 3aABa

Animal Bioacoustics: Acoustic Ecology and Biological Soundscapes

Laura Kloepper, Chair
Department of Biological Sciences, University of New Hampshire, 230 Spaulding Hall, Durham, NH 03824

Contributed Papers

8:00

3aABa1. Passive acoustic localization and tracking of Rice's whales (Balaenoptera ricei) in the northeastern Gulf of Mexico. Ludovic Tenorio (CIMAS/NOAA, 75 Virginia Beach Dr., Key Biscayne, FL 33149, ludovic.tenorio@noaa.gov), Pina Gruden (Joint Inst. for Marine and Atmospheric Res., Honolulu, HI), Heloise Frouin-Mouy (Cooperative Programs for the Advancement of Earth System Sci., Univ. Corp. for Atmospheric Res., Miami, FL), Amanda Debich, Ashley Cook (CIMAS/NOAA, Miami, FL), Lance Garrison (Southeast Fisheries Sci. Ctr., NOAA, Miami, FL), Eva-Marie Nosal (Ocean & Resources Eng., Univ. of Hawaii, Honolulu, HI), and Melissa Soldevilla (Southeast Fisheries Sci. Ctr., NOAA, Miami, FL)

The endangered Rice's whale (Balaenoptera ricei) is endemic to the Gulf of Mexico and is also the Gulf's only resident baleen whale. Limited knowledge of this species' ecology combined with the high industrialization of this region raises serious conservation concerns. As part of an ongoing project to study the Rice's whale in its core habitat, an array of moored passive acoustic monitoring sites has been near-continuously deployed in the northeastern Gulf of Mexico since May 2021. While detecting Rice's whale calls in these data provides valuable insight into the species' spatiotemporal distribution, further analyses to localize and track individual whales would allow better characterization of their acoustic behavior and movement patterns, possibly allowing for density estimation. Here, we present a method for automated passive acoustic localization and tracking of calling Rice's whales. The two key components of this approach are (1) the use of opportunistic sound sources to time-synchronize recordings across sites given clock-drift between acoustic recorders and (2) the implementation of automatic multi-target tracking techniques based on the Gaussian mixture probability hypothesis density filter. Analyses of the first four months of data show potential for measuring basic behavioral parameters such as calling rate, source level, and average swim speed.

8:15

3aABa2. Short-term changes in the underwater soundscape of Mosquito Lagoon after Hurricane Ian. Nathan Wolek (College of Arts and Sci., Stetson Univ., 421 N Woodland Blvd., Unit 8252, DeLand, FL 32723, nwolek@stetson.edu), Katrina Early, Marvel Olson, and Dylaney Sabino (College of Arts and Sci., Stetson Univ., DeLand, FL)

During late September 2022, Hurricane Ian passed over the east coast of Florida and Mosquito Lagoon. This 4740-acre body of water is part of a larger estuary system sandwiched between the mainland and narrow barrier islands. Two-thirds of the lagoon is located within Canaveral National Seashore, which helps maintain a balance between protecting its rich biodiversity and allowing access for recreational boating. As part of a previously planned study, we deployed two HydroMoths in Mosquito Lagoon to capture the underwater soundscape for two months from September 24 to November 19. Our sampling protocol recorded 5 min of continuous sound at the top and bottom of every hour, with the idea that we could begin to explore diurnal and lunar cycles. When Hurricane Ian's path took it directly over our recorders, we were given an unplanned opportunity to document how the underwater soundscape would respond. The most striking change came from the oyster toadfish (Opsanus tau). Calls from males of this species are a prominent part of the underwater soundscape in Mosquito Lagoon during their breeding season. In the days immediately following Hurricane Ian, they lowered the fundamental frequency of their calls leading to an audible change in pitch.

8:30

3aABa3. Long-term and short term effects of elevated underwater sound on oyster toadfish (*Opsanus tau*). Benjamin R. Colbert (Univ. of Maryland Ctr. for Environ. Sci., 146 Williams St., Solomons, MD 20688, bcolbert@umces.edu)

The effects of underwater sound on fishes are of increasing concern for managers, regulators, and researchers. While there has been work which demonstrated changes in behavior, these studies have generally evaluated of immediate or near-term responses. Understanding acute changes in behavior is necessary, but questions about long-term responses to underwater sound, including any habituation and the relative influence of environmental conditions versus anthropogenic sound must also be investigated. To further elucidate potential response to chronic underwater vessel generated sound, passive acoustic monitoring was used to measure the response of vocalizing oyster toadfish (*Opsanus tau*) to vessel sounds within the Chesapeake Bay. An automatic detector was configured which allowed for counting of vessel passages and toadfish vocalizations. The response of vocalizing toadfish individual vessel movement was then evaluated. Next the response of toadfish to elevated sound levels across a 30-day period was measured and compared to short-term response. In addition, the influence of environmental variables and contrasted effects of these on the effects of increased anthropogenic sound was investigated.

8:45

3aABa4. Estimating populations of dense animals using acoustics. Valerie M. Eddington (Biological Sci., Univ. of New Hampshire, 6 Stonecroft, Apt 6, Portsmouth, NH 03801, valerie.eddington@unh.edu), Easton White (Biological Sci., Univ. of New Hampshire, Durham, NH), and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

Acoustic censusing methods offer an efficient and noninvasive method to monitor populations of dense aggregations of animals on a large scale. Large, dense colonies of bats are difficult to census, and current monitoring methods are often invasive and time consuming. Using acoustic recordings collected during emergence, we developed a linear regression model that uses multiple measures of acoustic energy to estimate the number of bats per second. The model was trained using data from audio recordings collected at seven cave locations and two species of bats, Brazilian free-tailed bats (*Tadarida brasiliensis*) and gray bats (*Myotis grisenscens*). Visual counts from synchronized thermal videos were used to ground truth the model. Environmental factors, such as distance from microphone to emergence, were integrated to create flexible model that is widely applicable across species and habitats. This method can be extended to other species that vocalize in large aggregations, such as sea birds and frogs. Ultimately, this model can be integrated into autonomous monitoring stations to monitor species in real time. It is imperative that efficient and noninvasive methods

for monitoring bat populations are developed to increase population data available for informing crucial management decisions.

9:00

3aABa5. A generic AI pipeline for identifying known species and discovering new species. Marconi Campos-Cerqueira (Sci., Rainforest Connection, PO BOX 1180, Barranquitas 00794, Puerto Rico, marconi@rfcx.org)

To counterattack the current biodiversity crisis, we need innovative tools and technologies that can greatly improve species identification and discovery at scale. Here, we describe a data analysis pipeline to automatically identify biological sound categories in soundscape data and accelerate the discovery of new species. The pipeline consists of two components: (1) Unsupervised audio event detection and clustering that enables automatic detection and categorization of biological sounds in large audio datasets and (2) supervised deep learning-based signal detection that enables automated cluster labeling and detection of known sound categories in new audio. We evaluated the results of our unsupervised analysis tools applied to a labeled dataset of $\sim$18 h of recordings from Panama. Experts assigned a label to each sound event with a max frequency of 6 kHz. Our results show that clusters generally had high homogeneity and completeness and indicate that our cluster analysis tool can successfully detect and group like-sounds together, improving data processing and species identification. In addition, we were able to implement these analytical tools in a user-friendly data analysis platform that will provide scientists, wildlife managers, conservationists, and public and private environmental organizations with the information they need to make informed conservation and management decisions.

9:15

3aABa6. Potential biological impacts of very low frequency acoustical energy produced by offshore wind turbine energy generation. Michael Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@ocr.org)

Offshore wind turbine farms are being planned and installed around the US Outer Continental Shelf. These will be introducing a panoply of noises into the environment, from siting, through installation, to operations, and eventually decommissioning. Noise concerns from these activities are typically framed within frequency bands of concern defined by governmental regulatory agencies. A less considered noise is the very low frequency thump generated by the motion of the turbine blades as they intersect the "stagnate wind area" on the windward side of the mast. This "infrasonic" thump contains a lot of energy and is in the perceptual range of migratory birds and baleen whales. Given that whales and birds use "infrasonic" sound and barometric pressure signals for navigation and migration cues, the installation of thousands of turbines off the Atlantic Outer Continental Shelf, and hundreds off the Pacific OCS, may impose significant impacts on migratory passerine birds along the "Atlantic Flyway," Procellariiformes on both the Atlantic and Pacific OCS, and all baleen whales within the infrasonic energy propagation reach of the turbines.

9:30

3aABa7. Beneath the tree: The sounds of a trembling giant. Jeff Rice (Acoust. Atlas, Montana State Univ., P.O. Box 173320 Centennial Mall, Bozeman, MT 59717-3320, jrice1000@icloud.com) and Lance Oditt (Friends of Pando, Seattle, WA)

The Pando aspen grove (*Populus tremuloides*) consists of more than 40 000 genetically identical aspen stems spread across 106 acres in south-central Utah. While Pando resembles a forest, it is a single organism connected at the roots, making it the world's largest tree and one of the largest living things on the planet. First identified in 1976, Pando has not been studied as a bioacoustic subject. Focusing on recordings captured simultaneously above and below ground on July 12, 2022 during a thunderstorm, our presentation reveals a unique acoustic portrait of this botanical wonder. We compare audible sounds of Pando's leaves with hydrophone recordings made in connection with the tree's root system. This dual soundscape shows that leaf motion causes vibrations that pass throughout the organism, from its branches to its base. The recordings suggest that the grove's massive, interconnected root system is highly resonant with potential for future recordings and research.

Session 3aABb

Animal Bioacoustics: Animal Vocal Communication and Physiology

Laura Kloepper, Chair
Department of Biological Sciences, University of New Hampshire, 230 Spaulding Hall, Durham, NH 03824

Contributed Papers

10:30

3aABb1. Adaptive signal adjustments in humpback whale songs. Vitalis Wagome (Alabama State Univ., 915S Jackson St., Montgomery, AL 36104, vwagome0975@myasu.alasu.edu) and Eduardo Mercado (Univ. at Buffalo, SUNY, Clarence, NY)

The use of vocal flexibility by humpback whales (Megaptera noavaengliae) often described as "singing" is a highly sophisticated communicative process, possessing a multilayered hierarchical structure across the various acoustical parameters. A deeper spectral analysis of songs made available through Google's Pattern Radio website studied in the month of February 2015 display several adaptive signal adjustments in humpback whale songs. As single singer variations in song properties become more clear, multi-singer variations in song properties remain hypothetical. This raises an important question of incorporating a temporal analysis approach to determine the multi-singer variations in song properties.

10:45

3aABb2. Acoustic properties of chimpanzee pant-hoots reflect male mate quality. Nisarg Desai (Neurosci., Univ. of Minnesota, 2021 6th St. S.E., Minneapolis, MN 55455, desai054@umn.edu), Pawel Fedurek (Psych., Univ. of Stirling, Stirling, United Kingdom), Katie Slocombe (Psych., Univ. of York, York, United Kingdom), Adam Clark Arcadi (Anthropology, Cornell Univ., Ithaca, NY), Lisa O'Bryan (Elec. and Comput. Eng., Rice Univ., Houstan, TX), Charlotte Uhlenbroek (London, United Kingdom), and Michael Wilson (Ecology, Evolution, and Behavior, Univ. of Minnesota, Minneapolis, MN)

Sexual selection theory predicts that acoustic structure may provide cues of individual traits correlated with mate quality. Chimpanzee pant-hoots are complex, conspicuous calls that may be products of sexual selection. Insofar as pant-hoots are difficult to produce, requiring individuals to approach their physiological limits of sound production, they may serve as honest signals of physical condition. The highest pitch elements in pant-hoots appear the most difficult elements to produce, as they sometimes exhibit distortions in acoustic structure known as non-linear phenomena (NLP). Producing high pitch vocalizations with fewer NLPs may, thus, signal superior physical condition. We examined whether the proportion of NLPs, the pitch, and noise in these elements of the pant-hoots contain cues of mate quality including age, rank, and a measure of health (infection with the simian immunodeficiency virus, SIVcpz), and if they predict male mating success. Consistent with predictions from sexual selection, we found that (i) the proportion of NLPs was associated with age in a non-linear fashion—specifically, subadult and old males exhibited higher proportions of NLPs compared to males in their prime mating age; (ii) a lower proportion of NLPs predicted higher mating success; and (iii) SIVcpz positive individuals exhibited more noise in their pant-hoots.

11:00

3aABb3. Regularities in spectral entropy variations within humpback whale song sessions. Eduardo Mercado (Univ. at Buffalo, SUNY, 4360 GUNNVILLE Rd., Clarence, NY 14031, emiii@buffalo.edu), Mariam Ashour (Univ. at Buffalo, SUNY, Buffalo, NY), Gala Krsmanovic, and Julia Hyland Bruno (New Jersey Inst. of Technol., Newark, NJ)

Consecutive songs produced by humpback whales recorded off the coast of Hawaii show recurrent patterns of acoustic variation. Analyses of spectral entropy, a measure of the uniformity of energy distribution in the frequency domain, revealed that spectral entropy cycles in predictable patterns within song sessions. At time scales spanning tens of minutes, variations in spectral entropy appear to track changes in received intensity levels, which are correlated with song duration and may reflect the diving behavior of singers. Intra-individual variations in spectral entropy suggest that singing humpback whales may also vocally control entropic features of songs, both by gradually modulating the acoustic characteristics of units and by varying the amount of time they spend producing specific patterns of units. Preliminary analyses suggest that variations in spectral entropy within song cycles are indicative of the amount of time and energy singers spend producing constant-frequency (CF) elements versus more frequency-modulated/broadband elements. CF elements of units are generally more difficult to spatially localize, but easier to detect over long distances, suggesting that singing humpback whales may be varying the production of consecutive songs in ways that affect auditory spatial processing.

11:15

3aABb4. Behavioral responses of bald eagles (*Haliaeetus leucocephalus*) to acoustic stimuli in a laboratory setting. JoAnn McGee (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, VA Loma Linda Healthcare System, Loma Linda, CA 92357, mcgeej@umn.edu), Christopher Feist, Christopher Milliren (St. Anthony Falls Lab., Univ. of Minnesota, Minneapolis, MN), Lori Arent, Julia B. Ponder (The Raptor Ctr., Univ. of Minnesota, St. Paul, MN), Peggy Nelson (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), and Edward J. Walsh (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, Loma Linda, CA)

In our previous work, auditory function and vocalization acoustics were studied in a cohort of adult and juvenile bald eagles (*Haliaeetus leucocephalus*) to guide the development of acoustic deterrence technologies intended to discourage encroachment into wind energy airspaces. Findings from those studies set the stage for an investigation designed to evaluate the efficacy of sound stimuli to elicit behavioral responses from eagles in a laboratory setting. To achieve this goal, a handler positioned individual eagles on a perch in the center of an acoustically damped room equipped with two stationary speakers and a video camera. Stimulus presentation was controlled remotely

from outside the study space and initiated after the subject acclimated to the space and remained stationary. Both natural and synthetic stimuli were delivered randomly to one of the speakers. Several volunteers were recruited to assess stimulus-driven behavioral responses from recorded videos. We will report on results of behavioral responses for each stimulus type, as well as habituation to stimuli. Early results are promising and set the stage for exportation of behavioral studies into a more natural environment to evaluate the effects of acoustic stimuli on bald eagles during creance training flights. [Work supported by DOE-EERE (#DE-EE0007881) and LCCMR (#2021-294).]

11:30

3aABb5. Modeling and interpreting the head-related transfer function to understand directional hearing in bottlenose dolphins. Yeon Joon Cheong (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ycheong@apl.washington.edu), Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Alexander Ruesch, Matt Schalles, Jana Kainerstorfer, and Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA)

Toothed whales have evolved to communicate, forage, and navigate effectively underwater using sound. It is generally accepted that toothed whales receive sounds through their lower mandible and the associated fat body, which guide sound to the tympano-periotic complexes (TPCs) enclosing the cochleae. However, little is known about how the direction of an impinging sound wave affects acoustic interactions with these and other structures in the head to alter the signals driving the left and right TPCs. In this work, we constructed a three-dimensional head model using computed tomography (CT) images of a live bottlenose dolphin. Using a finite-element model to simulate sound-structure interactions, we computed how left and right TPC signals vary with sound direction for multiple frequencies to generate dolphin head-related transfer functions (HRTFs). The simulated HRTFs vary strongly with frequency. Importantly, HRTFs for sources off midline exhibit complex frequency-dependent differences, which are acoustic features that could be used to estimate sound source location. We also observed scenarios where interaural level differences (ILDs) may not be reliable directional cues. Results like these can identify which acoustic cues, at which frequencies, support robust directional hearing in toothed whales. [Work supported by ONR.]

11:45

3aABb6. Using passive acoustic monitoring to characterize vocal behaviors of sperm whales (*Physeter macrocephalus*). Cara L. Rankin (Eckerd College, 4200 54th Ave. S, St. Petersburg, FL 33711, rankincara16@gmail.com), John Joseph, and Tetyana Margolina (Oceanogr., Naval Postgrad. School, Monterey, CA)

Sperm whales (*Physeter macrocephalus*) have been observed but not well studied in the Monterey Bay National Marine Sanctuary (MBNMS). Vessels transiting through MBNMS may pose a threat to sperm whales and modify their behavior. Knowledge of their activity in relation to vessel presence in MBNMS is largely unknown. In this study, passive acoustic monitoring (PAM) data collected at the Sur Ridge, in close proximity to shipping lanes, are used to characterize vocal and foraging behaviors of sperm whales in relation to vessel transits through MBNMS. PAMGuard automated detectors are used to detect sperm whale clicks and measure inter-pulse intervals with feature-based post-processing to reduce false positives. Probability density functions are used to determine distribution of sperm whale size, sex, and presence in relation to vessel transits. Foraging clicks were positively detected in sequences of bouts from June-August, 2020. Sperm whales observed were adult males. Container ship signatures were observed to cover the same frequency bandwidth as sperm whales with a higher amplitude. This study provides insight on the importance of MBNMS to sperm whales and informs policy makers on potential effects that shipping may have on sperm whale behavior, while providing a novel method to analyze PAM data.

Session 3aBAa

Biomedical Acoustics: Lung Ultrasound

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

Chair's Introduction—8:00

Contributed Papers

8:05

3aBAa1. Correlation between lung maturity and backscatter coefficient calculations in fetal lung. Simulations. Raul Esquivel-Sirvent (Instituto de Fisica, Universidad Nacional Autonoma de Mexico, Circuito de la Investigacion S/N, Mexico, Coyoacan 01000, Mexico, sirventr@gmail.com), Irene Quiterio (Instituto de Fisica, Universidad Nacional Autonoma de Mexico, Mexico, Coyoacan, Mexico), Yubia Amaya-Guel (Instituto Nacional de Perinatología, Mexico, CDMX, Mexico), and Fabian Torres-Robles (Instituto de Fisica, Universidad Nacional Autonoma de Mexico, Mexico, Coyoacán, Mexico)

Fetal lung maturity is a public health problem, and developing quantitative ultrasonic (QUS) techniques has become a possible way of assessing lung development. To estimate the feasibility of using quantitative medical ultrasound, we performed numerical simulations of radio frequency signals scattered from lung tissue. Given the distribution of sizes of alveoli, and wall lining thicknesses corresponding to different stages of lung maturity, we calculate the effective speed of sound using an effective medium approximation. The attenuation and backscattering coefficient is calculated from many-body scattering techniques. Also, for the attenuation and the backscattering coefficient, different statistical configurations are considered to determine the correlation between size of alveoli and acoustic properties. The results using QUS are compared to light scattering techniques (1). Our simulations are compared with ultrasonic measurements made with sponges with different pore sizes following the work of . Once the measurements are analyzed, the sponges are cut and analyzed under a dark-filed microscope to obtain the average size of pores. As a proof of concept, we show that quantitative medical ultrasound can provide information on fetal lung maturity accurately. (1) M. S. Durkee, G. K. Fletcher, C. Carlson, K. Matheson, S. K. Swift, *et al*. Opt. Lett. Vol. 43, No. 20, 5001 (2018).

8:20

3aBAa2. Random matrix theory (RMT) to quantify scattering behavior in lung mimicking phantoms. Zihan Dong (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., EB3, Raleigh, NC 27606, zdong7@ncsu.edu), Azadeh D. Cole, Chukwuka W. Onuorah, Henry O. Ware (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

As we previously reported in rodent lungs, RMT parameters (Expected value *E(x)*, and $\lambda_{\max}$, the eigenvalue with the highest probability) extracted from singular value decomposition (SVD) of inter-element response matrix (IRM) show a significant correlation with fibrosis histology scores. The lack of fibrotic models for larger lungs such as pigs motivated us to investigate porous 3D printed PEGDA phantoms with controllable strut size and alveolar density. We hypothesize that *E(x)*, and $\lambda_{\max}$ can distinguish phantoms with different alveolar size, by evaluating multiple scattering in the backscattered signals. IRMs were acquired using a 128-element linear array L7-4 (Verasonics, at 5.2 MHz central frequency) connected to a Verasonics Vantage ultrasound scanner. Phantoms of 1-inches size with different strut diameters of 0.085, 0.17, and 0.26 mm were used. Attenuation constants for these samples were measured using the Substitution Method at 1 MHz, 2.25 MHz, and 5 MHz to evaluate scattering attenuation. *E(x)*, and $\lambda_{\max}$ were evaluated using SVD of the IRM. Attenuation constants, *E(x)*, and $\lambda_{\max}$ all show that the phantoms with larger strut size exhibit more multiple scattering than phantoms with smaller strut size. These preliminary results suggest that such phantoms could be used to mimic pulmonary fibrosis.

8:35

3aBAa3. Ultrasound multifrequency strategy applied to the estimation of lung surface roughness. Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Via Sommarive, 9, Povo, Trento, Trento (TN) 38123, Italy, federico.mento@unitn.it), Matteo Perini, Ciro Malacarne (Polo Meccatronica (ProM), Rovereto, Italy), and Libertario Demi (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italia, Italy)

There exist lung pathologies (e.g., COPD) characterized by an enlargement of air-spaces. Since air-spaces of different sizes can generate different levels of roughness along the lung surface, estimating the roughness may provide an indirect characterization of the lung state. This study, thus, analyzes the possibility to develop a multi-frequency quantitative approach for the estimation of lung surface roughness. Specifically, the presented technique focuses on the analysis of ultrasound image intensity variation along the lung surface as a function of frequency. *In silico* and *in vitro* results will be presented. First, k-wave was used to study the effect of different levels of lung surface roughness on numerically generated ultrasound images. Data were simulated with center frequencies from 1 to 10 MHz (bandwidth = 0.5 MHz). The same acquisition strategy was then used to acquire data from 3D printed steel models, made to mimic an acoustic interface with high reflection coefficient (similar to air-tissue interfaces) and realized with different levels of roughness. As expected, due to diffuse scattering, the increase of the "air-spaces" diameter is linked to a decrease of the frequency at which a significant drop of intensity was observed. This ultimately provides a way to indirectly estimate lung source roughness.

8:50

3aBAa4. Random matrix theory to quantify micro-structural changes in rodent lungs due to pulmonary diseases. Azadeh D. Cole (Mech. and Aerosp. Eng., North Carolina State Univ., Eng. Bldg. III (EB3) 3141, Raleigh, NC 27695, adashti@ncsu.edu), Roshan Roshankhah (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), John Blackwell (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Stephanie A. Montgomery (Pathol. and Lab Medicine, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

We exploit the random matrix theory to detect changes in rodent lungs exhibiting pulmonary fibrosis and edema. Coherences in the backscattered

signals are stronger when single scattering dominates (fibrosis/edema). On the contrary, healthy lungs exhibit more apparent randomness due to multiple scattering. This leads to differences in the distribution of eigenvalues, which can be retrieved using Singular Value Decomposition of the Inter-element Response Matrix (IRM). We use features of the eigenvalue distribution (*E(x)*, the expected value, and, the eigenvalue with the highest probability) to quantify changes in lung parenchyma and investigate whether they can improve the specificity of quantitative ultrasound to lung diseases. IRMs were acquired from 51 rat lungs (10 controls, 18 edematous, 17 fibrotic, 6 fibrotic rats, which were treated with Nintedanib) using a 128-element linear array (Verasonics L11-4v, 7.8 MHz). Severity of fibrosis and edema were quantified by histology and the ratio of wet to dry weight. Both parameters showed significant differences between edematous and fibrotic lungs, and between control and fibrotic lungs, which was significantly correlated to both the severity of fibrosis and edema. E(x) was significantly correlated to the severity of fibrosis. This suggests that these parameters could be part of a toolkit for the quantitative assessment of lung diseases.

9:05

3aBAa5. Investigating pulmonary edema in rat lungs using separation of multiple scattering and single scattering contribution. Roshan Roshankhah (Mech. and Aerosp. Eng., North Carolina State Univ., 1840 Entrepreneur Dr., Raleigh, NC 27606, rroshan2@ncsu.edu), John Blackwell, Hong Yuan (Univ. of North Carolina, Chapel Hill, NC), Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC)

Lung ultrasound imaging is challenging due to multiple scattering (MS) from alveoli. Conventional B-mode does not provide lung microstructure images. However, MS signals can provide valuable information about structure and alveolar distribution and investigating conditions such as pulmonary edema. Lung edema results in fluid buildup in interstitial spaces and alveoli, affecting density of alveoli. Previously, we demonstrated that changes in the distribution of scatterers due to edema result in changes in the wave diffusion regime and the scattering mean free path was sensitive to lung injury due to induced edema in rodents. In the present study, we introduce a novel way of quantifying MS in lungs by isolating the single scattering (SS) and MS contributions and processing them separately. After inducing different severity of edema using ischemia reperfusion injury in 18 rats, full synthetic aperture transmit sequences were used to acquire backscattered signals. The SS/MS contributions were separated using singular value decomposition. The separated SS/MS intensities were calculated and a new parameter defined as the rate of decay in intensity with depth. To assess edema severity and method validation, *ex vivo* CT lung images were assigned scores compared with this novel biomarker ($R = 0.47$, $p = 0.021$). Lung wet/dry ratio was also compared ($R = 0.52$, $p = 0.009$).

9:20

3aBAa6. Quantitative lung ultrasound spectroscopy, an *in vivo* clinical study conducted over 101 patients. Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Via Sommarive, 9, Povo, Trento, Trento (TN) 38123, Italy, federico.mento@unitn.it), Mattia Perpenti (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italy), Giuliana Barcellona, Elena Torri, Tiziano Perrone (Emergency Dept., Humanitas Gavazzeni Bergamo, Bergamo, Italy), and Libertario Demi (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italia, Italy)

Lung ultrasound (LUS) is widely adopted to assess the state of the lung surface. However, as standard ultrasound imaging assumptions are unmet in the lung due to the presence of air, essentially this organ cannot be anatomically investigated. Indeed, LUS is mainly based on the analysis of imaging artifacts (horizontal and vertical). Of particular interest is the analysis of vertical artifacts, as they correlate with several pathologies. More specifically, their dependence on frequency was demonstrated to carry important diagnostic information capable to improve LUS specificity. In this study, a new dataset was generated acquiring raw radiofrequency (RF) data from 101 patients (affected by different pathologies, e.g., COVID-19 pneumonia and cardiogenic edema). The ULA-OP research platform was used to collect the data, and a multifrequency approach implemented on both linear and convex probes was adopted. Ultrasound data were thus acquired utilizing orthogonal sub-bands with a 1-MHz bandwidth and with different center frequencies (2–6 MHz). About 12 000 and 3600 multifrequency frames were acquired with convex and linear probes, respectively. Clinical parameters (e.g., F_IO_2 and LDH) were also stored for all patients. During this meeting, the dataset will be presented together with preliminary results, and its potential applications discussed.

9:35–9:50 Break

9:50

3aBAa7. Effectiveness of transferring ultrasound deep learning models from adults to pediatrics for frame based pneumonia classification. Russell Thompson (Dept. of Comput. Sci., Worcester Polytechnic Inst., 240 Canal St., Apt 536, Lawrence, MA 01840, rthompson@wpi.edu), Umair Khan (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy), Jason Li (Dept. of Comput. Sci., Boston Univ., San Francisco, CA), Lauren Etter (Dept. of Medicine, Univ. of Wisconsin, Madison, WI), Ingrid Camelo (Pediatric Infectious Diseases, Augusta Univ., Augusta, GA), Rachel Pieciak (Dept. of Global Health, Boston Univ., Boston, MA), Ilse Castro-Aragon, Bindu Setty (Dept. of Radiology, Boston Medical Ctr., Boston, MA), Christopher Gill (Dept. of Global Health, Boston Univ., Boston, MA), Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy), and Margrit Betke (Dept. of Comput. Sci., Boston Univ., Boston, MA)

Hand-held ultrasound devices are emerging as a promising intervention to aid in diagnosing deadly early childhood pneumonia in the developing world. Lung ultrasound (LUS) data, however, can be difficult to read and interpret accurately, and thus require trained professionals. A variety of deep learning (DL) models have been developed to aid professionals in this task, but the difficulty curating quality training datasets limits the generalization capabilities of these models. To combat this data scarcity, we utilized a variety of DL models trained on LUS data collected from adult COVID-19 patients in Italy and were able to apply transfer learning to increase pneumonia related imaging pattern classification in LUS data from children and infants in rural Zambia. In this regard, we found that transfer learning significantly increased frame classification F1 performance as compared to training models from scratch. These findings indicate the promising potential for developing generalizable AI for LUS classification in diverse contexts given limited data. This analysis demonstrates that DL models effectively transfer their capabilities when fine tuned on a new population demographics, allowing the possibility to clinically deploy these models without the need to acquire large amounts of new data.

10:05

3aBAa8. Coronavirus disease 2019 patients prognostic stratification based on low complex lung ultrasound video compression. Umair Khan (Dept. of Information and Commun. Eng., Univ. of Trento, Via Sommarive, 9, Trento, Trento 38123, Italy, umair.khan@unitn.it), Sajjad Afrakhteh (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Trentino, Italy), Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Povo, Trento, Trento (TN), Italy), Andrea Smargiassi, Riccardo Inchingolo (Pulmonary Medicine Unit, Dept. of Medical and Surgical Sci., Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Francesco Tursi, Veronica Narvena (UOS Pneumologia di Codogno, Asst Lodi, Lodi, Italy), Tiziano Perrone (Emergency Dept., Humanitas Gavazzeni Bergamo, Italy, Bergamo, Italy), Giovanni Iacca, and Libertario Demi (Dept. of Information and Commun. Eng., Univ. of Trento, Trento, Italy)

In the last years, efforts have been made towards automating semi-quantitative analysis of lung ultrasound (LUS) data. To this end, several methods have been proposed with a focus on frame-level classification. However, no extensive work has been done to evaluate LUS data directly at the video level. This study proposes an effective video compression and classification technique for assessing LUS data. This technique is based on maximum, mean, and minimum intensity projection (with respect to the temporal dimension) of LUS video data. This compression allows preserving hyper- and hypo-echoic regions and results in compressing a LUS video down to three frames, which are then classified using a convolutional neural network (CNN). Results show that this compression not only preserves visual

3a WED. AM

artifacts appearance in the reduced data, but also achieves a promising agreement of 81.61% at the prognostic level. Conclusively, the suggested method reduces the amount of frames needed to assess LUS video down to 3. Note that on average a LUS videos consists of a few hundreds frames. At the same time, state-of-the-art performance at video and prognostic levels are achieved, while significantly reducing the computational cost.

10:20

3aBAa9. Synthetic lung ultrasound data generation using autoencoder with generative adversarial network. Noreen Fatima (Information Eng. and Comput. Sci., Univ. of Trento, Via Calepina, 14, 38122 Trento TN, Italy, Trento 38122, Italy, noreen.fatima@unitn.it), Riccardo Inchingolo, Andrea Smargiassi (Pulmonary Medicine Unit, Dept. of Medical and Surgical Sci., Rome, Italy), Gino Soldati (Diagnostic and Interventional Ultrasound Unit, Valle del Serchio General Hospital, Lucca, Italy), Elena Torri (Dipartimento di Emergenza ed Urgenza, Humanitas Gavazzeni Bergamo, Brescia, Italy), Tiziano Perrone (Dipartimento di Emergenza ed Urgenza, Humanitas Gavazzeni Bergamo, Bergamo, Italy), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy)

The goal of this study is to test the applicability of a generative adversarial network (GAN) to solve the class unbalancing problem in lung ultrasound (LUS) data. We introduce a supervised autoencoder with conditional latent space. During training, the generator utilizes the weights of the decoder and conditional latent space to generate synthetic LUS images, whereas the discriminator utilizes the weights of the encoder together with the class labels, which also allows classifying each synthetic image into the different classes. A customized gradient penalty loss function was utilised. This approach is tested on a dataset costing of 6500 LUS images, collected from 35 COVID-19 patients and labelled according to a validated 4 level scoring system [10.1109/TMI.2020.2994459]. 1000 synthetic images were generated to balance this dataset. The quality of the synthetic images was evaluated through similarity measures, computed with respect to training and unseen data. Moreover, the synthetic images were also evaluated by expert clinicians concerning their capability to mimic a realistic LUS image. In conclusion, the proposed approach appears to be capable to solve the class unbalancing problem by generating LUS images carrying novel information content, comparable to that of real data from new patients.

Invited Paper

10:35

3aBAa10. The vulnerable air-tissue interface: Summary of ultrasound related bioeffects on lung and their implications for the use of different imaging modes and applications beyond. Frank Wolfram (Lung Cancer Ctr., SRH Wald Clinic, Strasse des Friedens 122, Gera 07548, Germany, frankwolfram@gmx.de)

Introduction: Lung tissue is sensitive to mechanical stress. Animal studies showed induction of pulmonary capillary hemorrhage (PCH) in lung in the diagnostic regime at intensities. Therefore, this work summarizes novel findings regarding the interaction of ultrasound at the alveolar epithelium-gas interface. Method: Literature research related to ultrasound bio-effects on lung was carried out. Furthermore, recent studies reporting thresholds of PCH induction depending on imaging mode and duration will be discussed. Results: Ventilated lung tissue is more vulnerable to mechanical forces than solid tissue. From peak negative pressure below 1.5 MPa the likelihood of PCH induction increases with exposure time. Such PCH is asymptomatic but can falsify the diagnostic outcome. The causes remain still unknown but is not related to cavitation or thermal effects. The extent of PCH in lung is depending of physiological beside acoustic parameters. The extend of PCH is, therefore, not entirely predictable. In addition, observational studies on humans could not confirm the findings as investigated on pathological level on animal models. Conclusion: The vulnerability of lung should become aware in the lung ultrasound community. Current recommendations for output limitations given by AIUM and EFSUMB should be followed and only increased for diagnostics needs.

Session 3aBAb

Biomedical Acoustics and Education in Acoustics: Best Practices in Mentoring for Biomedical Acousticians

Julianna Simon, Cochair
Penn State University, 201E Applied Science Bldg., University Park, PA 16802

Kevin J. Haworth, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

Chair's Introduction—8:00

Invited Papers

8:05

3aBAb1. Graduate student mentoring by Lawrence A. Crium. Lawrence A. Crum (Appl. Phys. Lab, Univ. of Washington, 4662 175th Ave. SE, Bellievue, WA 98006, lacuw@uw.edu)

For a period in excess of 55 years, I have had the opportunity and privilege to serve as an advisor to over 50 students at various levels of their academic instruction and development, particularly at the graduate and postgraduate level. Many of my students have achieved noted success in their careers and have served as Society Presidents, as Deans of Engineering schools, and Directors of Government Labs. Having no established approach to student mentorship, this success has been one of selecting good students with a potential for success in any field and providing them with an opportunity to develop their own careers. Indeed, my style has been one of "benign neglect," if any. Nevertheless, I will attempt in this lecture to describe what I see is an approach that has resulted in successful career development for my students.

8:25

3aBAb2. Help your students and postdoctoral fellows realize their worth. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Hallmarks of a wonderful lab include exciting projects, collaborative publications, and students and postdoctoral fellows who have found good jobs. The head of the lab knows how to and is willing to pass along skills, guidance, and networking opportunities for regional and national recognition to other members of the lab. Strategies will be discussed for providing access to funded grant proposals, manuscripts in preparation, and lab protocols. A "one-pager," which includes career goals, research goals, and planned time off, can be solicited and posted digitally for open access by lab members. Additionally, each student and postdoctoral fellow can be encouraged to write an individual development plan that identifies career goals, objectives necessary for achieving career goals, professional development needs, and progress toward achieving career goals. Examples of a "one-pager" and an individual development plan will be discussed. Mentoring students and postdoctoral fellows is one of the most important things that can be accomplished, as the impact will be long term.

8:45

3aBAb3. Mentoring through an inclusive lens. Tyrone M. Porter (Biomedical Eng., Univ. of Texas at Austin, 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu)

There is a great deal of building equitable and inclusive teams in STEM professions, but there are limited resources on how to achieve this goal. Today I will share on my mentoring philosophy and my efforts to lead a research team that values diversity, equity, and inclusion.

9:05

3aBAb4. Jack of all trades: Challenges and opportunities for mentoring in interdisciplinary research. Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Biomedical acoustics spans a particularly broad range of disciplines from neuroscience to quantum physics and everything in between. While this diversity is exciting and constantly stimulating, it can pose considerable challenges for students and mentors in finding the appropriate balance between breadth and depth. The aim of this talk is to share the author's experience of designing graduate and undergraduate courses in biomedical acoustics and particularly of supervision and team building for graduate and post-doctoral researchers from different disciplines. Key topics will include the importance of co-supervision, promoting understanding and implementation of the scientific method; encouraging curiosity driven experiments; vital lessons from science communication; and finally the surprising insights provided by training to teach dance.

Contributed Paper

9:25

3aBAb5. Value of multidisciplinary peer mentoring. Kevin J. Haworth (Dept. of Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3950, Cincinnati, OH 45267, hawortkn@ucmail.uc.edu), Carl J. Fichtenbaum, Eric P. Smith (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Nives Zimmerman (Dept. of Pathol. & Lab. Medicine, Univ. of Cincinnati, Cincinnati, OH), and Margaret V. Powers-Fletcher (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Biomedical acoustics is a multidisciplinary field. As such, both the scientific field and its members have the potential to benefit from the interactions and cross-fertilization of different subfields. Peer mentoring groups (PMGs) can promote multidisciplinary interaction. A PMG involves the intentional, structured, and consistent interaction of individuals at similar career stages. The horizontal organization of a PMG can promote a safe and supportive environment, enabling greater knowledge sharing. PMG knowledge sharing is bidirectional, providing benefits to all participants and an opportunity to leverage diversity, which yields outcomes. In one example from our institution, a research-active group of junior faculty from different academic divisions within a department of internal medicine met biweekly to review, on a rotating basis, active grant applications, critiques of rejected proposals, and career development challenges. Another example was an interdepartmental graduate program where doctoral students studying a range of fields (e.g., infectious diseases, cancer, metabolism, and cardiovascular disease) engaged in a journal club and a seminar series. In both settings, participants had the opportunity to benefit from the recognition of thematic and methodological overlap in their otherwise disparate studies, which promoted best practices in research. In summary, PMGs provide value to both participants and the field.

9:40–10:00 Break

Invited Papers

10:00

3aBAb6. Lessons learned from mentoring and being mentored in a large environment. Meaghan O'Reilly (Sunnybrook Res. Inst., 2075 Bayview Ave., Rm C736a, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

The Focused Ultrasound Lab at Sunnybrook Research Institute has over 60 regular members and trains an additional 30 + summer students each summer. Drawing on the speaker's experience, as well as trainee data, this talk will focus on approaches to effective mentoring in a large environment. The role of mentor teams within biomedical acoustics and in other related fields will be discussed.

10:20

3aBAb7. Structured mentoring process. Hong Chen (Washington Univ. in St. Louis, 6338 Washington Ave. University City, MO 63130, chenhongxjtu@gmail.com)

I am fortunate to have great mentors who believe in me and encourage me to reach beyond my self-doubts and make it possible for me to achieve more than I initially thought possible. I am now a professor, and how to be a great mentor has always been a question that I ask myself. Over the past seven years, I have developed a structured mentoring process to guide my practice. This structure has four phases: (1) 1–4 months to decide whether there is a good fit between the mentor and mentee; (2) 4–12 months to build the relationship by working closely with mentees to understand their career goals, passions, strengths, and weakness and working with mentees to formulate mentee development objectives; (3) 1–4 years to maintain the momentum of mentoring and regularly (every semester) check on the progress in mentoring; (4) End formal mentoring when mentees leave the lab but maintaining lifelong support to mentee's career development. I will share how this structured mentoring process works in my lab and solicit feedback and suggestions on how to improve this process.

10:40

3aBAb8. Mentoring from the other side of the fence: Student perspectives on good advising. Megan S. Anderson (George Washington Univ., 800 22nd St NW, Washington, DC 20052, andersonm@gwmail.gwu.edu)

As a fourth-year graduate student and ASA Student Council representative, I have had access to the opinions of students and gained insight on what they expect from their mentors and advisors. In this talk, I hope to demystify the student perspective by discussing the themes that have emerged from both Blackstock Mentor Award applications and a mentoring survey sent to Biomedical Acoustics students. One question drives the analysis: What kind of mentor/mentee relationship leads to the most successful outcomes and how is that success measured by students?

Contributed Paper

11:00

3aBAb9. A four-prong approach for engaging biomedical engineering students. Thomas L. Szabo (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

My involvement in a large medical imaging company for nearly 20 years shaped my approach to teaching biomedical engineering. These experiences are described in Medical Physics International Journal, 6,602–620, May 2021. At Boston University, I taught undergraduates using a four pronged approach. The first part included a "lessons learned" career overview lecture and discussion to incoming freshman. This talk emphasized shifting paradigms in medical device development and changing trends in research and development and how to best prepare for a biomedical engineering career by prudent course selection, practical experience and life-long networking. Second, I designed two ultrasound lab modules which were part of the mandatory physiology lab all juniors took. These laboratories provided condensed theory with hands-on experiments on imaging and Doppler processing/imaging and data analysis. Third, I taught ultrasound and medical imaging courses which included rigorous theory, applications and examples. My book, Diagnostic Ultrasound Imaging: Inside Out, second edition, is being used in ultrasound courses throughout the world. A new book and curriculum, coauthored with Peter Kaczkowski, and sponsored by Verasonics, is underway. Finally, for 15 years, I taught a usually two semester course in entrepreneurship, innovation and engineering design which culminated in a team-based senior project.

WEDNESDAY MORNING, 10 MAY 2023 LINCOLNSHIRE 1/2, 8:45 A.M. TO 10:45 A.M.

Session 3aCA

Computational Acoustics: Topics in Computational Acoustics

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

Contributed Papers

8:45

3aCA1. Specialized finite element basis functions for mid-frequency simulations. Benjamin M. Goldsberry (Appl. Res. Labs. at The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu)

Finite element analysis (FEA) has over the years become a common numerical tool for simulating time and frequency acoustic wave scattering and radiation problems due to its flexibility to model multiphysics coupling and complex geometries. For mid-frequency acoustic simulations, the computational mesh must be sufficiently fine to represent the wave field and the geometry, which becomes computationally expensive and impractical to use. Recent works in partition of unity FEA and isogeometric analysis have shown that using smooth basis functions that well-represent the local solution typically produce increased accuracy per degree of freedom compared to traditional FEA with Lagrange basis functions. In this work, the concept of designing accurate basis functions that can be easily incorporated into a finite element framework is explored. This process makes use of the Helmholtz-Kirchhoff integral defined along the edges of the mesh to locally represent the solution in the interior of each element. Therefore, this method only requires degrees of freedom on the element boundaries and does not require any *a priori* knowledge of the local solution for enrichment. Numerical dispersion results will be presented, and the practical implementation into FEA codes will be discussed.

9:00

3aCA2. Correlation of electric field and acoustic output on a prismatic biaxial transducer: A finite element study. Javier D. Salazar Cabral (Escuela de Ingeniería y Ciencias, Tecnologico de Monterrey, Pedro Guzman Campos 17 Torre 27, Depto. 302, Rincon de la Montaña, Atizapan de Zaragoza, Mexico 52975, Mexico, javiersalazarcabral@gmail.com), J. E. Chong-Quero (Escuela de Ingeniería y Ciencias, Tecnologico de Monterrey, López Mateos, Mexico State, Mexico), Sagid Delgado, and Samuel Pichardo (Radiology, Univ. of Calgary, Calgary, AB, Canada)

In single multiaxial transducers, the ultrasound beam can be steered by shifting the phases of the signals of two or more orthogonal electric fields applied to the piezoelectric. These capabilities can improve focusing on therapy applications such as transcranial focused ultrasound. A biaxial piezoelectric transducer (DL47, DelPiezo, FL) was modelled using the finite element software COMSOL Multiphysics®. The upper face was in contact with a volume of water, while the rest were free. The steering angle was controlled by varying the difference between the electrodes' phases at the crystal's resonant frequency. To reduce bias, the acoustic particle velocity was recorded at various lengths above the upper transducer's face while the average electric field was calculated for different areas on the surface. Linear regression fits of the steering angle with respect to the average electric field and particle velocity were calculated. A correlation can be observed between the average electric field at the center of the piezoelectric and the

steering angle. Furthermore, by reducing the recorded length of the particle velocity by 30%–40% there is a high correlation with the steering angle. A model can be derived using these correlations in which the ultrasound beam direction can be known by obtaining the electric field at the center of the crystal. [The authors want to acknowledge the support of the Government of Canada by the scholarship provided by Global Affairs Canada through the Emerging Leaders in the Americas Program.]

9:15

3aCA3. Numerical and Theoretical Investigations on Twin Standing-wave Thermoacoustic Prime Movers. Lixian Guan (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, dan.zhao@canterbury.ac.nz)

In this work, we perform computational fluid dynamics investigations to predict the performance of the twin thermoacoustics engine (TAE) under various working gas/medium and at different charge pressures. The numerical model is validated by comparing with those of DeltaEC (Design Environment for Low-amplitude ThermoAcoustic Energy Conversion) predictions and previous experimental results. Additionally, the effects of (1) the geometry shape of the twin TAE and (2) different types of working gases on its performance are examined. The results indicate that the working gas/medium and charge pressure in the engine are critical to the twin thermoacoustic engine. Although the acoustic pressure amplitude is higher with air or nitrogen as the working gas, the acoustic oscillation frequency is higher with helium or argon and the value of the acoustic growth rate is larger, i.e., the limit cycle is observed growing faster in the twin TAE. Furthermore, the pressure-ratio is increased in the tapered resonator, because its velocity amplitude is decreased. It means the tapered resonator is the best choice for the twin TAE. The acoustic characteristics predicted are much closer to the experimental results than those of DeltaEC estimations, which provides a design tool for designing and predicting performances of twin TAEs.

9:30

3aCA4. Computational aeroacoustics in flexible conduits with application to biomedical fluid dynamics. Marco Laudato (Eng. Mech., KTH Royal Inst. of Technol., Osquars Backe 18, Stockholm 11428, Sweden, laudato@kth.se), Mihai Mihaescu, Elias Zea, and Susann Boij (Eng. Mech., KTH Royal Inst. of Technol., Stockholm, Sweden)

Flexible tubes are simple yet powerful tools for the modeling of respiratory and circulatory systems [1, 2]. In the last decade, access to high-performance computational resources has allowed the implementation of realistic numerical models of human vessels based on Computational Fluid Dynamics (CFD) and Fluid-Structure Interaction (FSI) simulations. Interestingly, the recently proven correlation [3] between CFD observables (like the pharyngeal airway's resistance) and clinical data (such as the Apnoea Hypopnea Index in Obstructive Sleep Apnoea patients) suggests a possible development of diagnostic methods based on numerical simulations. This work aims to investigate the extension of these correlations to the aeroacoustics characteristics of flexible conduits by means of a fully coupled FSI numerical model based on the Large Eddy Simulation method. These results have relevant applications to studying diseases of the human upper vocal tract, voice production, obstructive sleep apnoea, and adventitious lung sounds, such as wheezing and crackling. [1] Grotberg, J. B., & Jensen, O. E. (2004). *Annu. Rev. Fluid Mech.*, **36**, 121–147. [2] Schwartz, A. R., & Smith, P. L. (2013). *The Journal of physiology*, *591*(Pt 9), 2229. [3] Schickhofer L., Malinen J., Mihaescu M., *J. Acoust. Soc. Am.* - JASA, **145** (4): 2049–2061, 2019. https://doi.org/10.1121/1.5095250

9:45

3aCA5. Fast and accurate modeling of parametric array loudspeakers in the frequency domain. 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) and Yun Jing (Graduate Program in Acoust., Pennsylvania, State College, PA)

The parametric array loudspeaker (PAL) is known for its ability to generate highly directional audio beams. It has been widely used in a variety of audio applications, such as the sound field reproduction and active noise control. Fast and accurate computations of the audio sound field generated by a PAL are challenging due to the complexity of nonlinear interactions between airborne ultrasonic waves. This paper presents an overview of the commonly used computational methods in the frequency domain under the quasilinear assumption, which include the direct integration method, convolution directivity model, Gaussian beam expansions, cylindrical expansions, spherical expansions, and finite element methods. The computational efficiency and the accuracy of these methods in the near field, Westervelt far field, and the inverse-law far field are compared. Future challenges and perspectives are discussed.

10:00–10:15 Break

10:15

3aCA6. Numerical calculations with an exact analytical time-domain Green's function for the van Wijngaarden wave equation. Vaughn E. Holmes (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI) and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, Michigan State University, East Lansing, MI 48824, mcgough@egr.msu.edu)

An exact analytical time-domain Green's function is obtained for the van Wijngaarden wave equation when two of the constant coefficients satisfy a certain equality. This analytical expression enables numerical assessments demonstrating that lossy time-domain Green's function calculations with the inverse fast Fourier transform (IFFT) produce larger errors than expected for most temporal values, even when the IFFT is evaluated with several million points. This analytical time-domain Green's function also provides the foundation for efficient numerical calculations of spatial impulse responses, which are demonstrated for a 1 cm radius circular piston radiating in water at axial distances of 1 cm, 10 cm, and 1 m. As the axial distance increases, the difference between the lossy and lossless spatial impulse responses also increases, which is particularly evident in the paraxial region. However, the difference between lossy and lossless spatial impulse responses diminishes as the observation point transitions from an on-axis location to another position with the same axial distance just outside of the paraxial region. The observed differences between the lossy and lossless spatial impulse responses transitioning out of the paraxial region are explained by comparing the relative temporal extents of the lossless spatial impulse response and the lossy time-domain Green's function.

10:30

3aCA7. Infrasound propagation from a vertically extended source. Roger M. Waxler (Univ. of MS, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu), Bin Liang (Univ. of MS, Oxford, MS), and Claus Hetzer (Univ. of MS, Tempe, AZ)

The propagation of infrasound from a vertically extended source is investigated. The source is allowed to have azimuthal and vertical variation. The radial extent of the source is assumed to be small compared to an acoustic wavelength. No such assumption is made for the vertical extent. The problem is treated in cylindrical coordinates, as is appropriate for outdoor sound propagation, in a way analogous to the treatment of radiation from a compact source in unbounded three dimensional space. Several equivalent formulations will be presented along with example cases.

Session 3aEA

Engineering Acoustics, Signal Processing in Acoustics, Underwater Acoustics, Physical Acoustics, and Biomedical Acoustics: Additive Manufacturing of Acoustic Transducers

Michael R. Haberman, Cochair
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Thomas E. Blanford, Cochair
The Pennsylvania State University, The Pennsylvania State University, State College, PA 16804

Chair's Introduction—9:00

Invited Papers

9:05

3aEA1. Unnatural hearing—3D printing functional polymers as a path to bio-inspired microphone design. Andrew Reid (Elec. and Electron. Eng., Univ. of Strathclyde, 99 George St., Glasgow G1 1RD, United Kingdom, andrew.reid@strath.ac.uk)

In nature, auditory organs are rarely passive transducers of their environment. Highly localized material properties and complex interdependencies mechanically filter the acoustic signal, reducing the burden of signal processing to the nervous system. Capturing these design traits in engineered systems is important for device miniaturization and energy efficiency, but manufacturing a functional electromechanical device, like a microphone, at the microscale using biologically inspired 3D designs and highly anisotropic materials remains extremely challenging. Our research uses one-pot synthesis methods for a variety of tissue-like hydrogels, polymers, and functionalized piezoelectric composites that are compatible with vat-based photopolymerization 3D printing. These materials can be used to produce reproductions of microphone designs using functional photopolymers for the conductive, piezoelectric, elastomeric, and structural elements. Used to produce mimetic structures, for example, in a device based on the directional sensitivity of the parasitoid fly *Ormia ochracea*, we can reproduce *O. ochracea*'s sound localization capability while addressing the impracticalities in frequency, sensitivity and scale of MEMS or micromachined designs. Each of the materials used has a unique set of acoustic, electrical and mechanical properties which can be tailored by altering the synthesis process leading to a truly vast design space which we are only beginning to explore.

9:25

3aEA2. Direct ink writing of textured piezoelectric ceramics. Chloe Fellabaum (Mater. Sci. and Eng., Penn State, 201 Old Main, State College, PA 16802, caf5739@psu.edu), Christopher Eadie (Appl. Res. Lab., Penn State Univ., State College, PA), and Mark Fanton (Appl. Res. Lab., Penn State Univ., Freeport, PA)

Textured ceramics are often formed via tape casting in order to align platelet particles in the preferred direction, allowing the ceramic to have properties approaching those of single crystals without the complexities of growing such a crystal. However, tape casting produces significant waste material, as flat plates are machined to produce the desired geometries, e.g., rings. Additive manufacturing reduces these issues by allowing the complex geometries to be formed in the green state, thus reducing waste and improving design freedom. This presentation will discuss progress to-date on direct ink writing (DIW) of textured PMN-PZT. The focus will be on the methodology used to develop a suitable paste for printing. Development of a stable, shear thinning paste with a quick recovery time is critical. Zeta potential and rheological data were collected as a function of pH and other relevant slurry parameters such as percent solids loading and binder content. Sintered filaments and bulk parts were characterized to evaluate the efficacy of template alignment in the process as a function of slurry rheology, nozzle geometry, and print parameters.

9:45

3aEA3. The design and evaluation of additively manufactured piezoelectric acoustic transducers. Justin Tufariello (The MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, jtufariello@mitre.org), Barry Robinson (MSI Transducers Corp., Littleton, MA), Shawn Allan (Lithoz America, LLC, Troy, NY), Casey Corrado, Alex Angilella (The MITRE Corp., Bedford, MA), and Brian Pazol (MSI Transducers Corp., Littleton, MA)

Ceramic additive manufacturing (AM) has been demonstrated as a capable method to fabricate piezocomposite acoustic transducers that exhibit augmented sensitivity, increased bandwidth, and controlled directivity by virtue of geometry. These improvements are intrinsic to the novel piezoelectric ceramic structure, which may be printed in forms that cannot be fabricated through conventional manufacturing methods. The AM process allows for functionally-graded apertures, 3-3 piezocomposite structures, and auxetic lattices which have been simulated, printed, and measured with compelling results. This presentation expands upon previous work completed through a collaboration between Lithoz America, LLC (Lithoz), MSI Transducers Corp. (MSI), and the MITRE Corporation (MITRE) to produce novel piezoelectric structures and validate acoustic and hydrostatic pressure performance of 1-3 piezocomposite structures under operationally relevant conditions. The presentation will encompass a discussion of Finite Element Analysis (FEA) of piezoelectric metamaterials conducted at MITRE, the lithography-based ceramic manufacturing (LCM) process of said structures at Lithoz, and empirical measurements of piezoelectric and acoustical specifications of the printed structures conducted at MSI. Finally, test data recorded at Woods Hole Oceanographic Institute (WHOI) will be shown, demonstrating the resilience of AM 1-3 piezocomposite under 10 000 PSI of hydrostatic pressure derived by means of *in-situ* electrical impedance measurements.

10:05

3aEA4. Template alignment optimization in additively manufactured piezoelectric ceramics. Benjamin Cowen (Penn State Univ., University Park, State College, PA 16801, bmc6220@arl.psu.edu), Christopher Eadie, Jules Lindau, and John Mauro (Penn State Univ., State College, PA)

Sonar transducer performance is greatly impacted by the microstructural alignment of the ceramics from which they are fabricated. Textured ceramics are a desirable material source for these parts because their deliberately aligned microstructures allow for tailored anisotropic properties that rival those of single crystal ceramics, but they also maintain the mechanical robustness and bulk manufacturability of polycrystalline ceramics. Current methods for manufacturing textured ceramics, e.g., tape casting, severely limit design freedom and require excess material waste, and so the advantage of texturing has not been fully realized. This study is focused on the enhancement of direct ink writing of textured ceramics. Direct ink writing is an additive manufacturing (AM) technique that enhances design freedom on a macro and micro scale and reduces waste by producing near-net shape textured ceramics. However, shear stresses during the AM process affect the microstructural alignment of the ceramic in an extremely complex manner that is difficult to control directly. This study first validates a novel alignment metric, derived from a computational fluid dynamics model that simulates the printing process, against real-world data. Then, an optimization algorithm is used to maximize alignment with respect to the nozzle geometry.

10:25–10:40 Break

10:40

3aEA5. Scaling considerations for additive manufacturing of textured piezo-ceramics for acoustic transduction. Mark Fanton (Appl. Res. Lab., Penn State Univ., 559A Freeport Rd., Freeport, PA 16229, maf146@arl.psu.edu), Chloe Fellabaum, Jules Lindau (Appl. Res. Lab., Penn State Univ., State College, PA), and Christopher Eadie (Appl. Res. Lab., Penn State Univ., Freeport, PA)

Textured piezo-ceramics opened the door to an expanded design space for acoustic transducers by shifting from a random microstructure to one that is highly aligned with respect to important crystallographic orientations. This alignment results in substantial increases in critical parameters such as displacement, coupling, and voltage sensitivity. On a laboratory scale, this is easily achieved via tape casting. However, scaling to large quantities, large parts, and complex geometries is limited by this approach. Additive manufacturing presents an opportunity to impact production speed as well as part shape and geometry. We will present a comparison of conventional tape casting technology to additive manufacturing, with a focus on direct ink write technology and its benefits. The ability to produce high texture fractions, achieve performance equivalent to tape casting, and create shapes such as hemispheres and cylinders will be discussed in comparison to conventional ceramic processing technology. The potential uses of this technology in a manufacturing environment will also be addressed for both textured and conventional piezo-ceramics.

11:00

3aEA6. Fabrication of optimized transducer head masses via additive manufacturing. Christopher Eadie (Appl. Res. Lab., The Penn State Univ., University Park, State College, PA 16801, cde139@psu.edu), Scott Brumbaugh, Chloe Fellabaum, Thomas Cleckner, and Richard J. Meyer (Appl. Res. Lab., The Penn State Univ., State College, PA)

The Applied Research Laboratory conducted an investigation into additive manufacturing (AM) as an avenue to improve transducer head mass performance. Techniques, such as powder bed fusion (PBF) of metals and direct ink writing (DIW) of ceramics, offer benefits, including increased design freedom and faster design iteration. The transducer head mass acts as an acoustic radiator and receiver, and is required to have a high stiffness-to-weight ratio. Aluminum is commonly used for this application due to a balance between performance and cost. AlSi10Mg, an aluminum alloy, is readily printed via PBF, making it an excellent candidate for geometric optimization. A technique known as topology optimization has been used to great effect, reducing weight of the part and bridging the performance gap between aluminum and AlBeMet. High performance composites and ceramics are sometimes used for head masses, however their high price and long turnaround time often eliminate them as an option. Recently, DIW has been used to manufacture boron carbide head masses. DIW reduces the need for expensive machining, reducing the cost and time required to employ such a high performance material. This presentation will discuss the optimization of the transducer head mass, including topology optimization of AlSi10Mg and DIW of B_4C.

Contributed Paper

11:20

3aEA7. 3D printed acoustic metamaterial filters for the mitigation of inaudible ultrasound attacks on smart speakers. Joshua Lloyd (Dept. of Mech. Eng., Rowan Univ., Glassboro, NJ, lloydj39@students.rowan.edu), Cole Ludwikowski (Dept. of Mech. Eng., Rowan Univ., Glassboro, NJ), Derek Phansalkar, Cyrus Malik (Dept. of Elec. and Comput. Eng., Rowan Univ., Glassboro, NJ), and Chen Shen (Dept. of Mech. Eng., Rowan Univ., Glassboro, NJ)

Acoustic metamaterials have shown great potential in manipulating acoustic waves with a small footprint and versatile functionality. While their usefulness has been demonstrated in many fields including noise reduction and biomedical ultrasound, not many studies have been done in the field of audible acoustics and speakers. In this work, we show that an acoustic metamaterial-based filter can enhance the security of smart speakers, which are susceptible to inaudible ultrasound attacks due to the shadow effect of the microphones. The filter contains several resonant unit cells which collectively modulate the received signals in the ultrasound spectrum and, thus, mitigate inaudible attacks on smart speakers. On the other hand, normal audible signals are not affected and, therefore, regular functionality is not disturbed. The filtering effect is verified numerically using finite element analysis. Measurements are performed to validate the concept, where it is shown that inaudible ultrasound attacks are effectively blocked when the 3D-printed metamaterial filters are installed. This demonstration shows a possible way to apply acoustic metamaterials in consumer electronics such as smart speakers.

WEDNESDAY MORNING, 10 MAY 2023 INDIANA/IOWA, 8:40 A.M. TO 11:00 A.M.

Session 3aMU

Musical Acoustics: Acoustics of Stringed Instruments

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

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

Chair's Introduction—8:40

Invited Papers

8:45

3aMU1. Coupled simulation of vibration and sound field of Stradivari violin. Masao Yokoyama (Information Sci., Meisei Univ., 2-1-1 Hodokubo, Hino 191-8506, Japan, masao.yokoyama@meisei-u.ac.jp), Amane Takei (Univ. of Miyazaki, Miyazaki, Japan), Ryuji Shioya, and Genki Yagawa (Toyo Univ., Kawagoe, Japan)

The aim of this study is to clarify the relationship between the vibration of violins made by Antonio Stradivari and the acoustic radiation around it. The highly precise geometry of the violin was determined with a micro-computed tomography scanner, and the material properties of the violin body (spruce and maple) were set as the orthotropic properties in the numerical simulation. Then, the numerical simulations coupling the vibrations by a forced oscillation on a bridge with the acoustic field pressure around the violin were performed. Furthermore, the sound pressure emanating in a huge rectangular box was successfully calculated by large scale computing and the open-source parallel acoustic analysis software.

9:05

3aMU2. Material variability compensation in a finite element model of a classical guitar by appropriate geometry modifications. Alexander Brauchler (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 sound of some old master instruments like the Stradivari violins or the Torres guitars is still unrivaled for many musicians. The common practice among instrument makers is to copy these instruments by building instruments with the same geometry as the original. However, audible differences between these copies and the original instruments are unavoidable, mainly due to the natural variability of the wood characteristics. A suitable measure to objectively characterize an instrument's sound is evaluating its modal parameters. Our project aims to develop a methodology to design copies of the old master instruments that match these modal parameters of the original instruments instead of copying their geometry. After showing that such an approach is appropriate for an individual guitar soundboard in a recent publication, we now present a parameter study to find suitable geometric modifications on an entire guitar to compensate for the natural material variation. Therefore, this study is based on a detailed finite-element model that shall predict the effect of suitable geometry alterations on the instrument's modal parameters and suited optimization techniques. Geometric modifications like the shape of the braces, the thickness of the fretboard, and the thickness of the plates are taken into account.

9:25

3aMU3. Archtop guitar dynamics—*Continuing research of archtop guitar acoustics.* Thomas Nania (D'Addario & Co., 595 Smith St., Farmingdale, NY 11735, houseofluthiery@gmail.com)

The contemporary archtop guitar, designed by Lloyd Loar of the Gibson Mandolin-Guitar Mfg. Co. Ltd., will be 100 years old in 2023. Since the first L5 shipped in 1923, the archtop guitar has taken on a life of its own, having been reinvented by luthiers worldwide. Research of the archtop guitar is a new field in instrument acoustics. Measurement techniques developed by the bowed instrument research community have been adapted for guitar to offer a glimpse into the acoustic behavior of a century old, yet timeless guitar design. This talk will focus on the ongoing research that has been conducted since the 2021 publication of Archtop Guitar Dynamics by the Violin Society of America Papers. The archtop guitar shares features of violin family instruments, including a carved top, ff-hole perforations, and raised bridge. Beyond aesthetics, the archtop shares acoustic similarities to both violin and guitar family instruments resulting in a unique acoustic signature. Continuing research includes analysis of historically significant instruments from Gibson Mandolin-Guitar Mfg. Co. Ltd and the works of New York luthier, John D'Angelico as part of the Archtop Project: New York sponsored by D'Addario & Co. and the Archtop Foundation.

Contributed Papers

9:45

3aMU4. Factors that affect guitar timbre. Gordon P. Ramsey (Phys., Loyola Univ. Chicago, Phys. Dept., Loyola University Chicago, Chicago, IL 60660, gramsey@luc.edu) and Jessica R. Moore (Phys., Loyola Univ. Chicago, Chicago, IL)

This is part one of two presentations on the same project. Guitars have been used for centuries, but their versatility due to technical developments have made them very popular in most genres of music. Our research has investigated how various physical properties of the guitar affect its timbre. We have investigated the effects of (a) string excitement location, (b) string type and gauge, (c) pick versus pluck excitement, and (e) the placement of a capo on an acoustic guitar. The main goal is to investigate how each of these factors play a role in certain genres of music where guitars are one of the main instruments. Measurements were taken at the LU acoustics lab and the anechoic chamber at Northwestern. This talk will present theoretical basis for our measurements and discuss the effects of the pick versus pluck and placing a capo on the neck of the guitar. The following talk will discuss the remaining factors and how our study contributes to the capability of the player to create different moods.

10:00

3aMU5. How musicians can utilize the guitar's acoustic properties. Jessica R. Moore (Phys., Loyola Univ. Chicago, 1032 W Sheridan Rd., Chicago, IL 60660, jrmoore482@gmail.com) and Gordon P. Ramsey (Phys., Loyola Univ. Chicago, Chicago, IL)

Guitars are extremely versatile instruments due to the myriad of ways in which the player can alter the timbre. Specific adjustments are made to the guitar that correspond to stylistic distinctions between genres of music. This study explores how the timbre of the guitar is affected by the string type, gauge, and excitement location, in addition to the previous talk topics. Using the spectra, we can determine how our results can be utilized by musicians. Continuing from the first talk on the physical implication of these alterations, I will focus on how the results justify guitarist's artistic decisions in Flamenco, Irish Traditional, and country/folk music. Understanding the connection between the technical and artistic aspects of the guitar can guide a musician in creating the effects appropriate for each genre.

10:15

3aMU6. Examination of the static and dynamic forces at the termination of a bowed string. 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), Alexander Mayer (Dept. of Music Acoust. (IWK), mdw – Univ. of Music and Performing Arts Vienna, Vienna, Austria), Montserrat Pàmies-Vilà (Dept. of Music Acoust. (IWK), mdw – Univ. of Music and Performing Arts Vienna, Vienna, Österreich, Austria), and Vasileios Chatziioannou (Dept. of Music Acoust. (IWK), mdw – Univ. of Music and Performing Arts Vienna, Vienna, Austria)

In bowed-string instruments, the force exerted by the string on the bridge may be used for analyzing the excitation process. This force is caused by the vibration of the string due to the musician's bowing action. The dynamic component of this force has been investigated in the relevant literature. In this study, focusing on a monochord, an approach is proposed to simultaneously measure both the static and the dynamic components of the force exerted by the string on a rigid support. The use of a robotic arm to excite the string via a linear bowing action permits to measure this force under controlled excitation conditions. The static and dynamic components of the force are analyzed in the transverse plane to the string at several moments of the bow stroke, as well as for selected bowing positions and bow velocities.

10:30

3aMU7. Measured changes in the bridge mobility and radiativity of violins due to material creep following tensioning of strings. Seth Lowery (Phys., Central Washington Univ., 400 E. University Way, Dept. of Phys., Ellensburg, WA 98926, seth.lowery@cwu.edu) and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

Violins have been observed to undergo a small, but measurable, deformation of the corpus after being brought into tune following a prolonged period without string tension. This behavior, which is analagous to the flexing of a bow, can be attributed to the viscoelastic properties of wood. Because of the relatively long time scale of deformation, the process is sometimes referred to as material "creep." To assess the impact of creep on the sound of a violin, we measured the vibrational and acoustic response of multiple violins during this period of deformation. The measurement process involves tapping the bridge with a small modal impact hammer and recording the velocity response of the top plate using a laser Doppler vibrometer, as well as the far-field acoustic response in an anechoic chamber. Results show modest changes in the amplitude of the main body modes, most notably the CBR and B1- modes, and a consistent shift in the peak frequency of higher modes, over a two-day period after the strings were tuned.

10:45

3aMU8. Acoustic effects of varying the waist width of violin bridges: A physical experiment. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com) and Nathaniel Taft (Taft Violins, Madison, WI)

We explore the relationship between the bridge waist and sound production in the violin. Models of violin acoustics suggest that the stiffness of this portion of the bridge has a considerable effect on an instrument's sound production. In practice, the proportions of the waist vary significantly from bridge to bridge and instrument to instrument, which suggests there are many ways that a luthier can influence a bridge's stiffness. Our experiment connects modeling and real-world instrument sound adjustment. We incrementally remove material symmetrically from the bridge in the waist area, recording sound production at each increment. The bridges are destroyed in the process, which is repeated across multiple bridges. Sound stimuli include a sine wave at the bridge with dampened strings and a weighted bow drawn across freely ringing strings. We map changes in bridge geometry to changes in maximum output and acoustic measures of tone quality across several instruments. Since stiffness correlates with thickness, subject to the nature of the bridge material, there may be optimal shape adjustments for maximizing sound production, tone quality, or both.

WEDNESDAY MORNING, 10 MAY 2023 CHICAGO C, 8:00 A.M. TO 10:25 A.M.

Session 3aNS

Noise, Architectural Acoustics, Underwater Acoustics, Engineering Acoustics, and Acoustical Oceanography: Post Pandemic Soundscapes Part I (Hybrid Session)

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. The meaning of post-pandemic soundscapes. Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath 52134, Germany, bschulte_f@web.de)

Following the pandemic, current times classified as post-pandemic have to be seen from a multidisciplinary view point when it comes to soundscape research. During the pandemic, it was under duress, that worldwide daily life was changed and new habits had to be developed as for example according to home office, home schooling, family life in narrow space, lock downs. At the beginning, it was causing irritation and helplessness, but later it was changed to habits in regard to so called daily routines. Following, post-pandemic soundscape research must be related to daily life in its status and noticeable changes. Post-pandemic soundscapes will show changes for examples in regard to pioneering city-planning involving the concepts of smart growth and smart cities. Taking this into account, soundscape research in post pandemic times has to get new technologies under scrutiny as well as new approaches in soundscape methodologically also including virtual reality. Expected is a revised perception of the acoustic environment based on collected experiences and knowledge of people concerned. The review of the respective soundscape research will summarize the current situation in post-pandemic times.

8:25

3aNS2. Are new requirements for soundscape research resulting from the pandemic? Andre Fiebig (Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, andre.fiebig@tu-berlin.de)

The pandemic caused short-term but also medium-term changes in many areas of life, which moderated the demands on the acoustic environment. For example, the flexibilization of the workplace led to increased requirements for environmental conditions to allow for effective work. At the same time, outdoor spaces took on new meanings and uses as people increasingly sought opportunities for social interactions and recreation. The change of the acoustic environments and the adaptation of individual and social needs proceeded equally; and it appears likely that certain pandemic related developments endure to a certain extent in post-pandemic soundscapes. Therefore, the question appears justified to what extent requirements for noise protection in urban context would have to take account any of the new conditions of post-pandemic times. Data regarding the change of acoustic environments based on long-term measurements over the last years as well as observations on recent noise complaints behavior are presented and discussed from the perspective of current noise legislative and urban health.

8:45

3aNS3. New methods for predicting soundscape. Klaus Genuit (Res., HEAD Acoust. GmbH, Ebertstraße 30a, HEAD Acoust. GmbH, Herzogenrath 52134, Germany, klaus.genuit@head-acoustics.com)

The ISO 12913 series Soundscape standard provides procedures to describe the acoustic environment in context and cognition and to interpret it with the help of psychoacoustics. This presupposes the presence of an acoustic environment and to interview people living in this soundscape. As during the pandemic interactive communication was limited, a greater focus was led on creating artificial acoustic environments. With the help of a simulation method, a human-focused acoustic virtual environment for the auralization of traffic noise is implemented based on existing traffic data. Traffic noise is an important attribute for the design of attractive living spaces. A reliable prediction of this burden in urban development and traffic planning offers a great potential. For this purpose, source, transmission, and receiver properties as well as movement profiles from traffic data and possible noise reduction measures are considered. Methods and concepts for simulation as well as human perception are focused. The potential to exploit available data from traffic and planning is demonstrated using example scenarios. The simulation enables the auralization and evaluation of the perception of traffic noise in different scenarios.

9:05

3aNS4. The dystopian soundscape: A post COVID soundscape exploration. Keely M. Siebein (Siebein Acoust., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com) and Gary W. Siebein (Siebein Assoc., Inc., Gainesville, FL)

The pandemic has caused a huge upheaval in the global landscape. From travel restrictions, illness testing and vaccination requirements, to unrest and tensions among so many, this marks an unprecedented time. As certain states and countries enforced draconian restrictions, people have sought refuge leaving some areas and repopulating others. From a soundscape of a small city whose recent booming population since covid has found increasing tension in protests and religious demonstrations, to a large city, dealing with new growth and noise concerns from entertainment establishments, short term rentals and loud vehicles. New technology is emerging to help police some of these concerns, but is there an ethical conscience in the enforcement of these? How does our role in soundscape and acoustic design help shape the future? What does the handling of this pandemic foreshadow about similar circumstances that are to come, and what does this mean for the soundscape of the future?

9:25

3aNS5. Soundscape experiences of vulnerable groups in the pandemic and post-pandemic era: Acoustic environments are diverse but not inclusive. Yoshimi Hasegawa (Dept. of Acoust. Design, Kyushu Univ., 4-9-1 Shiobaru, Minami ku, Fukuoka 815-8540, Japan, y.hasegawa.a88@m.kyushu-u.ac.jp)

People perceive sound environments in diverse ways. Experiencing substantial changes in soundscapes due to disruptions of the coronavirus disease (COVID-19) pandemic was relatively adaptable for some people but painfully unacceptable for other people, especially physically, mentally, or socially disadvantaged persons (i.e., vulnerable groups). This talk presents soundscape experiences of those vulnerable groups during the COVID-19 pandemic and highlights their adverse and heterogeneous perceptions to the pandemic-related changes in acoustic environments. Reviewing a collected COVID-19 pandemic-related studies shows that people from socially vulnerable groups found their surrounding soundscapes detrimental and bothersome. Highly heterogeneous experiences of the acoustic environments were observed among people with atypical hearing characteristics such as hearing impairments and tinnitus. While extreme acoustic environments (i.e., being in silence or exposed to excessive noise) were found as unfavorable for all listeners, there are too many non-auditory confounding factors influencing the soundscape experiences in their real-life situations. The findings would advance knowledge on their diverse auditory experiences and support resilient post-pandemic soundscapes for broader individuals, which will potentially contribute to designing inclusive sound environments for human well-being.

9:45

3aNS6. Soundscape of war. Arezoo Talebzadeh (Ghent Univ., 126 TechLane Ghent Sci. Park, Ghent 9052, Belgium, arezoo.talebzadeh@ugent.be)

Soundscape refers to the perception of the sonic environment; the recent pandemic has changed our perception of our living environment and reminded us of the need to study unpredicted and unpleasant soundscapes. For many, the pandemic brought quietness and a lack of stimuli; however, the soundscape was chaotic and fearful for healthcare workers and frontline workers. The experience also reminded us that there are neglected soundscapes to evaluate. One acoustic environment rarely studied or evaluated is the soundscape of war. There is insufficient attention to the auditory sensation of horror and terror caused by conflicts. When talking about the soundscape of war not referring to sonophobia, as one may not fear loud noises, yet being in the conflict has its specific auditory effect. How to use soundscape methods to evaluate the war soundscape? When soundwalk and interview are not practical, what will be the best way to evaluate the soundscape? Where are terror and horror in eight scale evaluation of pleasant, chaotic, vibrant, uneventful, calm, annoying, eventful and monotonous? There are studies on the effect of sound on veterans with PTSD, but not on people living in war zones. Should research use Virtual Reality to evaluate the soundscape of war? The goal is to find ways to assess the short-term and long-term effects of war soundscape on people and to find ways to improve their well-being.

10:05

3aNS7. Soundscape perception in a shifting context—Lessons from COVID-19. Pam Jordan (Heritage, Memory, and Material Culture, Univ. of Amsterdam, Postbus 94203, Amsterdam, Nord-Holland 1090GE, Netherlands, p.f.jordan@uva.nl) and Andre Fiebig (Eng. Acoust., TU Berlin, Berlin, Germany)

The ISO-12913 standards acknowledge the primacy of context in perceiving acoustic environments. In soundscape assessments, context is constituted by both physical surroundings and psychological, social, and cultural factors. Many studies have found that people assess soundscape perception similarly in comparable physical surroundings (such as urban parks) despite differing individual associative contexts. However, studies at the Berlin Wall Memorial historic site have shown that providing historic contextual information to study participants can shift their soundscape perception away from the generalized baseline. The COVID-19 lockdown measures enacted in 2020 in Germany dramatically altered user activity in the memorial landscape, introducing a new environmental and behavioral context. Building on previous investigations at the memorial, this paper investigates what effect the restrictions had on the soundscape context and its perception by visitors. Informal interviews paired with comparative measurements indicated that expectations and perceptions shifted for local stakeholders in this new context. Perceived silence by visitors did not match measurements, and tourist absence affected perception for local users. This holds repercussions for soundscape and heritage site designs serving multiple populations with divergent expectations and perceptions. The impacts of soundscape assessments being neither static nor generalizable across stakeholders are discussed with suggestions for further research.

Session 3aPA

Physical Acoustics and Structural Acoustics and Vibration: Acoustics and Elasticity of Consolidated and Unconsolidated Granular Materials II (Hybrid Session)

John Yoritomo, Cochair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, D.C. 20375

Richard Weaver, Cochair
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James Bittner, Cochair
Michigan Technological University, P.O. Box 22, Houghton, MI 49931

Chair's Introduction—8:25

Invited Papers

8:30

3aPA1. Using high-speed ultrasound to visualize dynamic jamming of dense suspensions. Heinrich Jaeger (Phys., Univ. of Chicago, 929 E 57th St., Univ of Chicago, Chicago, IL 60637, h-jaeger@uchicago.edu)

A remarkable property of concentrated, or dense, suspensions comprising nanometer- to micrometer-size particles in a liquid is their ability to transform from a liquid-like to a solid-like state under applied stress. This jamming phenomenon can be triggered by impact at the surface of a dense suspension and generates a shear front that propagates into the bulk of the material. Tracking and visualizing such front is difficult because suspensions are typically opaque optically, and the front can propagate at several meters per second. This talk will discuss high-speed ultrasound imaging (10 000 frames per second) to extract the flow field associated with the front propagation and visualize the associated shear jamming process. Unlike densification in dry granular media, jamming by shear does not involve significant changes in the local particle fraction while driving the transition to solid-like behavior. On the basis of these findings, we discuss ideas that link the transient stress response of dense suspensions to current theoretical models, which almost exclusively focus on steady-state shearing conditions.

9:00

3aPA2. Ultrasound and acoustic wave propagation measurements in rocks and granular media made concurrently with *In Situ* synchrotron x-ray imaging. Ryan Hurley (Mech. Eng., Johns Hopkins Univ., 3400 N Charles St., Malone 140, Baltimore, MD 21218, rhurley6@jhu.edu) and Chongpu Zhai (Xi'an Jiaotong Univ., Xi'an, China)

Ultrasound transport through granular media has been of significant interest to physicists, civil engineers, and geophysicists studying soils and rocks for decades. Of particular interest are the features which govern wave speeds, dispersion, and attenuation in these materials. We will discuss experiments in which ultrasound waves were propagated through disordered 3D granular media while samples were imaged with x-ray computed tomography (XRCT) and 3D x-ray diffraction (3DXRD) – together, these measurements provide the particle contact topology and inter-particle forces. We show through data analysis and modeling that both contact topology and contact forces control the fastest measured wave speeds and dispersion in these materials, while contact topology alone explains most of the observed wave attenuation. We will also discuss ongoing studies of acoustic emissions (AE) generated in granular materials and sandstones during triaxial compression. These studies, also performed with *in-situ* XRCT and 3DXRD, aim to directly visualize changes in microstructure (fracture, flow) and fluctuations in stress associated with AE events.

9:30

3aPA3. Understanding contact acoustic nonlinearity through coupled synchrotron x-ray imaging and dynamic acoustoelastic measurements under stress. Chun-Yu Ke, Prabhav Borate, Clay Wood (Penn State Univ., University Park, PA), Mark Rivers (Univ. of Chicago / Argonne National Lab., Chicago, IL), Jacques Riviere (Penn State Univ., University Park, PA), and Parisa Shokouhi (Penn State Univ., 212 EES Bldg., University Park, PA 16802, pxs990@psu.edu)

Dynamic acoustoelastic testing (DAET) is a vibro-acoustic technique, which has been used to measure acoustic nonlinearity in disparate solid media. DAET is a pump-probe approach, where the strain pump-induced changes in the elastic properties of the medium are monitored by a pair of ultrasonic probes at frequencies much higher than that of the strain pump. In a fractured/cracked medium, the measured changes in the elastic properties are attributed to fracture breathing (opening/closing) or shearing caused by the low-frequency

pumping. In a set of unprecedent experiments, we use time-lapse synchrotron x-ray imaging in tandem with quasi-static loading/unloading and DAET to visualize the fast dynamics mechanisms in four different materials systems under various stress levels. These include two samples with localized CAN: fractured Westerly granite and Berea sandstone and two samples of granular media with distributed CAN: glass beads and sand. The synchrotron x-ray images are analyzed to obtain true contact areas and changing strain fields under stress. The ultrasonic signals are analyzed in relation to the changes in contact areas during both the quasi-static and dynamic loading to examine the state-of-the-art understanding of ultrasonic transmission in fractured media as well as acoustoelasticity and dynamic acoustoelasticity.

10:00–10:15 Break

10:15

3aPA4. Earthquake fault slip and nonlinear dynamics. Paul A. Johnson (Geophys., Los Alamos National Lab., MS D446, Los Alamos, NM 87545, paj@lanl.gov) and Chris W. Johnson (Geophys., Los Alamos National Lab., Los Alamos, NM)

Earthquake fault slip under shear forcing can be envisioned as a nonlinear dynamical process dominated by a single slip plane. In contrast, nonlinear behavior in Earth materials (e.g., rock) is driven by a strain-induced ensemble activation and slip of a large number of distributed features—cracks and grain boundary slip across many scales in the volume. The bulk recovery of a fault post-failure and that of a rock sample post dynamic or static forcing ("aging" or the "slow dynamics") is very similar with approximate log(time) dependence for much of the recovery. In our work, we analyze large amounts of continuous acoustic emission (AE) data from a laboratory "earthquake machine," applying machine learning, with the task of determining what information regarding fault slip the AE signal may carry. Applying the continuous AE as input to machine learning models and using measured fault friction, displacement, etc., as model labels, we find that the AE are imprinted with information regarding the fault friction and displacement. We are currently developing approaches to probe stick-slip on Earth faults, those that are responsible for damaging earthquakes. A related goal is to quantitatively relate nonlinear elastic theory (e.g., PM space, Arrhenius) to frictional theory (e.g., rate-state).

10:45

3aPA5. Laboratory characterization of induced seismicity. Sai Kalyan Evani (General Electric (GE)-Global Res. Ctr., Niskayuna, NY) and John Popovics (Civil and Environ. Eng., Univ. of Illinois, 205 N. Mathews St. MC-250, Urbana, IL 61801, johnpop@illinois.edu)

Subsurface fluid injection causes an increase in seismic activity near injection sites. To ensure public safety and improve public acceptance of subsurface fluid injection, it is important to understand the underlying mechanisms giving rise to these behaviors and characterize the factors affecting induced seismicity. The work reported here aims to better understand induced seismicity in faulted/fractured subsurface rock formations. A series of experiments are conducted in core flooding and triaxial configurations on test samples with a preexisting fracture/fault to study frictional slipping at small and large scales respectively. The geometries of both surfaces on either side of the fault are characterized using x-ray CT. The extent of fault gouging in a specimen after slipping is quantified by defining a cumulative gouging parameter using x-ray CT measurements before and after the experiment. Acoustic emission (AE) events emanating from the specimens in both configurations are monitored. The results demonstrate that (1) the orientation of fault relative to the major principal stress direction affects the slip characteristics, (2) frictional slipping at a fault can occur at different length scales, (3) with an increase in the scale of slipping, the spectral ratio of low-frequency bins increases while high-frequency bins reduces, and (4) locations on the fault surface that exhibit a sudden change in surface normal are most susceptible to gouging/damage during frictional slipping.

11:15

3aPA6. Predicting timescales of relaxation in the shallow Earth's subsurface. Luc Illien (GFZ Potsdam, Telegrafenberg Bldg. F, Rm. 423, Postdam 14473, Germany, lillien@gfz-potsdam.de) and Christoph Sens-Schönfelder (GFZ Potsdam, Potsdam, Germany)

Earthquakes introduce long-lasting transient mechanical damage in the subsurface, which cause postseismic hazards such as enhanced landsliding and can take years to recover to steady-state values. This observation has been linked to relaxation, a phenomenon observed in a wide class of materials after straining perturbations. In this presentation, I analyze the successive effect of two large earthquakes on ground properties through the monitoring of seismic velocity from ambient noise interferometry in the Atacama desert in Chile. The absence of rainfall in this area allows study of the mechanical state of the subsurface by limiting the potential effect of variable groundwater content. I show that relaxation timescales are a function of the current state of the subsurface when perturbed by earthquakes, rather than ground shaking intensity. Our study highlights the predictability of earthquake damage dynamics in the Earth's near-surface and potentially other materials. I propose to reconcile this paradigm with existing physical frameworks by considering the superposition of different populations of damaged contacts.

Contributed Paper

11:45

3aPA7. One-dimensional experimental verification of weak-form homogenization theory for heterogenous systems. Gordon M. Ochi (U.S. Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil), Kyle G. Dunn, Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH), and Michelle E. Swearingen (Construction Eng. Res. Lab., US Army ERDC, Champaign, IL)

Metamaterials are synthetic composite materials that exhibit specific effective properties for long-wavelength stimuli. Modeling of these materials often relies upon homogenization methods to replace inhomogeneous systems with more simple to analyze homogenous systems. In this work, we discuss a one-dimensional experimental verification of a weak-form homogenization method based on Hamilton's principle for mechanical media. In the experiment, a heterogeneous structure is fabricated out of wood and placed within a four inch diameter section of PVC pipe. This heterogeneous structure is designed such that it restricts the cross-sectional area at regular intervals, yielding alternating sections of restriction/no-restriction along the pipe length. The source is a compression driver outputting weakly non-linear sinusoidal waveforms, located at one end of the PVC pipe, while the other end of the pipe is terminated with anechoic foam to negate end reflections. Data are gathered through implementation of the two-microphone method at multiple locations along the pipe length, which enables isolation of the forward traveling wave. Verification of the homogenization theory in the one-dimensional case is discussed, as well as the ability for the heterogeneous structure to reduce the shock formation distance.

WEDNESDAY MORNING, 10 MAY 2023 — CHICAGO AB, 8:00 A.M. TO 11:15 A.M.

Session 3aPP

Psychological and Physiological Acoustics and Speech Communication: Sensory and Non-Sensory Influences on Auditory Development

Laurianne Cabrera, Cochair
Integrative Neuroscience and Cognition Center, CNRS, 45 rue des Saints Peres, Paris, 75006, France

Bonnie K. Lau, Cochair
University of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195

Chair's Introduction—8:00

Invited Papers

8:05

3aPP1. Short periods of perinatal acoustic experience alter the developmental trajectory of auditory cortex. Stephen Lomber (Physiol., McGill Univ., 3655 Promenade Sir William Osler, McIntyre Blg., Rm 1223, Montreal, QC H3G1Y6, Canada, steve.lomber@mcgill.ca) and M. A. Meredith (Anatomy and Neurobiology, Virginia Commonwealth Univ., Richmond, VA)

Compared to hearing subjects, psychophysical studies have revealed specific superior visual abilities following hearing loss early in development. The neural substrate for these superior abilities resides in auditory cortex that have been reorganized through crossmodal plasticity. Furthermore, the cartography of auditory cortex is altered following the loss of auditory input early in life. This study examined how perinatal exposure to brief periods of acoustic stimulation alters the developmental trajectory of auditory cortex. Movement detection, localization in the visual periphery, and face discrimination learning are superior in congenitally deaf cats. To examine the role of acoustic experience in mediating these enhanced functions, hearing animals were chemically deafened at increasing ages postnatal. Animals had 1–16 weeks of acoustic exposure prior to deafness onset. Overall, >9 weeks of acoustic experience resulted in no enhanced visual abilities. With >4 weeks of acoustic exposure, enhanced motion detection was not evident. As acoustic experience increased during development, the overall size of auditory cortex, and the size of individual auditory areas also expanded. These results demonstrate that increasingly longer periods of perinatal acoustic experience result in reduced enhanced visual abilities and an increased size of auditory cortex.

8:20

3aPP2. Maturation of cortical responses to sound in the first year of life. Bonnie K. Lau (Univ. of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195, blau@uw.edu), Samu Taulu, Patricia K. Kuhl, and Adrian K. C. Lee (Univ. of Washington, Seattle, WA)

Infants show sophisticated sound discrimination from the time they are born but the neural mechanisms that support infant auditory perception are not well understood. A central question is the discrepancy between the early onset of auditory skills and the protracted and extended maturation of the auditory cortex. To investigate the cortical processing of sound, we have been employing magnetoencephalography (MEG) with recent advancements in movement compensation to obtain functional neural measures of sound processing in awake infants. MEG responses from an Elekta Neuromag 306-channel system were recorded longitudinally at 3 (n = 27), 6 (n = 19), and 11 (n = 14) months to an ecologically salient bi-syllabic word and an amplitude-modulated complex tone pair in typically hearing infants. The neural generators of the MEG signals were determined using an equivalent current dipole (ECD) model. Our preliminary analyses have focused on two measures: cortical responses to acoustic change and the development of hemispheric lateralization to speech. Our preliminary analyses show that high quality MEG data with good signal-to-noise ratios can be obtained by 3 months and dipole modeling of MEG signals in combination with advanced movement compensation offers a temporally precise method of investigating the maturation of auditory cortical networks in infants.

8:35

3aPP3. Exploring the impact hearing differences have on the development of speech perception during infancy. Kristin M. Uhler (Dept. of Physical Medicine and Rehabilitation, Univ. of Colorado Anschutz Medical Campus, 12631 East 17th Ave., Academic Office 1, Ste. 1201, Aurora, CO 80045, kristin.uhler@cuanschutz.edu), Kerry A. Walker, Daniel Tollin (Dept. of Physiol. & Biophys., Univ. of Colorado Anschutz Medical Campus, Aurora, CO), and Phillip Gilley (Inst. of Cognit. Neurosci., Univ. of Colorado, Boulder, CO)

Early periods of perceptual skill development are driven by the dynamic interplay between language experiences and maturation of auditory sensory pathways. Among infants with normal hearing (INH), the first year of life is a seminal period for refining speech perception abilities shaped by exposure to language. Research suggests that refinement of speech discrimination abilities depends on an infant's exposure to speech sounds, thus infants who are hard-of-hearing (IHH) are susceptible to atypical development during this period. Currently, the impact of inconsistent auditory cue access on development of speech perception is unknown among IHH. To investigate this, our lab has employed both electroencephalography (EEG) and a conditioned head turn (CHT) paradigm to examine auditory cue access and speech perception abilities, respectively, over the first year of life. We will provide a broad overview of our recent and continuing work among IHH and INH using EEG and CHT findings which suggest: (1) early EEG measures of cue access predict later behavioral speech perception abilities, (2) hearing age (duration of time between hearing aid fitting and testing) is related to speech perception abilities measured by CHT, and (3) a significant relationship between infant speech perception and both spoken expressive and receptive language abilities.

8:50

3aPP4. Cortical tracking of speech in quiet and noise in infants and children. Talat Jabeen (Univ. of Washington, Seattle, WA 98195, tjabeen@uw.edu), Sharon Wong, and Bonnie K. Lau (Univ. of Washington, Seattle, WA)

Past research has shown that in multi-talker listening situations, cortical tracking of the speech envelope is observed for the target speech stream in adults. While children are known to perform worse than adults under noisy conditions, many aspects of how the cortical processing of speech develops is unknown. Here, we recorded EEG responses to continuous speech in infants and 7-to-18-year-olds in Quiet, Co-located Noise, and Segregated Noise. The target speech stream consisted of infant directed speech or an audiobook for children. The noise consisted of four-talker babble and was constructed from four audiobooks read by 2 males and 2 females. Thirty-two channels of EEG data were recorded with the stimuli presented at an overall level of 65 dB SPL for children and 70 dB SPL for infants with a +5 dB target-to-noise SNR via speakers at 0°, +90°, and −90° azimuth. EEG signals were analyzed using the decoding model of the Multivariate Temporal Response Function toolbox to assess how well the stimulus envelope could be reconstructed from the recorded neural responses. Preliminary results show evidence of cortical tracking of the speech envelope for both infants and children in quiet and that cortical tracking degrades in both Co-located and Segregated Noise.

9:05

3aPP5. Non-auditory processing of cochlear implant stimulation after unilateral auditory deprivation in children. Karen A. Gordon (The Hospital for Sick Children, Rm 6D08, 555 University Ave., Toronto, ON M5g 1x8, Canada, karen.gordon@utoronto.ca), Carly Anderson (The Hospital for Sick Children, London, United Kingdom), Salima Jiwani (The Hospital for Sick Children, Toronto, ON, Canada), Melissa J. Polonenko (The Hospital for Sick Children, Minneapolis, MN), Dan Wong, Sharon Cushing, and Blake Papsin (The Hospital for Sick Children, Toronto, ON, Canada)

We have aimed to identify effects of unilateral hearing on the developing brain in a series of studies in children who have one deaf ear and access sound in the other ear through normal hearing, a hearing aid, or a cochlear implant (CI). Multi-channel EEG in these cohorts revealed strengthening of auditory pathways from the hearing ear to the auditory cortices. Response differences found in frontal and precuneus cortical areas in adolescents with long term unilateral CI use suggested involvement of non-auditory cortical areas during passive hearing. We also examined cortical effects of leaving one ear deprived of sound during development in children with unilateral hearing (listed above) who received a CI in their deaf ear. Atypical responses to CI stimulation showed strong activation of the ipsilateral auditory cortex which increased with duration of unilateral deprivation. Graph theory analysis including community detection revealed a network that was organised into a temporal-occipital module and a temporal-frontal module, with left Heschel's gyrus and left lingual gyrus central to information flow between the two. Results across these studies identify neuroplasticity in children with unilateral hearing/deprivation and suggest that regions outside of the auditory cortex can modulate sound processing through CIs.

9:20–9:35 Break

9:35

3aPP6. Reduced processing efficiency impacts auditory detection of amplitude modulation in children: Evidence from an experimental and modelling study. Irene Lorenzini (Integrative Neurosci. and Cognition Ctr., CNRS, Paris, France), Léo Varnet, Christian Lorenzi (Ecole Normal Superieure, CNRS, Univ Paris Sci. et Lettres, Paris, France), and Laurianne Cabrera (Integrative Neurosci. and Cognition Ctr., CNRS, 45 rue des saints peres, Paris 75006, France, laurianne.cabrera@parisdescartes.fr)

Detection of amplitude modulations (AM) improves until 10 years of age. This development may not be explained only by sensory maturation but also by improvements in processing efficiency: the ability to make *efficient* use of available sensory information. This hypothesis was tested on 86 6-to-9-year-olds and 15 adults using AM-detection tasks assessing sensitivity, masking and response consistency. Sensitivity was estimated by the detection thresholds of a sinusoidal AM applied to a pure-tone carrier; AM masking was estimated as threshold elevation produced when replacing the pure-tone carrier by a narrowband noise; response consistency was estimated using a double-pass paradigm where the same set of stimuli was presented twice. Results showed that AM sensitivity improved from childhood to adulthood. AM masking did not change with age, indicating that the selectivity of AM filters was adult-like by 6 years. However, response consistency increased developmentally, suggesting reduced processing efficiency in childhood. At the group level, double-pass data were well simulated by a model of the auditory system assuming higher internal noise for children than adults. At the individual level, a sub-optimal decision strategy was added to better capture inter-individual variability. Thus, both systematic and stochastic inefficiencies may explain worse AM detection in childhood.

9:50

3aPP7. Development of speech perception in noise: Neural correlates of stream segregation in children and adolescents. Axelle Calcus (Université libre de Bruxelles, 50, av. F. Roosevelt, Brussels 1050, Belgium, acalcus@ulb.ac.be) and Elena Benocci (Université libre de Bruxelles, Brussels, Belgium)

In noisy backgrounds, listeners perform the auditory scene analysis: they parse the different auditory streams ("stream segregation"), and selectively focus on one stream as it unfolds over time ("selective attention"). Auditory scene analysis is thought to mature slowly. Here, we sought to investigate developmental changes in the neural signature of auditory stream segregation. Children ($n = 17$, aged 8 to 17) and young adults ($n = 10$) were presented with sequences of sounds consisting in stochastic variations of figures and backgrounds. These figure-ground sequences have been shown to elicit distinct "signature" EEG responses, including the object-related negativity (ORN) and P400, which reflect the processing concurrent auditory objects. Participants were also presented with a consonant identification task, where the consonant was presented in quiet, in the presence of one interfering talker, and in the presence of speech-shaped-noise. Preliminary results indicate a developmental effect on figure-ground segregation. Interestingly, we observe a significant correlation ($r = 0.45$) between stream segregation and speech perception in noise. However, we do not observe a developmental effect on the ORN/P400. Results will be discussed in light with the literature on the topic of auditory scene analysis and the development of the central auditory pathways.

Contributed Papers

10:05

3aPP8. Contribution of listening effort and cognitive processing to psychometric functions during adolescence and young adulthood. Julia J. Huyck (Speech Pathol. and Audiol., Kent State Univ., 1325 Theatre Dr., CPA A144, Kent, OH 44242, jhuyck@kent.edu)

Hearing and listening are critical to how adolescents communicate, learn new information, and engage with technology and culture. However, performance on auditory perceptual tasks takes a long time to mature. Age-related improvements in executive functions and other cognitive functions likely contribute to this long developmental trajectory. To examine how listening effort and cognition relate to psychometric functions on auditory perceptual tasks in adolescents and young adults, we used the method of constant stimuli to test 10- to 23-year-olds on frequency discrimination, temporal interval discrimination, and gap detection. During task performance, an eye tracker was used to measure pupil size and blink rate as proxies for listening effort or engagement. Typically, larger pupil size and fewer blinks are associated with higher engagement. All listeners also completed a battery of cognitive tests including tests of verbal and nonverbal reasoning, working memory, processing speed, and attention. We will demonstrate any changes in the estimated thresholds and psychometric function slopes during adolescence and will relate these developmental changes to individual differences in listening effort and cognition. [Work funded by NIDCD.]

10:20

3aPP9. Auditory cortex plasticity supports social learning. Dan Sanes (Ctr. for Neural Sci., New York Univ., 4 Washington Pl., Fl. 6 Rm. 621, New York, NY 10003, dhs1@nyu.edu), Nihaad Paraouty (Ctr. for Neural Sci., New York Univ., New York, NY), Justin Yao (Dept. of Otolaryngol., Rutgers Univ., New York, NY), Léo Varnet (Laboratoire des Systèmes Perceptifs, École Normale Supérieure, Paris, France), Chi-Ning Chou, and Sue Yeon Chung (Ctr. for Computational Neurosci., Flatiron Inst., Simons Foundation, New York, NY)

The acquisition of new auditory skills can be facilitated by experiencing a conspecific performing a well-defined behavior (i.e., social learning). Although the neural bases for auditory social learning remain uncertain, one plausible hypothesis is that social experience induces long-term changes to auditory cortex, thereby facilitating the subsequent acquisition of an auditory skill. To explore this idea, we developed a social learning paradigm in which naïve observer gerbils are exposed to a demonstrator that performs an amplitude modulation (AM) rate discrimination task on the other side of an opaque divider. Thus, observers have no access to visual cues. When exposed to a demonstrator for five successive days, observers subsequently acquire the AM task more rapidly than controls. Two experiments suggest that auditory cortex is necessary and sufficient for social learning. First, inactivating the observer's auditory cortex during each social exposure session significantly delayed task acquisition. Second, recordings from the observer's auditory cortex revealed that individual neurons displayed a significant improvement in AM stimulus discrimination across the five exposure sessions, and the magnitude of neural improvement correlated with an animal's subsequent rate of task acquisition. Together, these findings suggest that auditory cortex plasticity plays a pivotal role in social learning.

Invited Paper

10:35

3aPP10. Effects of acoustic-phonetic access on audiovisual speech perception in children. Kaylah Lalonde (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68104, kaylah.lalonde@boystown.org), Adam K. Bosen, Grace A. Dwyer, and Abby Pitts (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Visual speech helps compensate for degraded acoustic input, especially when redundancy between the accessible acoustic and visual cues is low. Recent findings indicate that children who are hard of hearing benefit more from visual speech than their peers with normal hearing, yet the two groups demonstrate similar speechreading ability. It is unclear whether this increased benefit reflects differences in auditory and multisensory development or differences in redundancy between the visual and acoustic cues available to each group. This study examines the extent to which acoustic-phonetic access (acoustic frequency content) influences auditory and audiovisual word recognition in children with normal hearing. Isolated word and word-in-sentence recognition accuracy are measured for auditory-only and audiovisual speech that were high-pass filtered and low-pass filtered with a 2 kHz cutoff frequency. Data collection is ongoing. We hypothesize that children will demonstrate greater audiovisual benefit when acoustic speech is low-pass filtered than when it is high-pass filtered, because low-frequency acoustic content is less redundant with visual speech cues than high-frequency acoustic content. These findings will elucidate whether acoustic-phonetic access can explain differences in audiovisual benefit between children with normal hearing and children who are hard of hearing. [Work supported by NIH R21DC020544.]

10:50–11:15
Panel Discussion

3a WED. AM

Session 3aSA

Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics: Historical Perspectives in Structural Acoustics (Hybrid Session)

Alexey Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, Bethesda, MD 20712

Kuangcheng Wu, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., West Bethesda, MD 20817

A. J. Lawrence, Cochair
Walker Department of Mechanical Engineering, University of Texas at Austin, 204 E. Dean Keeton Street, Stop C2200, Austin, TX 78712-1591

Chair's Introduction—8:30

Invited Papers

8:35

3aSA1. Fourier acoustics: Uncovering the origins of sound. Earl G. Williams (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. Code 7106, Washington, DC 20375, earl.williams@nrl.navy.mil)

No doubt, Jean-Baptiste Joseph Fourier has had a great impact in science. At the root of Fourier Acoustics, the Fourier series, when employed as a starting point and manipulated, allows us to uncover the origins of sound in a simple and elegant way, not only in acoustics but also in electromagnetics. A well-known stepping stone on the path is nearfield acoustical holography (NAH) and its theory can be recast almost completely, and simply, in terms of two-dimensional (2D) Fourier series and applied to any geometry. For separable coordinate systems, each of the components of the series can be propagated to a surface conformal to the measurement surface using an analytically defined propagator. But even when the surfaces are not part of a separable system, the spirit of Fourier's expansions is mirrored with the singular value decomposition, applied to a numerically derived propagator constructed using the equivalent source method. A comprehensive demonstration of Fourier acoustics applied to aeroacoustics was in the cabin of a Boeing aircraft, in flight, using a two-dimensional spherical array. Reconstructions of vector intensity in a volume from correlations of the measured pressure to panel accelerometers showed the direction and location of the dominant fuselage sources of noise—uncovering the origin of sound. [Work supported by the Office of Naval Research.]

9:00

3aSA2. Trends in the 40+-year development of advanced structural acoustic and vibration computational modeling. Robert M. Koch (US Navy, 304 White Horn Dr., Kingston, RI 02881, Robert.M.Koch@navy.mil)

Over the course of the past 40-45 years of computational modeling in the fields of structures, structural acoustics, and vibrations, there have been extreme advancements made in numerous areas that have greatly increased and improved the size, accuracy, efficiency, and complexity of numerical solutions possible. Most of this progress has been made through a combination of improvements in the distinct areas of computer processing approaches/speed and available memory, primarily evolutionary enhancements in detailed numerical material and continuum mechanics models, development of entirely new physics-based numerical vibroacoustic modeling approaches, etc. This paper presents a historical review of the detailed progress made in computational structural acoustics and vibration from the "early days" in the 1980s through current state-of-the-art advanced numerical approaches, much of which would not have been dreamed possible back four decades ago! While most of the examples utilized in this talk are pulled from the speaker's specific technical area of undersea vehicle/ system structural acoustics, the overall trends presented are certainly represented across the broad spectrum of vibroacoustics modeling applications.

9:25

3aSA3. My journey in structural acoustics. David Feit (4601 N Park Ave., Apt. 213, Chevy Chase, MD 20815, dvdfeit@yahoo.com)

Structural acoustics refers to problems of the vibrational response of structures in the presence of an ambient compressible fluid. In this presentation, I present my introduction to the subject by virtue of my interaction with Miguel Junger, one of the pioneering investigators in this field. At his suggestion and with his insight, I addressed two canonical problems that launched my journey into the subject. The first is that of a spherical object spring mounted, moving axially, and radiating into a compressible fluid medium. The other is that

of a force excited elastic plate radiating into a fluid half space. Because of their simplicity both problems can be solved exactly, and their solution provides physical insight into the solution of more practical configurations, which were pursued further and included collaboration with Carderock colleagues Gideon Maidanik and Murray Strasberg.

9:50–10:05 Break

10:05

3aSA4. Fifty five years following a long and winding road through structural acoustics and vibrations. Jerry H. Ginsberg (5661 Woodsong Dr., Dunwoody, GA 30338, j.h.ginsberg@comcast.net)

Being an academic at institutions that did not have a dedicated mission in acoustics allowed me to pursue diverse research topics. Probably because of the challenge presented by structural acoustics and vibrations, my efforts were primarily focused in those areas. They proceeded from nonlinear shell theory, to pipeline instability, to nonlinear vibrations, to nonlinear acoustics, to variational methods for acoustic radiation, to shock response of submarines, to experimental modal analysis, to coupled acoustic-structure resonance of a flying telescope with an open cavity. I will attempt to attempt to explain what led me to this progression, and to place these efforts in the context of communal efforts.

10:30

3aSA5. The 1980s—The glory years for the development of active noise and vibration control. Scott D. Sommerfeldt (Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

In the field of acoustics in general, and structural acoustics in particular, a new "hot topic" seems to burst upon the scene once or twice each decade. Everyone wants to get involved, numerous sessions on the topic appear at our meetings, and progress on the topic seems to leap forward—at least until the next hot topic comes along. In the 1980s, active noise and vibration control burst upon the scene, becoming the then current hot topic, as advances in digital signal processing made the technique much more attractive. The number of publications and presentations on the topic exploded, and active control was the bandwagon that everyone wanted to be on. The author was just beginning his career during this time period and was able to begin pursuing research on this topic in the early stages of the explosion. The intense interest in the topic continued until about the mid-1990s. This paper will review some of that early activity, what advances were made, why the interest seemed to decline in the 1990s, and what the current state of the discipline is as the research has continued to slowly move forward over the intervening years.

10:55–11:20
Panel Discussion

Session 3aSC

Speech Communication and Education in Acoustics: Infusing Social Justice in Speech and Hearing Acoustics Pedagogy: Principles and Case Studies (Hybrid Session)

Benjamin Munson, Chair
University of Minnesota, 115 Shevlin Hall, Minneapolis, MN 55455

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. Centering social justice in speech and hearing acoustics pedagogy. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu) and Marisha Speights (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Science can be seen as a collaborative activity designed to help humans be better adapted to the world that we find ourselves in and that we and our ancestors created. It is imperative that all scientists, including acousticians, create knowledge that helps all human beings, which acknowledges salient characteristics of the worlds we find ourselves in, and which highlights the human-created features of those worlds. This talk opens the special session by talking about three issues. We argue that there is woefully poor representation of the speech, language, and hearing (SLH) of people of color and of other marginalized groups. Studies of the normative behaviors of human beings are based on a woefully small and unrepresentative subset of languages and speech communities. We must also grapple with the consequences of our fields' historical focus on the normativity of different speech and hearing behaviors. This involves interrogating what broader social structures benefit from a focus on normativity and pathologization. We argue that our field should follow the lead of Plaut (2010, Psych. Inquiry) in understanding not just differences in acoustic communication across individuals and groups, but differences in the perception of differences in SLH ability.

8:20

3aSC2. Infusing social justice in the development and instruction of a course on the history of phonetics. Marc Garellek (UC San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu), Shai Nielson, Tamara L. Rhodes, and Emily Clem (UC San Diego, La Jolla, CA)

We describe the design and implementation of a mixed undergraduate and graduate seminar on the history of phonetics (from the mid-19th to mid-20th centuries). The goal of the course was to teach and learn about the history of the field using a pedagogical framework in which all aspects of the course are framed with the values of antiracism, social justice, diversity, equity, and inclusion. Planning for the course involved discussion between the instructor, the graduate student DEI committee, the department Curriculum Committee, the linguistics librarian, and the teaching and learning resource center. Seminar topics included the development of systems of phonetic annotation and speech technology; the emergence of modern phonetics, speech-language pathology, and linguistics; and advancements in speech acoustics. In this presentation, we will also discuss students' participation and evaluation of the course, which were very enthusiastic. Overall, we show how a course on the history of phonetics can be modeled and taught in such a way as to show the ways in which our field has marginalized and discriminated against certain communities, as well as to consider the ways in which we may learn from our history so as to be a more just and inclusive discipline moving forward.

8:35

3aSC3. Addressing diversity in speech science curricula. Paul E. Reed (Dept of Communicative Disord., Univ. of Alabama, Box 870242, Tuscaloosa, AL 35487, pereed1@ua.edu) and Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR)

A fundamental understanding of speech science is a critically important component in fields such as linguistics, communication disorders and sciences, cognitive science, and speech technology. Therefore, courses in speech communication are integral to the undergraduate and graduate curricula of these and other subjects. Despite being at the forefront of pedagogical innovations, speech science courses have lagged in representing cultural and linguistic diversity in the classroom. Many speech scientists understand that linguistic diversity is fundamental to human language systems. However, discussions of language variation tend to be relegated to a single section within a course. The lack of inclusion of diverse language varieties results in a lack of engagement with many social variables such as race, ethnicity, and gender identity, among others. We argue that this "status quo" in speech science courses must change, such that linguistic diversity must be addressed in and throughout all courses, even those where diversity is not the focus. We will present concerns with current approaches to teaching about cultural and linguistic diversity. Furthermore, we will explain the benefits of including diversity instruction throughout the curriculum. Finally, we present specific recommendations for instructors to incorporate teaching of linguistic diversity throughout their curriculum and speech science courses.

8:50

3aSC4. Vague terminology and pedagogy. Rachel Elizabeth Weissler (Linguist, Univ. of Oregon, 1451 Onyx St., Eugene, OR 97403, rweissle@uoregon.edu)

A key challenge for any academic field is using, and teaching about, field-specific terminology. Speech science faces a unique set of circumstances, since many students and non-experts believe they understand a specific term. However, their definitions may vary drastically from the agreed upon definitions within the field. This becomes even more complicated when a specific piece of terminology is contentious within the field. Take, for example, the term "non-native." Recent work has demonstrated that this term is simultaneously imprecise (i.e., does not refer to all and only individuals who meet specific criteria) and othering (i.e., inherently places one group in opposition to the "normal" group of native speakers). However, this terminological conundrum provides a ripe ground for pedagogical shift. In this presentation, we address both the pedagogical opportunities available in addressing these terminological issues, including metalinguistic discussions of the notions of categories and how this may tie into discussions around linguistic relativity. Furthermore, this approach allows for a broader discussion in the classroom that changing terminology only solves part of the problem; regardless of the terminology we use, we do not mitigate the bigger issues of interest. In fact, altering terminology alone can result in virtue-signaling (miles-hercules & Muwwakkil 2021).

9:05

3aSC5. Active cultural responsivity to terminology in speech and hearing science. Elizabeth Ancel (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr SE115 Shevlin Hall, Minneapolis, MN 55455-0279, ancel014@umn.edu)

People are often the main focus of speech and hearing science. That being so, it is essential that scholars in the field recognize that all people are active participants in a larger sociocultural context. This context has an effect on how we refer to, describe, and study people from different backgrounds. Because of this, research and pedagogy in speech and hearing science have a responsibility to recognize and integrate this context into their practices. This presentation will focus specifically on the usage of terminology that describes different social and cultural groups. I will first review examples of the terminological history of a few commonly encountered groups. This section will also serve as a practical resource for the current best practices for terminology usage in accordance with the groups' preferences. Then, I will provide general recommendations for being culturally responsive to terminology usage in research and in the classroom, with examples from common topics taught in speech and hearing science classes.

9:20–9:35 Break

9:35

3aSC6. Incorporating a gender expansive perspective into speech science pedagogy. Brandon Merritt (Rehabilitation Sci., The Univ. of Texas at El Paso, 1101 N. Campbell, El Paso, TX 79902, bmmerritt@utep.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

This talk challenges the assumption of a traditional gender binary in speech science pedagogy. We first define and operationalize notions of sex and gender as they relate to speech communication. We will broadly consider acoustic properties that relate to anatomical differences across humans (such as vocal tract length and vocal fold mass) and acoustic properties that relate to socio-phonetic variation. We challenge the notion that anatomical differences are the only source of speech variability across gender categories. We discuss problems in the literature related to how speakers and listeners are categorized (typically either as binary male/female or binary man/woman with the assumption of cisgender gender identity). We move beyond the binary assumption of the ways that "males" and "females" speak by demonstrating speakers' agency in constructing linguistic styles that are rich in social information and that convey nuanced identities. We highlight the variability in listeners' perception of speaker gender and consider potential sources of this variability. Attendees will be provided with instructional tools that incorporate gender diversity into the speech science curriculum and that can be used in classroom discussions and activities.

9:50

3aSC7. Kids talk too: Linguistic justice and child African American English. Yolanda F. Holt (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu)

African American English (AAE) is a language form used primarily, though not exclusively by Black Americans of historical African descent. The language is rule governed, robust, and resistant to assimilation to the white American English (WAE) dialects that surround it. Although public schools in the United States have been putatively integrated since 1954, the 2020 census data reveals most children continue to live in segregated communities. White children live in communities that are on average 69% white and Black children in communities that are on average 55% Black. As children acquire the language forms of their community peers, we should expect most Black children will enter school using some AAE. Speech and language scientists can participate in linguistic justice by teaching the systematic nature of AAE speech most likely to perturb listeners and lead to misidentification of AAE speech variation as disorder. We will focus on weak syllable deletion and final consonant variation. We will show how to use AAE audio recordings and simple spectral analysis to decrease misinterpretation of typical AAE child speech as speech errors.

10:05

3aSC8. Comparing tools for automation of formant estimation in diverse pediatric populations. Hannah Valentine (Communicative Sci. and Disord., New York Univ., 665 Broadway, Ste. 610, New York, NY 10012, hannahvalentine17@gmail.com), Joel MacAuslan (Speech Technologies and Appl. Res. Corp, Lexington, MA), Maria Grigos (Communicative Sci. and Disord., New York Univ., New York, NY), and Marisha Speights (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Pediatric vowel productions are notoriously hard to reliably quantify. Listener judgments and transcription are subjective, show low intra- and interrater reliability, and are influenced by listener bias. Formant estimation may offer more objectivity, but child speech is fraught with acoustic challenges. Children have less refined articulatory control and they make more production errors. They have high fundamental frequencies, wide formant bandwidths, more variable formant values, and increased subglottal coupling relative to adult speech. Historically, estimation of pediatric formants has been done manually, which is laborious and time-consuming. In recent years, automation tools have been developed to speed up the process. However, these tools have not been widely tested on pediatric speech samples. Even more critically, these tools have not been tested on diverse children. This study uses speech samples from children of color, children with disabilities, and children with both of these identities to compare three automation tools: SpeechMark® (Boyce *et al*., 2012), Fast Track™ (Barreda, 2021), and a custom Praat® (Boersma & Weenink, 2021) script written by the first author. Outcomes of each tool will be compared and contrasted. The discussion will review the considerations, benefits, and tradeoffs of each automation tool when working with diverse pediatric speech samples.

10:20

3aSC9. Interventions to improve undergraduate communication with international instructors. Valerie Freeman (Oklahoma State Univ., 042 Murray Hall, Dept. of Commun. Sci & Disord., Stillwater, OK 74078, valerie.freeman@okstate.edu), Sara Loss (Oklahoma State Univ., Stillwater, OK), and Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR)

U.S. undergraduates (USUGs) have negative biases toward non-native speaking (NNS) instructors like international teaching assistants (ITAs), which can lower student comprehension and ITAs' course evaluations (Rubin & Kang 2009). Our long-term collaboration seeks to improve attitudes and intelligibility so USUGs may assume more of the communicative burden with NNSs. We are piloting multiple activities ("interventions") in various university courses, including accent familiarization, non-English transcription, oral English proficiency scoring, and live or simulated USUG-NNS contact. Outcomes are measured qualitatively (e.g., written reflections) and quantitatively through USUG pre- and post-test intelligibility tasks and "first impression" teaching and personality ratings after listening to audio clips of unfamiliar ITAs. We compare the most and least promising interventions for improving intelligibility and attitudes as measured by these auditory tasks (three semesters, $N > 750$ students, 6 intervention types + controls). Analysis is ongoing, but in the pilot cohort ($N = 117$), longer, more involved interventions trended toward higher intelligibility scores and improved overt attitudes, although covert attitudes remained stable. Time permitting, we can also discuss practicalities of implementing such anti-racism interventions into existing courses (instructor buy-in, ITA cooperation, student compliance, etc.).

10:35

3aSC10. Care-full and reproducible research: Teaching research skills and ethics to undergraduate researchers using critical replication studies. Alayo Tripp (Speech Lang. Hearing Sci., Univ. of Minnesota, Twin Cisties, Minneapolis, MN) and Rachel Hayes-Harb (Linguist, Univ. of Utah, 255 S Central Campus Dr., LNCO Bldg., Ste. 2300, Salt Lake City, UT 84112, r.hayes-harb@utah.edu)

Course-based replication studies facilitate broad student participation in science, allowing early-career researchers a direct and fast route to impactful research questions and high-quality methods and analyses. They also provide students opportunities for exciting hands-on experience engaging with contemporary research questions as both theorists and experimentalists. However, reproducibility requires thoughtful attention to generalizability, and new scholars must be taught to critically consider study motivations, assumptions and goals. Teaching reproducibility in this way promotes public trust in science as well as productive skepticism. We discuss a reproducibility-focused undergraduate course where the goal is to provide a meaningful, authentic, and responsible research experience for students. While a heavy emphasis on replication can lead to an uncritical reproduction of harmful science, the present course emphasizes a more inclusive approach to knowledge production, where research skills development is inseparable from education on responsible conduct of research, social justice, and open science values and practices. We cover topics including: selecting a study for replication, intentional development of students' collaboration skills, finding a balance between replication and extension, students' reflexivity practice and positionality, human participants considerations, the responsible dissemination of students' research, course learning outcomes and assessment, and research mentoring "at scale."

10:50–11:15
Panel Discussion

Session 3aSP

Signal Processing in Acoustics, Education in Acoustics, and Physical Acoustics: My Favorite Signal Processing Homework Problems

John R. Buck, Cochair
Electrical and Computer Engineering, UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Kathleen E. Wage, Cochair
George Mason University, 4400 University Drive, Fairfax, VA 22151

Invited Papers

8:00

3aSP1. Chirps, harmonics, and aliasing. Thaddeus B. Welch (ECE, Boise State Univ., 4213 West Quail Ridge Dr., Boise, ID 83703, tbwelch@gmail.com) and Cameron H. Wright (ECE, UWyo, Laramie, WY)

An invited paper for a special session entitled "My Favorite Signal Processing Homework Problems." Abstract: The use of recorded real-world audio signals tends to motivate almost all of the students in our signal processing classes. Early in any course where analog-to-digital conversion is presented, the topic of aliasing is always discussed. One of our favorite, three-part homework problems is to: 1. import an instructor provided recording of an audio frequency whistle (a chirp) into MATLAB, 2. plot its spectrogram, and 3. identify aliasing on the spectrogram. Given the rich harmonic structure of the provided signal, aliasing is clearly present on the properly implemented spectrogram.

8:20

3aSP2. Building an understanding of sampling and filtering. Jill K. Nelson (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, jnelson@gmu.edu)

While signals and systems homework problems often focus on giving students practice in procedures and computations, some of the problems I most enjoy are those that ask students to step away from familiar procedural approaches and think outside the box. In this talk, I will describe three problems that encourage conceptual thinking about sampling and filtering. In the first, which focuses purely on sampling, students are asked to determine how various operations on a generic signal change its Nyquist sampling rate. The second problem asks students to consider how aliasing may be allowable (and even desirable) when sampling will be followed by discrete-time filtering. The final problem asks students to characterize a sampler followed by a discrete-time system based on observation of the input and the output, which requires students to tease apart the effects resulting from sampling and those resulting from filtering. These three problems are drawn from signals and systems material I have encountered over many years of teaching and through a variety of sources. I appreciate the way they encourage students to think of sampling and filtering as tools that can be used creatively rather than as a prescribed series of steps.

8:40

3aSP3. The further adventures of Lynn Eyar, Ty Minvariant, and Connie Volution. John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu), Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

My favorite homework problems challenge students to grow beyond direct procedural mathematics, engaging their reasoning with higher level tasks from Bloom's Taxonomy: Analyze, Evaluate, & Create. This whirlwind tour presents four flavors of signal processing homework problems. First, graph-based conceptual problems require students to reason from figures, rather than deriving equations. These problems scaffold students as they transfer intuition from closed form analytic equations to the often messy time series and power spectra obtained with real world data. Second, Matlab problems require students to work with audio data on problems that are often deliberately underspecified. These problems introduce students to the cyclic trial and error nature of engineering design. Third, module problems require students to design systems to meet a specification, while also considering some form of cost. Real engineering is almost always a multivariate optimization problem balancing performance with constraints. Finally, story problems require students to identify common misconceptions in the mouths of fictional peers. Team dynamics reward engineers who can succinctly identify the shortcomings or misconceptions in poor approaches as well advocate on behalf of their own designs. This talk may include some audience participation.

9:00

3aSP4. Using acoustic signals for demonstrating key signal processing and communication systems concepts. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

In this talk, I will highlight the importance of reinforcing mathematical concepts in signal processing and communication systems theory with Matlab-based data analysis. In particular, I will provide examples and demos of homework and project problems I assigned in my linear systems and communication systems courses, and discuss how it helped the students learn in an agile hybrid virtual setting during the pandemic, and whether repeating the same (or similar) problems in an in-person or hybrid setting made a difference. I will also discuss the importance of playing with real-world acoustic signals, and lessons learned, including changes I made in the problem assignments based on student feedback, student performance, and classroom discussions.

9:20

3aSP5. Project-based learning engages students at a level more directly applicable to their future careers. Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, 110 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801, acsinger@illinois.edu)

Some of my favorite homework problems are not homework problems at all, but, rather, more open-ended project-based assignments that elevate students' thinking toward the top of Bloom's Taxonomy. While some students may have the background to jump into a project-based assignment, most benefit from an increasing level of scaffolding. This not only brings all students to a common level from which they can develop a more coherent project, but it also provides multiple opportunities for them to solidify their understanding and build on curated, worked examples. This talk will explore two different types of courses, one theory-based and one experiential, each benefiting from a scaffolded framing for a more open-ended signal processing or data analysis project.

Contributed Paper

9:40

3aSP6. R2-D2's broken beamformer: A galactically consequential homework exercise. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu) and Blaine M. Harker (Dept. of Phys. and Astronomy, Brigham Young Univ., Asheville, NC)

Identify the Imperial spy, assist the Rebellion, and save a galaxy far, far away! In "Acoustical Measurement Methods," a Brigham Young University physics course typically taken by seniors and new graduate students, the semester's culminating topic is beamforming. Students write their own delay-and-sum (DAS) beamformers for a uniform line array and test them experimentally by finding the direction of a loudspeaker. The topic of their technical memorandum, however, is a homework-style exercise in which they use their validated DAS codes to help R2-D2 locate the Imperial spy in the Mos Eisley cantina. With the spy's inquiry about the Rebel base masked by a cacophony of sounds coming from different directions, and with its real-time processor on the fritz, the dependable droid needs help processing its phased-array signals to separate the sources and locate the spy. Though this exercise doesn't follow the Star Wars® canon in the least and certain dimensional constraints are temporarily cast aside in the DAS problem creation, students do not seem to care. They engage deeply in the problem and generally produce their best-written technical memorandum of the semester in pursuit of intergalactic (though still academic) valor and fame. "Help me, [acoustics student], you're my only hope!"

9:55–10:10 Break

Invited Paper

10:10

3aSP7. Interactive demonstrations in acoustic signal processing using Jupyter notebooks. Randall Ali (Elec. Eng., KU Leuven, Kasteelpark Arenberg 10, Leuven 3001, Belgium, randall.ali@esat.kuleuven.be) and Toon van Waterschoot (Elec. Eng., KU Leuven, Leuven, Belgium)

In studying acoustic signal processing, students are faced with seemingly abstract concepts such as continuous-time and discrete-time representations, time-frequency transformations, convolution, vector and inner product spaces, orthogonality, and optimal and adaptive filtering. While mathematical development of the theory and illustration through simple examples during teaching can help shed light on some of these concepts, they are not always sufficient for students to achieve a full level of understanding. As a means to improve this situation, we have consequently developed several interactive demonstrations in acoustic signal processing, covering a range of topics, from the fundamentals of sampling and quantisation to the more advanced optimal and adaptive filtering. These demonstrations have been made with Jupyter notebooks and offer several advantages. Apart from being open source, students immediately get to work with realistic data as they can generate and record acoustic signals using their computer's loudspeaker and microphone. Using the live coding aspect of the notebooks, they can also quickly process, visualise, and listen to the results from the processing of these acoustic signals, as well as repeat this workflow for various parameter sweeps. We will present some of these demonstrations and show how they can serve as an effective teaching aid.

Contributed Paper

10:30

3aSP8. Design and implementation of a least-mean-square adaptive notch filter. Randall Ali (Elec. Eng., KU Leuven, Kasteelpark Arenberg 10, Leuven 3001, Belgium, randall.ali@esat.kuleuven.be), Hannes Rosseel, and Toon van Waterschoot (Elec. Eng., KU Leuven, Leuven, Belgium)

The task of removing sinusoidal components from observed signals can be accomplished by using a notch filter with a specific attenuation at a particular frequency. In some applications, however, such as acoustic feedback control, the frequency at which attenuation is required is unknown and possibly time-varying, and hence an adaptive notch filter is a more appropriate solution. Transitioning from a fixed notch filter to an adaptive one is by no means trivial and involves the understanding of a range of digital signal processing (DSP) topics from pole-zero placement techniques for designing infinite impulse response filters to optimal and adaptive filtering algorithms. In the signal processing algorithms and implementation graduate course taught at KU Leuven (Belgium), we study the design of an adaptive notch filter, which is based on a constrained biquadratic IIR representation, and whose parameters are updated using a least-mean-square algorithm. Students also have to implement the algorithm on a 16-bit DSP TMS320C5515. In this presentation, we will discuss the design and implementation challenges of this adaptive notch filter and how it serves as an illustrative example/homework problem where several aspects of DSP are interwoven.

Invited Papers

10:45

3aSP9. Statistics and probabilities for spectral analysis. Christ D. Richmond (Dept. of Elec. and Comput. Eng., Duke Univ., Rm. 327, Gross Hall, Box 90984, Durham, NC 27708, christ.richmond@duke.edu)

Spectral analysis of data plays a central role in many signal processing applications, and comparing the relative output levels between filters often provides clues about the actual signals present in the data. Since data are often corrupted by noise, interference, or some other source of randomness, these filter outputs are also subject to random variations. Proper interpretation of these filter outputs is made possible by an awareness of the nature of their statistical variability. Thus, homework problems that illustrate how basic fundamental properties of random variables/vectors lend significant insight into the nature of such filter output comparisons are of great value in engineering practice. This talk will describe a handful of interesting problems that help with the interpretation of filtered data outputs used in some form of spectral analysis (e.g., FFT filters, beamformers, etc.). The problems considered will leverage notions of invariances, regeneration properties, and a basic algebra of random variables.

11:05

3aSP10. Model and data-driven homework problems for learning signal processing concepts. Martin Siderius (Portland State Univ., 1600 SW 4th Ave., Ste. 260, Portland, OR 97201, siderius@pdx.edu)

The Electrical and Computer Engineering Department at Portland State University offers a course in Sensor Array Processing. The content is designed to be introductory and approachable for undergraduate, first year graduate and students from other disciplines such as Mechanical Engineering or Physics. The signal processing techniques explored use two or more sensors and can be applied to acoustics or electromagnetic waves. I have two favorite homework problems: one is based on simulated data and the other based on measured data and both help solidify important concepts. For the simulations, a model is developed based on the method of images and can be applied to signals from a sonar in the ocean or to voices in a room. These multipath environments are used to illustrate how a signal (such as an audio clip the student creates) would change in different environments or source/receiver geometry. The channel impulse response is modeled and together with convolution allows different waveform responses to be easily computed. My favorite problem using measured data is based on recordings of a broadband source on two sensors. Cross-correlation is used to determine the source direction. This problem explores the concepts of pulse compression and pre-whitening for time-delay and direction estimation.

11:25

3aSP11. Application of wavelet wransform wechniques to measured data. Kendal Leftwich (Phys., Univ. of New Orleans, 1021 Sci. Bldg., Univ. of New Orleans, New Orleans, LA 70148, kmleftwi@uno.edu) and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

Wavelet transform theory includes a powerful and abstract set of signal processing tools. Students often understand the underlying mathematics of how to perform the procedures but lack the conceptual and computational understanding of how and when to apply wavelets to measured data. In this talk, we will discuss a project designed to take what is learned from textbooks and classroom lectures and apply these skills and tools to measured data from the field. The project is to cover an entire semester with multiple points for instructor evaluation and student revision. The end goal is that students will learn more signal processing skills through computational methods and data analysis and will produce a paper that can be part of a thesis/dissertation as well as a presentation at a future meeting.

Contributed Paper

11:45

3aSP12. Fireworks. Thomas B. Gabrielson (Penn State Univ., State College, PA) and Daniel C. Brown (Penn State Univ., PO Box 30, State College, PA 16804, dcb19@psu.edu)

Once a year, in many parts of the country, transient acoustic sources light up the sky. A fourth-of-July fireworks display provides an excellent opportunity for exploring array-based detection and direction-of-arrival determination. Multiple-boom events are difficult to process; however, less-frequent single-boom events are often sufficiently isolated in time for unambiguous interpretation. Cross-correlations between channels of a four-microphone array provide time delays for angle-of-arrival determination and measures for event detection. The distinct time-domain wave shapes and high signal-to-noise enable straightforward checks of the cross-correlation delays. In addition, a chirp sequence broadcast from a known location provides a ground-truth measurement and opportunities to implement matched filtering and envelope processing. Detection strategies can include received power, rate of change of power, normalized cross-correlation coefficient, or Fisher F-statistic. A least-squares fit by plane-wave approximation gives an estimate of the actual slowness vector, which, in turn, permits estimation of error and of local sound speed. Conversion of the slowness vector to arrival azimuth and elevation angles provides a simple exercise in 3D interpretation. The homework exercise can be as simple as determination of inter-sensor time delays or as complicated as demonstrating a scheme for automatic detection and interpretation.

WEDNESDAY MORNING, 10 MAY 2023 MICHIGAN/MICHIGAN STATE, 8:00 A.M. TO 11:35 A.M.

Session 3aUW

Underwater Acoustics, Acoustical Oceanography, Noise, Physical Acoustics, Architectural Acoustics, and Computational Acoustics: 3D Acoustic Propagation

Tyler J. Flynn, Cochair
Acoustics & Electromagnetics, Johns Hopkins Applied Physics Lab, 1231 Beal Ave., Ann Arbor, MI 48109

Jennifer Cooper, Cochair
Johns Hopkins University, Applied Physics Laboratory, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723

Invited Papers

8:00

3aUW1. Development of a three-dimensional semi-coherent energy flux propagation model. Mark A. Langhirt (Penn State Univ., 201 Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, mal83@psu.edu), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Sheri Martinelli, and Daniel C. Brown (Penn State Univ., State College, PA)

The classic energy flux propagation model integrates an averaged intensity distribution over elevation angle for an azimuthally symmetric waveguide, and the method has also been extended to reincorporate some of the coherence in the mode-sum cross product, producing caustic features and convergence zones. Some ocean environments with asymmetric range-dependent bathymetry and acoustic medium properties can refract the acoustic wave horizontally which N-by-2D models cannot capture due to their assumed symmetry. Fully three-dimensional models can capture horizontal refraction but are typically computationally expensive to execute. Further development and study of the energy flux method is worthwhile since it avoids root-finding, is not a marching solution, and it invokes mode number invariance under the adiabatic modes assumption to map propagation angles and directly integrate the intensity distribution at the receiver location. A first-pass at analytically deriving a generalizable three-dimensional energy flux model was presented previously, and this current study focuses on further analytical development, numerical implementation, and efforts to verify the model using other three-dimensional acoustic propagation models.

8:20

3aUW2. Identifying the presence and importance of 3D acoustic effects in New England Shelf Break Acoustics experiment data.
Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu) and Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

During the 2021 New England Shelf Break Acoustics (NESBA) experiments, a real-time physical oceanographic and ocean acoustic modeling effort sought to predict planned acoustic transmissions between a network of moorings *in situ*. While positive comparisons between simulated acoustics and hydrophone observation were made during field experiments, updates to ocean environment models post-cruise have improved the model and data agreement. Of importance to this real-time modeling effort is to use the disagreement between acoustic simulation and acoustic data to estimate model environment errors. By identifying spatial locations of likely error in the modeled sound speed, *in situ* acoustic simulations can identify acoustically significant locations to perform measurements of ocean water state properties such as temperature and salinity, as well as marine geological features. This presentation will focus on comparison efforts, extracting model environment error estimates, and explore the influence of 3D effects on this process. [This research is supported by the Office of Naval Research.]

8:40

3aUW3. Out-of-plane arrivals recorded by surface drifters during the Northern Ocean Rapid Surface Evolution experiment.
Megan Ballard (Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu), Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Pierre-Marie Poulain (Sci. and Technol. Organisation, Ctr. for Maritime Res. and Experimentation (STO CMRE), LaSpezia, Italy), Jennifer MacKinnon (Scripps Inst. of Oceanogr., UCSD, San Diego, CA), Andrew J. Lucas (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), and Alejandra Sanchez-Rios (Scripps Inst. of Oceanogr., UCSD, San Diego, CA)

The Northern Ocean Rapid Surface Evolution (NORSE) focuses on characterizing the key physical processes that govern the predictability of upper-ocean rapid evolution events. The principal experimental site is Jan Mayen Channel, which connects the Greenland and Norwegian Seas. During the fall 2022 process cruise, signals from a moored source transmitting a 135-second-long LFM upsweep in the 500–1500 Hz band every four hours were recorded by three SVP Drifters equipped with hydrophone arrays. Over a three-day period, the drifters moved north across Jan Mayen Channel toward the moored source. The individual recordings are subject to variable levels of ambient sound caused by changing wind conditions and platform noise. In recordings with positive SNR, an in-plane arrival is observed. In a subset of these recordings, a second arrival is observed having travel time consistent with a propagation path from the moored source, reflecting off the ridge on the south side of the channel, and arriving at the drifters. A third arrival is also observed having travel time consistent with reflection from face of the bathymetric rise on the east end of the channel which forms Jan Mayen Island. This talk will present the measurements and explain the data though forward modeling. [Work supported by ONR.]

9:00

3aUW4. 3D finite element modeling techniques and application to underwater target scattering. Aaron M. Gunderson (Appl. Res. Laboratories, The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, aarong@arlut.utexas.edu)

Underwater acoustic target scattering measurements rely on high fidelity modeling for experimental comparison and understanding. Three-dimensional (3D) finite element models are well suited for this purpose, as they can account for arbitrary or unknown target properties and configurations/orientations within complex and asymmetrical seafloor environments. High acoustic frequencies and large physical distances associated with *in situ* scattering measurements pose challenges to 3D modeling efforts in terms of model sizes and runtimes. Certain model considerations must be made to keep the 3D model computationally efficient, yet accurate in predictive capability. Numerically determined Green's functions are demonstrated to permit 3D model reduction, while still preserving far-field scattering prediction capability through the Helmholtz-Kirchhoff integral. By determining Green's functions within the model, they need not be known or estimated for complex ocean environments a priori. Nontraditional scattering formulations and a survey of boundary truncation methods also are explored and implemented for maximal accuracy within small 3D computational domains. Model results for canonical elastic targets within varying seafloor environments are shown and compared to theory and experimentation. [This work has been supported by the Strategic Environmental Research and Development Program, and by the Office of Naval Research, Ocean Acoustics.]

9:20

3aUW5. 3D Gaussian ray bundling to improve the speed and accuracy of transmission loss eigenrays and multi-static reverberation, in littoral environments. Sean M. Reilly (Physical Sci. and Systems (PS2), Raytheon BBN Technologies, 127 John Clark Rd., Middletown, RI 02842, sean.m.reilly@raytheon.com)

The wavefront queue 3D (WaveQ3D) model is a research effort to create fast and accurate acoustic transmission loss eigenrays and multi-static reverberation, in littoral environments, for real-time sonar simulation/stimulation systems that require simultaneous updates from hundreds of acoustic contacts. WaveQ3D is a Gaussian ray bundling model that improves computation speed of rapidly evolving scenarios by computing propagation results on a spherical earth. We will show that the added complexity of the spherical coordinate system has almost no impact on execution speed, and how it avoids the laborious task of transforming the 3D oceanographic data into Nx2D radials. WaveQ3D improves the accuracy of transmission loss and reverberation calculations in littoral environments by including 3D effects, such as horizontal refraction from the ocean bottom. This talk will focus on the design decisions that made during model development and their impact on the utility of WaveQ3D for a few different applications.

9:40–9:55 Break

9:55

3aUW6. Inclusion of three dimensional acoustic propagation in nuclear event characterization. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Emanuel F. Coelho (Sci., Appl. Ocean Sci., LLC, Springfield, VA), Mario Zampoli, Tiago Oliveira, and Georgios Haralabus (CTBTO, Vienna, Austria)

The verification regime of the Comprehensive Nuclear-Test-Ban Treaty is the International Monitoring System that includes deep-water hydroacoustic stations for monitoring the world oceans for nuclear tests. As underwater sound propagation satisfies the acoustic wave equation, it is subject to three-dimensional (3D) effects (refraction, diffraction, reflections) when in the presence of horizontal gradients due to bathymetry, oceanography, or the presence of continents. The current processing system does not exploit these 3D effects, because it relies on in-plane propagation for all acoustic paths, which may lead to azimuth errors for event association. In this work, a set of acoustic propagation codes are being proposed to integrate 3D propagation into the automated event localization algorithm as well as into the analyst work flow. A summary of the acoustic models, demonstrations of observed 3D phenomenon for large underwater events, as well as a program plan to update the processing scheme will be presented.

10:15

3aUW7. Regional and global scale propagation simulations of infrasonic signals. Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., PO Box 1663, M/S F665, Los Alamos, NM 87545, pblom@lanl.gov)

Infrasonic signals are frequently observed at large propagation distances of 100s to 1000s of kilometers from the source and, in extreme cases, signals have been observed to circle the globe. Simulating acoustic propagation at these large scales requires a framework that not only accounts for 3D propagation effects including along- and cross-path winds, but also for the non-Cartesian geometry of the atmospheric layer surrounding the globe. Furthermore, interaction of propagating infrasonic energy with terrain features is known to have a notable impact on signals, particularly when the wavelength of the signals is comparable to the terrain scale. Finally, spatial variations in the atmospheric structure beyond simple stratification can have significant impacts on propagation paths. Numerical tools for acoustic ray tracing that account for these and other complications will be detailed and demonstrated using several recent infrasonic events of interest.

Contributed Papers

10:35

3aUW8. Introducing bellhopcxx/bellhopcuda: Modern, parallel BELLHOP(3D). Louis A. Pisha (Marine Physical Lab, Scripps Oceanogr., UC San Diego, 9500 Gilman Dr., San Diego, CA 92093, lpisha@eng.ucsd.edu), Joseph Snider, Khalil Jackson, and Jules S. Jaffe (Marine Physical Lab, Scripps Oceanogr., UC San Diego, San Diego, CA)

BELLHOP and BELLHOP3D are popular ray/beam based underwater acoustics simulation programs, written in Fortran by Dr. Michael B. Porter starting in 1983. We introduce bellhopcxx/bellhopcuda, a translation of these programs into a modern, multithreaded C++/CUDA codebase. This open-source codebase can be built and run on a multicore CPU or an NVIDIA GPU and can also be built as a library for integration into other software without requiring output data to be written to files. The new version is typically somewhat faster than the Fortran version even in single-threaded mode, and in multithreaded mode usually scales roughly with the number of logical CPU cores minus some overhead. In runs with large numbers of rays, depending on the hardware, the CUDA version can bring several times additional performance gain over the multithreaded CPU version, sometimes reaching over 100x the performance of BELLHOP or BELLHOP3D. The results of the new version closely match those of the original version in most cases, but there are occasionally mismatches, partly due to edge case issues inherent in the BELLHOP physics model. We have improved the handling of these cases and made other improvements to the Fortran version, bringing these changes to the new version.

10:50

3aUW9. Comparison of semi-analytical and numerical methods for the scattering of acoustic waves by rough, sloped seabed geometries. Yiyi Whitchelo (Dept. of Mathematical Sci., Univ. of Liverpool, Peach St., Liverpool L69 7ZL, United Kingdom, ywhitchelo@gmail.com), Stewart Haslinger, Daniel Colquitt (Dept. of Mathematical Sci., Univ. of Liverpool, Liverpool, United Kingdom), and Duncan Williams (DSTL: Defence Sci. and Technol. Lab., Salisbury, United Arab Emirates)

The seabed possesses variations in its material properties, layering structure, slope and roughness making prediction, and analysis of sound propagation in shallow water highly complex. For certain frequency ranges, the effect of roughness becomes significant. Numerous models have been adopted to account for the scattering of acoustic waves by rough surfaces, including Kirchhoff approximation (for relatively large scales of roughness), small perturbation theory (for relatively small scales of roughness), and small slope approximation, which bridges across the two regimes under certain conditions. Although grounded in analytical theory, these approaches invariably involve the computation of numerical integrals so additional approximation theories have been explored including stationary phase methods and Taylor series approximations. In this study, these semi-analytical methods are incorporated within a new sound propagation model and compared with a numerical graphics processing unit (GPU)-accelerated finite element model incorporating roughness parameters (root mean square height and correlation length) for rough, sloped seabed geometries (with gradients ranging from 0° to 45°) in shallow water environments. The effect of large- and small-scale roughness on sound propagation is investigated in two- and three-dimensional space. The finite element modelling is used to validate the optimal choice of semi-analytical method for the different environments considered.

11:05

3aUW10. Modeling shallow water scattering from rippled seabed in the Long Island Sound. Aaron M. Gunderson (Appl. Res. Laboratories, The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, aarong@arlut.utexas.edu) and Autumn Kidwell (Appl. Res. Laboratories, The Univ. of Texas at Austin, Austin, TX)

A high frequency (HF) AUV-mounted forward-looking sonar deployed in the shallow Long Island Sound recorded acoustic scattering measurements from large bedform ripples. The axis of transmission was horizontal and laterally oblique to the ripple plane, but at wide horizontal aperture, so that broadside scattering from the ripples was believed to be a significant contributor to the scattering return. The effects of the ripples are explored through finite element model evaluation and comparison. In this study, the seafloor ripples are orders of magnitude larger than the acoustic wavelengths used in the survey. In plane and out of plane, scattering effects are considered, as well as the effects of ripple asymmetry. Modeling techniques for dealing with high frequency systems over large physical distances are discussed. [This work was supported by the Office of Naval Research, Littoral Geosciences and Optics.]

11:20

3aUW11. Ray tracing of long-range underwater acoustic vortex wave propagation. Mark Kelly (Mech. Eng., Georgia Inst. of Technol., 313 Oakland St., Decatur, GA 30030, mkelly75@gatech.edu) and Chengzhi Shi (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The underwater acoustic communications environment is severely band-limited, which leads to a bottleneck in data transfer. Existing methods of data transfer in underwater acoustic communications applications typically rely primarily on conventional temporal and frequency modulation techniques and achieve bit rates peaking at approximately 40 kb/s. One method of easing the bottleneck and increasing the data rate is to explore further potential degrees of freedom which may be utilized. Acoustic orbital angular momentum (OAM) is a physical quantity that characterizes the rotation in a propagating helical pressure wavefront. The unique phase patterns of OAM carrying vortex waves form an orthogonal basis which may be useful as an additional degree of freedom in acoustics communications applications; however, the long-distance propagation of these waves is largely unstudied. By employing BELLHOP's ray tracing algorithm, the dominant features of a propagating OAM carrying vortex wave are tracked over long ranges (to and beyond 1 km) under various environmental conditions. This provides essential guidance in the design of the sending and receiving arrays of high-speed underwater communications systems, which rely on multiplexing acoustic OAMs.

Exhibit

An instrument and equipment exhibition will be located in Chicago D/E on the 5th floor.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Schedule

Monday, 8 May, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including a complimentary beverage.

Tuesday, 9 May, 9:00 a.m. to 5:00 p.m.: Exhibit Open Hours including a.m. coffee break and p.m. break with coffee and soft drinks.

Wednesday, 10 May, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. coffee break.

Session 3pAA

Architectural Acoustics: Hot Topics: Measuring Acoustic Properties and Beyond

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

Brandon Cudequest, Cochair
Threshold Acoustics, 141 W Jackson Blvd. Suite 2080, Chicago, IL 60604

Contributed Papers

1:00

3pAA1. Informing the acoustical privacy of mothers' rooms with measurement data. Blake R. Krapfl (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE 68182-0816, bkrapfl2@huskers.unl.edu), Jennifer M. Epstein (IP Design Group, Omaha, NE), Samuel H. Underwood, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

Mothers' rooms (also referred to as lactation rooms) have become more prominent in workplaces in the United States in recent decades. Since the amendment of Fair Labor Standards Act in 2010, employers in the United States with more than 50 employees have been required to provide a non-restroom space for mothers to nurse or pump breast milk. Such spaces are often designed to offer privacy and comfort—however, little consideration has been given to the acoustical criteria that would be appropriate for these spaces. To inform the development of best practice criteria for sound isolation, a sample of different breast pump models from different manufacturers has been measured to estimate the range of sound power levels and spectral content associated with lactation pump noise. Comparisons are then made between pump noise levels and common workplace wall constructions to test the efficacy of published sound isolation criteria. Additional consideration is given to the unique usage of this space type in comparison to common adjacency types to propose a data-driven sound isolation criteria for mothers' rooms. Results from this effort also aim to inform further work towards developing criteria for reverberation time and background noise levels in mothers' rooms.

1:15

3pAA2. Numerically modeling the sound absorption behavior of 3D printed fibrous absorbers. Amulya Lomte (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, axlomte@shockers.wichita.edu), Will Johnston, and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Recently, we have demonstrated the feasibility of using extrusion-based additive manufacturing methods to fabricate fibrous sound absorbers [1]. Here, we investigate the feasibility of predicting the sound absorption behavior of such structures by extracting their geometry-based transport properties using the hybrid, multiphysics modeling approach. The acoustical transport properties are calculated by modeling the fibrous unit cell and solving three boundary value problems over the representative elementary volume of the periodic fluid domain. Our results show that the absorption predictions obtained using the as-designed unit cell significantly differ from the experimental measurements obtained using conventional two-microphone impedance tube tests. Further investigations conducted using an optical microscope reveal that while the printed fiber diameter remains uniform over its central portion, the fiber diameter decreases drastically near the fiber root. Finally, we show that incorporating these geometrical differences within the model improves computational predictions and accounts for the deviations between the numerical and experimental absorption coefficients. [1] Johnston W, Sharma B. Additive manufacturing of fibrous sound absorbers. Additive Manufacturing. 2021 May 1;41:101984.

1:30

3pAA3. Testing the absorption coefficient of 3D printed perforated panels. Elizabeth G. Zusin (1140 Grant Ave., Pelham Manor, NY 10803, elizusin@icloud.com) and Kimberly A. Riegel (Farmingdale State College, Pelham, NY)

Sound absorption is common in most studios, but most residential spaces are not designed for good acoustics. While these rooms are non-ideal, they are sometimes used to record sounds and need to absorb unwanted frequencies. Perforated 3D printed panels have the potential to make inexpensive and modular home treatments. In order to determine if these panels would be effective, the absorptive properties need to be evaluated. There are several ways to test sound absorption, including the *in situ* subtraction method and the impedance tube transfer function method. Reference materials were used to test the effectiveness of both measurement methods. 3D printed perforated panels were then designed and the absorption was tested. Each surface was tested with the subtraction method, and many different setups for the equipment were tested to find the ideal setup for the most accurate absorption coefficient. The results show varying success. The 3D printed panels were then tested with the transfer function method. An impedance tube was made using PVC pipes and was used to test the transfer function method. This method showed varying results for the absorption coefficient tested, and results will be presented.

1:45

3pAA4. Mechanized acoustic panels for variable reverb control. Arya Nallanthighall (Univ. of Illinois- Urbana-Champaign, 603 E Clark, Champaign, IL 61820, bharath4@illinois.edu), Austin Lu (Univ. of Illinois Urbana-Champaign, Champaign, IL), Kanad Sarkar (Univ. of Illinois at Urbana Champaign, Urbana, IL), Manan Mittal (Elec. and Comput. Eng., Univ. of Illinois, Urbana-Champaign, Urbana, IL), and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Reverberation poses a challenge for speech processing systems and is unavoidable in real environments. As such, acoustic signal processing researchers are interested in robust algorithms that can perform well regardless of reverb severity. Measuring reverberant speech requires access to rooms with the desired dimensions, which may not be readily available. Furthermore, for heavily instrumented recording setups such as our Mechatronic Acoustic Research System (MARS), moving equipment between locations is not practical. We propose a system of actuated multi-textured panels which can significantly alter the reverb properties of a static room. By changing the angle of these panels, we can smoothly transition from high to low reverberation times. We show this empirically and develop a

system that provides a requested reverb level automatically, once configured for a fixed size installation. This tool can allow researchers to use a single room to take measurements applicable to a more diverse range of environments. We integrate our system into the MARS project, an automated tool for generating spatial audio datasets.

2:00

3pAA5. Archaeoacoustic measurement and simulation of the Jesuit Chapel at St. Mary's City, Maryland. Braxton Boren (Performing Arts, American Univ., 4400 Massachusetts Ave., NW, Washington, DC 20016, boren@american.edu)

St. Mary's City, Maryland was the original capital of Maryland, founded as a colony with protections for Catholics by Lord Baltimore. As a result of the colony's religious toleration, the city built a permanent Jesuit chapel, the earliest brick structure in Maryland, in 1667. The chapel was one of the earliest brick worship spaces in America, and the only permanent Catholic worship space in the English-speaking world. When Protestants later took control of the state and moved the capital away, the chapel was closed by the governor, and the Jesuits recycled its materials into other structures. As part of archaeological excavation and heritage preservation efforts at St. Mary's City, the Jesuit Chapel was rebuilt according to historic building practices and materials in the 2000s. The reconstructed chapel was known for its pleasant acoustics by many in the area, and impulse responses were measured in the rebuilt space. The measurements were used to calibrate a geometric acoustic model of the space, which is used as a starting basis for the acoustics of the original 1667 chapel. Some discussion is given to the role of physical versus virtual reconstruction in heritage acoustics and archaeoacoustic projects.

2:15

3pAA6. Case study of a Brothertown Indian Nation cultural heritage site—Toward a framework for acoustics heritage research in simulation, analysis, and auralization. Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, Indianapolis, IN), Seth Wenger (New York, NY, seth.wenger@yale.edu), and Hope Leonard (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, Indianapolis, IN)

The Brothertown Indian Nation and their ancestors have a centuries old heritage of group singing beginning along the Northeast Atlantic coast of what is now the United States and following them along their migration to the Midwest. The 18th century Curricomp cabin still stands in Connecticut and was an important location in the singing history and political movement that built Brothertown (Eeyamquittoowauconnuck). In this paper, a collaboration with Brothertown Indian Nation and an ongoing public humanities project regarding the Tribe's aural traditions, a case study will be presented that investigates (1) the results of acoustic modeling and simulation of the Curricomp cabin and (2) auralizations for both binaural listening and a spatial audio installation using those models. Significantly, this paper employs a novel theoretical framework for acoustic heritage research that allows for (3) an analysis of how the scientific process and technological practices mediate intangible heritage. The acoustic models and auralization techniques created in this case study provide acoustic access to a heritage location that would otherwise be inaccessible to the Brothertown Community. The auralizations used in the spatial audio installation serve as a public humanities tool to amplify the contemporary voice of the Brothertown Indian Nation.

2:30

3pAA7. Bridging the gap: Sound as a tool towards resettlement. Milad Hosseini-Mozari (Multidiscipliary Design, Univ. of Utah, 375 S 1530 E RM 235, RM 235, Salt Lake City, UT 84112, milad@design.utah.edu)

Over the past three years, various sound initiatives have been conducted in a collaboration between the College of Architecture and Planning at the University of Utah, and the International Rescue Committee (IRC). Led by Milad Mozari, an artist and researcher, the medium of sound is used as a tool for exploration, place-making, and spatialization between students and refugees. This paper presentation will go over three categories of how audible sound is used in this collaboration. (1) The use of omnidirectional microphones to create sound collages as a way of introducing a place. This was distinctly differently interpreted in proximity to sound sources by students and IRC clients. (2) Experiments in analog synthesis to create collaborative performances between individuals with different musical backgrounds. (3) The use of ambisonic mixing for virtual reality videos that former IRC clients and students create as a tool for the resettlement of future refugees. Overall, this ongoing initiative is at the intersection of social design and emerging technological tools, which marginalized communities are often unaware of. At its core concept, the project aims to increase accessibility to sound tools to communities that may not come across it in their resettlement process.

2:45

3pAA8. Free-field perceptual evaluation of virtual acoustic rendering algorithms using two head-related impulse response delay treatment strategies. Zane T. Rusk (Dept. of Architectural Eng., The Penn State Univ., 104 Eng. Unit A, University Park, PA 16802, ztr4@psu.edu), Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., University Park, PA), and Matthew Neal (Dept. of Otolaryngol. and Commun. Disord., Univ. of Louisville, Louisville, KY)

Sound scenes can be auralized over headphones using binaural rendering techniques in conjunction with a set of head-related impulse responses (HRIRs). If the directions of the sound objects to be rendered are known, either virtual loudspeaker or Ambisonic scene-based techniques may be used, each of which introduce spatial and timbral artifacts at lower spatial resolutions. Neal and Zahorik quantitatively evaluated the effect of separately applying the HRIR delays to time-aligned HRIRs for use with virtual loudspeaker array techniques, referred to as the prior HRIR delays treatment strategy (*J. Acoust. Soc. Am.* 2022). The present work aims to perceptually validate their quantitative results in a listening study. Free-field point sources were binaurally rendered using five different methods: vector-based amplitude panning, Ambisonics panning, direct spherical harmonic transform of the HRIR set, MagLS, and Principal Component-Base Amplitude Panning (Neal and Zahorik, *Audio Eng. Soc. AVAR Conference* 2022). Renderings were simulated at various directions and spatial resolutions, both with and without the prior HRIR delays treatment strategy and compared to direct HRIR convolution. Broadband noise served as a critical test signal for revealing changes in timbre and localization and a high-resolution binaural head HRIR dataset was used. The subjective data analysis will be presented.

3:00

3pAA9. Developing a virtual reality application for cultural heritage and room acoustics education. Sang Bum Park (School of Architecture and Eng. Technol., Florida A and M Univ., 1938 South Martin Luther King Jr. Blvd., Tallahassee, FL 32307, sang.park@famu.edu)

We tend to use visual factors, such as architectural styles, features, specific decorations, and historical contexts, to characterize cultural heritage. Sound is a transient, ethereal phenomenon that tends to be neglected in historical records. While photographs and drawings can preserve the visual aspect of a building or scene, documenting the sonic impact of the spaces is more complicated. Particularly, the historic places used for sonic activities, such as music halls, performance halls, and worship spaces, are essential to document and preserve the acoustic qualities. An immersive experience using virtual reality (VR) technology effectively promotes public awareness about cultural heritage's importance. It simulates the room acoustics using spatial audio technology. It also can make the VR environment interactive to manipulate architectural features that change the room acoustics such as room volume, finish materials, reverberation time, and sound barrier. The main goal of this paper is to develop a VR application that can be used as a template to create a VR environment where 3rd to 8th-grade students navigate and learn about the history and architectural features of cultural heritage and the basics of room acoustics using a Quest headset.

Session 3pAB

Animal Bioacoustics: General Topics in Animal Bioacoustics

Micheal Dent, Chair
University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 142260

Contributed Papers

1:00

3pAB1. Nonlinear dynamics of spontaneous otoacoustic emissions in the presence of internal noise in models of the inner ear. George Samaras (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr NW, Atlanta, GA 30313, gsamaras3@gatech.edu), Dani E. Agramonte, and Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Spontaneous otoacoustic emissions (SOAEs) are sounds generated by the inner ear in the absence of an external stimulus and are a direct consequence of the inner ear nonlinearity that arises from hair cell electromechanics. SOAEs with similar spectral characteristics have been observed in both mammals and non-mammalian species such as lizards, suggesting a common mechanism in their generation, despite striking differences in inner ear physiology between mammals and non-mammals. In this work, a model based on coupled limit-cycle oscillators (Vilfan and Duke, *Biophys. J*, vol. **95**, pp. 4622–4630, 2008) and a model based on standing waves (Bowling *et al*., *Sci. Rep*., vol. 11, pp. 1–14, 2021) are implemented to analyze their ability to predict key SOAE characteristics. As in experiments, the models predict discrete peaks in the SOAEs with quasi-periodic spacing. However, if noise is neglected, the SOAE peaks are unrealistically sharp. In this work, the effect of noise on SOAE spectral properties (e.g., bandwidth of SOAE peaks) and statistical properties (e.g., interpeak cross-correlations) is evaluated to assess whether either model is a better representation of SOAE generation. This is a key step in understanding the theoretical underpinnings of SOAE generation. [Research funded by NIH grant R01 DC016114.]

1:15

3pAB2. Source separation with an acoustic vector sensor for terrestrial bioacoustics. Irina Tolkova (Harvard Univ., School of Eng. and Appl. Sci., Cambridge, MA 02134, itolkova@g.harvard.edu) and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Passive acoustic monitoring is emerging as a low-cost, non-invasive methodology for automated species-level population surveys. However, systems for automating the detection and classification of vocalizations in complex soundscapes are significantly hindered by the overlap of calls and environmental noise. We propose addressing this challenge by utilizing an acoustic vector sensor to separate contributions from different sound sources. More specifically, we describe and implement an analytical pipeline consisting of (1) calculating direction-of-arrival, (2) decomposing the azimuth estimates into angular distributions for individual sources, and (3) numerically reconstructing source signals. Using both simulation and experimental recordings, we evaluate the accuracy of direction-of-arrival estimation through the active intensity method (AIM) against the baselines of white noise gain constraint beamforming (WNC) and multiple signal classification (MUSIC). Additionally, we demonstrate and compare source signal reconstruction with simple angular thresholding and a wrapped Gaussian mixture model. Overall, we show that AIM achieves higher performance than WNC and MUSIC, with a mean angular error of about 5°, robustness to environmental noise, flexible representation of multiple sources, and high fidelity in source signal reconstructions. Finally, we illustrate applications of AVS-based source separation for spatiotemporal analysis of birdsong in dawn chorus recordings.

1:30

3pAB3. EEG study of attention on an auditory target detection task in dolphins and humans. Matt Schalles (Neurosci. Inst., Carnegie Mellon Univ., 4400 Fifth Ave., Pittsburgh, PA 15213, matt.schalles@gmail.com), Alexander Pei (Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), Abigail Noyce (Psych., Carnegie Mellon Univ., Pittsburgh, PA), Jason Mulsow, Dorian Houser (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, San Diego, CA), Peter Tyack (Biology, Woods Hole Oceanographic Inst., Falmouth, MA), and Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA)

How do dolphins attend to echos and ignore background noise? Is the process similar to how humans attend to conversations in a crowded room? Studies of human selective attention have highlighted endogenous brain processes modulating the magnitude of early responses to sounds in auditory evoked potentials (AEPs), yet these rely on task-specific instructions difficult to employ in animal studies. We trained an adult male dolphin (*Tursiops truncatus*) to attend to a stream of rapidly presented tones, and provide a whistle response to a "target" tone while withholding responses to "background" tones in an amplitude discrimination task. The background sounds were designed to function as a tonal mismatch negativity paradigm with two frequencies: a standard tone presented 80% of the time, and a deviant presented 20% of the time. Depending on condition, the target could be the same frequency as the standard or the deviant. We observed an enhanced AEP response to a deviant of the same frequency as the target, while the AEP response to the standard-target condition was reduced. This was not predicted by previous human experiments, and we are currently following up with a pilot a human task modeled after the dolphin implicit target detection task.

1:45

3pAB4. Multispecies discrimination of seals (pinnipeds) using hidden Markov models (HMMs). Marek B. Trawicki (Marquette Univ., 1313 W. Wisconsin Ave., Milwaukee, WI 53233, marek.trawicki@marquette.edu)

Hidden Markov models (HMMs) were developed and implemented for the discrimination of five available Seals (Pinnipeds), namely, the Bearded Seal (*Erignathus Barbatus*), Harp Seal (*Pagophilus Groenlandicus*), Leopard Seal (*Hydrurga Leptonyx*), Ross Seal (*Ommatophoca Rossii*), and Weddell Seal (*Leptonychotes Weddellii*). The main objectives of the experiments were to study the impact of the frame size and step size and number of states for feature extraction and acoustic models on classification accuracy. Based on the experimentation using Mel-Frequency Cepstral Coefficients (MFCCs) extracted from the vocalizations (15 ms frame size and 4 ms step size), HMMs containing 20 states with single underlying Gaussian mixture model (GMM) produced discrimination of 95.77%. From the results, the framework could be applied to analysis for other marine mammals for both classification and detection of vocalizations and species.

2:00

3pAB5. Production and perception of vocalizations by a mouse model of autism. Payton Charlton (Psych., Univ. at Buffalo, Park Hall Rm B80, 211 Mary Talbert Way, Buffalo, NY 14260, paytonch@buffalo.edu), Sevda Abdavinejad (Evolution, Ecology, and Behavior, Univ. at Buffalo, SUNY, Buffalo, NY), and Micheal Dent (Univ. at Buffalo, SUNY, Buffalo, NY)

Autism spectrum disorders (ASDs) are neurodevelopmental disorders that 1 in 54 children are diagnosed with by the age of 8. ASD is characterized by social communication deficits and stereotyped, repetitive behaviors. Several researchers over the years have indicated hearing abnormalities in children with ASD, but it is not clear if these abnormalities contribute to communication deficits. Researchers have utilized several mouse models of autism to study communication deficits as these mouse models produce abnormal ultrasonic vocalizations (USVs) which have similarities to human communication signals. However, very few studies have investigated the hearing abilities of these mouse models, and none have looked at both the production and perception of USVs. We recorded USVs from BTBR to compare the number and proportion of USVs produced in three different social contexts. We also trained a group of BTBR and C57BL/6 mice using an operant conditioning paradigm with positive reinforcement to collect absolute thresholds for four USVs (chevron, upsweep, downsweep, and complex). Absolute thresholds for each strain were compared for differences. These experiments can provide important data for the relationship between perception and production in animal models of ASD.

2:15

3pAB6. Longitudinal effects of noise exposure and age on masked auditory brainstem response-derived estimates of auditory filters in laboratory mice. Kali Burke (Johns Hopkins Univ., 4314 Hampton Hall Ct, Belcamp, MD 21017, kaliburk@buffalo.edu), Grace Capshaw, Alexandra Wong (Johns Hopkins Univ., Baltimore, MD), Micheal Dent (Univ. at Buffalo, SUNY, Buffalo, NY), and Amanda M. Lauer (Johns Hopkins Univ., Balitmore, MD)

The peripheral auditory system in normal-hearing individuals is highly tuned to discriminate very minor changes in frequency, but this frequency resolution may be susceptible to the effects of age and noise. Unfortunately, high-intensity noise damages the auditory periphery and leads to temporary or permanent hearing loss. Using a streamlined method to measure auditory brainstem responses (ABRs) in simultaneous spectrally notched noise, we measured masked thresholds and estimated peripheral frequency selectivity in male and female CBA/CaJ mice across the lifespan before and after noise exposure (8–16 kHz narrowband noise at 100 dBA SPL for 2). Mice were grouped by age at the time of noise exposure (44, 144, and 479 days old) and tracked longitudinally. We recorded ABRs to a 16 kHz tone in quiet, in the presence of a broadband masker, and in a masker with a spectral notch of varying widths ($1/8^{th}$,$1/4^{th}$, 1/2, 1, and 2 octaves centered around 16 kHz) before and after noise exposure for up to 6 months post-exposure. Our findings show that masking differs with age and time after noise exposure, and that young adult mice (<2 months old at noise exposure) are especially susceptible to traumatic noise.

2:30

3pAB7. Behavioral measurements of hearing across the lifespan in several strains of mice. Mariam Ashour (Psych., Univ. at buffalo, Dept. of Psych., Buffalo, NY 14260, mashour@buffalo.edu), Payton Charlton (Psych., Univ. at Buffalo, Buffalo, NY), Zachary Zaharkin (Biology, Univ. at Buffalo, Buffalo, NY), Sevda Abdavinejad (Evolution, Environment, and Behavior, Univ. at Buffalo, Buffalo, NY), Cade Leinbach (Psych., Univ. at Buffalo SUNY, North Tonawanda, NY), and Micheal Dent (Univ. at Buffalo, SUNY, Buffalo, NY)

Preliminary studies have indicated a correlation between hearing loss and dementia. However, the relationship between these two factors remains unclear. Utilizing mouse models of Alzheimer's disease (AD), researchers can begin to understand the relationship between dementia and hearing loss. The present study examined hearing in two AD mouse models (5xFAD and APP/PS1) along with C57BL/6 control mice. Hearing thresholds for 14 kHz tones were collected from young and old, male and female mice using an operant conditioning nose poke procedure with positive reinforcement. Daily thresholds were collected for each mouse and plotted with regression lines to assess the rates of hearing loss. A linear mixed-effects model determined that thresholds differed by age, sex, and genetic background. These studies are important for understanding more about hearing in mouse models of AD, as well as the relationship between AD and hearing loss in the human population.

2:45

3pAB8. *In situ* approach to characterize effects of pile-driving on black sea bass behavior, *Centropristis striata*. Sierra Jarriel (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd. MS#50, Woods Hole, MA 02543, sierra.jarriel@whoi.edu), Youenn Jezequel, Seth Cones, Nathan Formel, Nadége Aoki, Jenni Stanley, and T. Aran Mooney (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA)

With the threat of climate change and global energy demands rapidly growing, offshore wind farms are quickly becoming a contender in renewable energy strategies in countries around the world, including the United States. In the US, numerous offshore wind turbines are in the planning process for coastal Mid-Atlantic waters, spurring investigation into potential effects of the high-intensity construction noise on local species. Black sea bass, *Centropristis striata*, is a commercially important mid-water fish that inhabits these same waters and are reported to experience behavioral changes in response to pile-driving, although only demonstrated in the lab. Verification and expansion on these results requires an *in situ* approach to minimize tank effects on sound propagation and animal behavior. This study, conducted in coastal waters of Woods Hole, MA, used video observation to characterize behavior of 40 caged black sea bass before and during 15-min bouts of actual impact and vibratory hammer pile driving, repeated 5 times per day. Pressure and particle motion were measured. We examined immediate reactions to the onset of noise, prolonged behavioral changes, and habituation. Initial results show increased hiding or sheltering behaviors, as well as bursts of escape-motivated movements in response to the noise. While preliminary, the results suggest that pile driving can induce a range of behavioral impacts on this important fisheries species, and some concern is warranted with respect to offshore wind development.

Session 3pBA

Biomedical Acoustics: Biomedical Acoustics Student Paper Competition

Kevin J. Haworth, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

Kenneth B. Bader, Cochair
University of Chicago, 5835 South Cottage Grove Ave., Department of Radiology, MC 2026, Q301B, Chicago, IL 60637

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.

1aBAa6. Super-resolution cellular ultrasound imaging via localization of nanodroplets
Student author: John Kim

1aBAa7. Direct spatiotemporal localization of microbubble trajectories for highly-resolved hemodynamics in ultrasound localization microscopy
Student author: Alexis Leconte

1aBAa9. Ultrasonic subwavelength imaging with blind structured illumination
Student author: Jinuan Lin

1aBAb3. Drug-mediated acoustic reporter genes for mammalian cell ultrasound imaging
Student author: Phoebe J Welch

1aBAb4. Cellular response to photo-mediated ultrasound therapy
Student author: Madhumithra Subramanian Karthikesh

1aBAb6. Ultrasound-spectroscopic-imaging study on the effects of local mechanical stimulation on living human induced pluripotent stem cells
Student author: Natsumi Fujiwara

1aBAb10. Investigation of the ultrasound-mediated toxicity mechanisms of various sonosensitive drugs
Student author: Kritika Singh

1aBAb11. Tracking macrophages with ultrasound
Student author: Ashley Alva

1aBAb12. Mechanistic study of blood brain barrier opening by microbubbles and focused ultrasound
Student author: LuNa Hu

1pBAa3. Development of binder-jetting based skull phantoms for transcranial ultrasound research
Student author: Kazi Safowan Shahed

1pBAa4. A model-based image reconstruction algorithm for near real-time transcranial photoacoustic imaging
Student author: Hyungjoo Park

1pBAb4. Shear wave elastography (SWE) for the measurement of lens elasticity in the context of monitoring a presbyopia treatment by ultrasonic cavitation
Student author: Alice Ganeau

1pBAb6. Modulation of the blood-retina-barrier permeability by focused ultrasound: Computational and experimental approaches
Student author: Sam Bleker

2aBAa3. Understanding the mechanisms of ultrasound-targeted microbubble cavitation-mediated blood brain barrier opening
Student author: Grace E Conway

2aBAb6. Cardioprotective efficacy of ultrasound-targeted Nitrofatty acid microbubbles in rat myocardial ischemia-reperfusion injury model
Student author: Muhammad Wahab Amjad

2aBAb8. In vivo soft tissue aberration correction for histotripsy using acoustic cavitation emissions
Student author: Ellen Yeats

2aBAb10. Heterogenous angular spectrum approach based holograms for trans-skull focused ultrasound therapy
Student author: Pradosh Pritam Dash

2pBAb6. Feasibility of MRI-guided focused ultrasound-mediated intranasal delivery in a large animal model
Student author: Siaka Fadera

2pBAb8. Dynamics of focused ultrasound-enhanced glymphatic transportation in the mouse brain
Student author: Yan Gong

2pBAb11. Shaveless focused-ultrasound-induced blood-brain barrier opening in mice
Student author: Lu Xu

3aBAa2. Random matrix theory (RMT) to quantify scattering behavior in lung mimicking phantoms
Student author: Zihan Dong

3aBAa4. Random matrix theory to quantify micro-structural changes in rodent lungs due to pulmonary diseases
Student author: Azadeh Dashti Cole

3aBAa7. Effectiveness of transferring ultrasound deep learning models from adults to pediatrics for frame based pneumonia classification
Student author: Russell Thompson

3aBAa8. Coronavirus disease 2019 patients prognostic stratification based on low complex lung ultrasound video compression
Student author: Umair Khan

3aBAa9. Synthetic lung ultrasound data generation using autoencoder with generative adversarial network
Student author: Noreen Fatima

4aBAa7. A Scholte wave based ultrasound elastography method for imaging superficial tissue
Student author: Abdullah A Masud

4aBAa9. A simulation framework for pulse wave and vector flow imaging using fluid-structure interaction and FIELD-II simulations
Student author: Pengcheng Liang

4aBAa10. Improving group velocity based estimates of arterial stiffness
Student author: Charles Capron

4aBAa12. A thermally-polarized 129Xenon phantom for MR elastography studies in a ultra-high field MRI system
Student author: Irene Canavesi

4aBAb3. Investigating the change in point spread function and resolution of random apodization passive cavitation images
Student author: Weston P Gaskins

4aBAb4. Contrast-specific imaging of histotripsy: Chirp-coded subharmonic imaging combined with volterra filtering
Student author: Vishwas Trivedi

4aBAb7. Comparison of passive beamformers for isolating cavitation activity originating in the spinal canal
Student author: Andrew Paul Frizado

4pBAa9. Nonlinear least-squares estimation of shear modulus and shear viscosity in viscoelastic media
Student author: Nicholas A. Bannon

4pBAa10. Homogeneity versus inhomogeneity in muscle elastography
Student author: Lara Nammari

4pBAa11. Uniaxial prestress and waveguide effects on estimates of the complex shear modulus using magnetic resonance elastography in a transverse isotropic muscle phantom and excised muscle
Student author: Melika Salehabadi

4pBAa12. Elastography using torsional wave motion in transverse isotropic material
Student author: Aime Luna

4pBAb9. Contribution of bubble activity to the efficacy of histotripsy and catheter-directed recombinant tissue plasminogen activator for treatment of porcine thrombi in vitro
Student author: Shumeng Yang

5aBAa1. Monitoring radiofrequency ablation in ex vivo human liver using 3D echo decorrelation imaging augmented by deep learning
Student author: Elmira Ghahramani Z.

5aBAa2. A preliminary numerical investigation of convolutional neural Network (CNN) techniques for filtering high-intensity focused ultrasound (HIFU) noise in images
Student author: Grace Farbin

5aBAa3. Machine-to-machine transfer function: Transferring deep learning models between ultrasound machines
Student author: Ufuk Soylu

5aBAa6. Quantitative ultrasound for preterm birth risk prediction - Part 1: Statistical evaluation
Student author: Mehrdad Mohammadi

5aBAa8. Evaluating postpartum cervical remodeling with quantitative ultrasound technology
Student author: Michelle Villegas-Downs

5aBAa10. The tradeoffs between grating lobes and larger array pitch using null subtraction imaging
Student author: Mick Gardner

5aBAa14. Ultrafast ultrasound beamformer for plane wave imaging with field programmable gate array
Student author: Zhengchang Kou

5aBAa15. Three-dimensional time-domain full-waveform inversion for ring-array-based ultrasound computed tomography
Student author: Fu Li

5aBAa16. Improving attenuation imaging of the breast with the spectral log difference technique and full angular spatial compounding
Student author: Mingrui Liu

5aBAb1. The effect of gas composition on the color Doppler ultrasound twinkling artifact
Student author: Eric Rokni

5aBAb2. The effect of elevated oxygen on kidney stone twinkling in vivo and ex vivo
Student author: Laura Brownstead

5aBAb3. Using 3D printed structures to evaluate the potential causes of the color Doppler twinkling signature
Student author: Benjamin Wood

5aBAb6. Time evolution of ambient pressure sensitivity of microbubbles subharmonic response as a function of hydrostatic pressure and the filling gas
Student author: Roozbeh Hassanzadeh Azami

5aBAb11. Automated identification of a radiological beads in breast tumors for purpose of calibrating quantitative ultrasound.
Student author: Yuning Zhao

5aBAb13. Experimental detection of skull-based ultrasonic Lamb waves as an intracranial pressure monitoring method
Student author: Dan Linh Nguyen

5aBAb14. Effect of focused ultrasound peripheral nerve stimulation on muscle mechanical properties: An in vivo murine model
Student author: Jacob C Elliott

5aBAb15. Quantification of swallowing movements by ultrasound imaging in normal and disordered subjects
Student author: Nicholas S. Schoenleb

5aBAb16. Numerical simulations for dharacterization of missing articulatory information in ultrasound imaging of speech
Student author: Sarah Rotong Li

Session 3pCA

Computational Acoustics, Architectural Acoustics, Structural Acoustics and Vibration, Underwater Acoustics, Engineering Acoustics, and Physical Acoustics: Boundary Conditions Across Acoustics

Jennifer Cooper, Cochair
Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723

Michelle E. Swearingen, Cochair
Construction Engineering Research Laboratory, US Army ERDC, PO Box 9005, Champaign, IL 61826

Invited Papers

1:00

3pCA1. Study of the impact of boundary conditions on acoustical behavior of granular materials and their implementation in the finite difference method. Zhuang Mo (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907-2099, mo26@purdue.edu), Guochenhao Song (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN), Tongyang Shi (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

Granular materials display significant differences in their acoustical response when tested in a standing wave tube, compared with the behavior of more traditional sound absorbing materials such as fibrous webs and foams. The latter materials can often be modeled as an equivalent fluid with the further assumption that the material properties do not depend on the input signal level. In contrast, the level dependence of the acoustical behavior of granular materials has been observed in measurements of glass bubbles, as reported in previous studies, for example. When the input level is low, the absorption coefficient of the glass bubble stack shows solid-like behavior with multiple peaks associated with modal response of the stack. On the other hand, when the input level is high, glass bubble stacks show fluid-like behavior, with the quarter wavelength resonance in the direction of the tube axis dominating the response. In the current work, the boundary conditions at the air/granule interface and the granule/tube wall boundary are studied, as is the mechanism causing the variation of the apparent stiffness of the granule stack. The proposed model is implemented with a finite difference approach, and the model predictions are compared with acoustic measurements of granule stacks.

1:20

3pCA2. Kalimba tine boundary condition models. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu)

The kalimba or African thumb piano is a musical instrument with thin metal tines plucked by the thumbs at their free end. The opposite end is secured to a wood resonator with a three-point clamp that provides downward force in the middle, between points of upward force at the bridge and the back bar. The simplest model is an Euler-Bernoulli thin beam, free at the plucked end and clamped at the mount. Chapman [J. Acoust. Soc. Am **131**, 945 (2012)] explored the benefits to modeling the bridge as a simply supported point, with a clamped condition at the load point in the middle. Considerations in the present work are related to the modes of vibration of the resonator box. (i) The bridge is the point where vibration is transmitted to the resonator box, itself designed to better radiate sound, and thus the mechanical impedance at that point provides a complex boundary condition to be implemented in the tine model. (ii) Measurements of the attack of the kalimba tones include harmonic content not predicted by the thin beam model; recent work on this "offset" boundary condition is approached computationally, extending beyond the theoretical thin beam model.

Contributed Papers

1:40

3pCA3. Biquaternion acoustic interface conditions. Roger Oba (Acoust. Div., US Naval Reseach Lab., 4555 Overlook Ave. S.W., Washington, DC 20375, roger.oba@nrl.navy.mil)

Four Imaginary units corresponding to each space-time dimension create an geometric-algebraic framework for acoustic computations. Multiplication of oriented, spatial imaginary units anti-commute; whereas the time unit commutes with every other imaginary unit. This gives space-time represented as a single biquaternion number. Elementary wave-functions appear as biquaternion valued functions that include non-dimensionalized pressure plus velocity in a single function. In particular, space-time Fourier terms contain propagating exponentials with directional, space-time imaginary units. The pressure-velocity Snell's law for acoustic propagation into an attenuating half-space demonstrates interface conditions. Matching biquaternion pressure-velocity fields across the interface require a single, biquaternion interface condition, requiring biquaternion functions such as arctangent or logarithm. Consequently, biquaternion transmission and reflection coefficients allow for novel computation for attenuation related phase and amplitude modulation, or impedance, for example. [This research is supported by

6.2 NRL base program sponsored by the Office of Naval Research. Distribution Statement A: Approved for public release, distribution is unlimited.]

1:55

3pCA4. Boundary conditions determined by experimentally measured surface properties for wave-based time-domain simulations. Ziqi Chen (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Boundary conditions are essential and necessary when simulating sound wave propagation inside an enclosed space. In this work, complex-valued surface acoustical properties for the actual materials are obtained from impedance tube measurements. This work applies model-based Bayesian inference to estimate the exact parameters for arbitrary boundary conditions. The Bayesian approach presented in this work demonstrates the capability to predict the complex-valued surface boundary function of frequency in broadband ranges. This approach is equivalently applicable in different boundary models in time domain depending on wave-based numerical methods used for acoustic modelings, such as the spectral element method or finite-difference time-domain method. With the accurately estimated boundary models, the efficacy of the boundary conditions is explored within the time-domain approaches.

WEDNESDAY AFTERNOON, 10 MAY 2023 — ARMITAGE, 1:00 P.M. TO 2:00 P.M.

Session 3pED

Education in Acoustics: Acoustical Education Prize Lecture

John R. Buck, Cochair
Electrical and Computer Engineering, UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—1:00

Invited Paper

1:05

3pED1. DJ Prof: Reflections on teaching. Kathleen E. Wage (George Mason Univ., 4400 University Dr., Fairfax, VA 22151, kwage@gmu.edu)

In a short-lived career as a cartoonist for Acoustics Today [Summer 2016], I sketched DJ Prof, an acoustics professor who mixes multiple modes of instruction to engage and excite students. DJ Prof illustrates the concept of active learning, which Freeman *et al.* called the "preferred empirically validated teaching practice" [PNAS, 2014]. Numerous studies show that active learning courses improve student learning, increase retention rates, and reduce performance gaps in STEM for economically disadvantaged students and females in male-dominated classes. This lecture reflects on my evolution as DJ Prof using examples from acoustic signal processing courses. I will review the literature on active learning, share a few favorites from my pedagogical playlist, and highlight open questions for the acoustics education community. The session will include interactive exercises. Please bring an open mind, a sense of humor, and a willingness to meet the people sitting near you.

Session 3pID

Interdisciplinary and Student Council: Hot Topics in Acoustics

Ann Holmes, Cochair
Psychological & Brain Sciences, University of Louisville, 2082 Douglass Blvd., Apt 5, Louisville, KY 40205

Ferdousi Sabera Rawnaque, Cochair
Penn State University, 210 East Hamilton Avenue, Apt. 31, State College, PA 16801

Samuel D. Bellows, Cochair
Physics and Astronomy, Brigham Young University, N247 ESC Provo, UT 84602

Zane T. Rusk, Cochair
The Pennsylvania State University, 104 Engineering Unit A, University Park, PA 16802

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Acoustical methods for remote sensing in seagrass meadows. Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kyle Capistrant-Fossa (Univ. of Texas at Austin, Port Aransas, TX), Andrew R. McNeese, Colby W. Cushing, Thomas S. Jerome, Preston S. Wilson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kenneth H. Dunton (Univ. of Texas at Austin, Port Aransas, TX)

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. After a brief historical overview, this talk will present results from both active and passive acoustical methods for ecosystem monitoring in seagrass meadows. From a propagation perspective, 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. Examples of both techniques will be presented based on data collected as part of an 18-month continuous deployment of an acoustical measurement system operating in a moderately dense seagrass bed dominated by Thalassia testudinum (turtle grass) in Corpus Christi Bay, Texas. 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 NSF.]

1:25

3pID2. A hearing aid "test drive": Using virtual acoustics to accurately demonstrate hearing aid performance in realistic environments. Matthew Neal (Otolaryngol. and Comm. Disord., Univ. of Louisville and Heuser Hearing Inst., 117 E Kentucky St., Louisville, KY 40203, matthew.neal.2@louisville.edu) and Pavel Zahorik (Otolaryngol. and Comm. Disord., Univ. of Louisville and Heuser Hearing Inst., Louisville, KY)

A common challenge for audiologists when fitting hearing aids is that quiet clinics are unlike the noisy, reverberant places where patients report difficulty. This prevents direct user feedback about hearing aid performance. Virtual acoustics can allow a patient to experience a hearing aid in such environments, without leaving the clinic. A virtual reality (VR) audio-visual demonstration has been created which "test drives" hearing aid features in real-world scenarios, dynamically rendered over custom wired hearing aids and headphones. The scenes were created from 360° photographs and acoustic measurements made in real rooms. A room acoustic model was used for the simulation, tuned to match the measured data. Audio was rendered using principal component-base amplitude panning (PCBAP), which can provide dynamic VR audio with broadband magnitude and phase accuracy. This talk will provide comparisons between PCBAP and higher-order Ambisonics (HOA) for rendering hearing aid beamformers in anechoic and reverberant environments. Results show that PCBAP can render a beamformer's directivity broadband with 2 to 3 dB error (95%) using only 36 filters in both anechoic

3p WED. PM

and reverberant conditions. Even seventh-order HOA (64 filters) does not achieve similar performance. The talk will also include videos of the demonstration, highlighting viability for clinical applications.

1:45

3pID3. High amplitude time reversal focusing of sound and vibration. Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Dept. of Phys. & Astron., N245 ESC, Provo, UT 84602, bea@byu.edu)

Time reversal (TR) is a signal processing technique that can be used to focus high amplitude sound or vibration at a desired location. TR focusing can be done with sources placed far from the desired focal location and the technique excels in complex environments. The impulse response between each source and the desired focal location must be obtained prior to the focusing and the environment must remain relatively unchanged for successful focusing. Multiple scattering or reverberation of waves off of many reflecting surfaces in the environment can actually be used advantageously by exploiting those reflections as additional image sources. This talk will provide an introduction to TR and then focus on the use of TR to provide high amplitude focusing of sound and vibration. Applications of high amplitude TR include lithotripsy of kidney stones, histotripsy of lesions, and locating cracks and defects in structures such as in human teeth, spent nuclear fuel storage casks, airplane wings, and automotive bearing caps. Recently, high amplitude TR of airborne sound has been studied in a reverberation chamber to generate a focused difference frequency and to study the nonlinear acoustics of the peak sound levels of 200 dB that have been attained.

WEDNESDAY AFTERNOON, 10 MAY 2023 INDIANA/IOWA, 1:00 P.M. TO 2:40 P.M.

Session 3pMU

Musical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Music Synthesis Techniques Applied to Sound Design and Sonification (Hybrid Session)

Jill A. Linz, Chair
Physics, Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866

Chair's Introduction—1:00

Invited Papers

1:05

3pMU1. Design choices in Max and Pure Data. Miller Puckette (Music, UCSD, 9500 Gilman Dr., La Jolla, CA 92093, msp@ucsd.edu)

Max and Pure Data are widely used patch languages for real-time media computation, especially live electronic music. Each program was developed with a particular musical production in mind: Philippe Manoury's Pluton (1998) for Max, and Sorensen/Steiger/Puckette's Lemma 1 in 1997 for Pure Data. At the same time, care was taken to make the features of both programs as aesthetically neutral as possible, in order to be useful in as wide a variety of applications as possible. One design principle has been to avoid encoding musical structures such as time or key signatures in either program. Noticeably absent is any built-in facility for machine learning or artificial intelligence. On the other hand, heuristics from programming language design such as modularity, encapsulation, and abstraction have been fundamentally influential in the designs of both Max and Pure Data.

1:25

3pMU2. Musical instruments as platforms for open sound design. Owen Osborn (224 S Maple St., Ambler, PA 19002, owen@critterandguitari.com) and Chris Kucinski (Ambler, PA)

The combination of open-source audio software and the miniaturization of computing hardware has led to new customizable musical instruments. Artists and sound designers can alter the musical behavior and sound character of such instruments themselves. This opens new opportunities for users to explore sound and fosters community collaboration. We will discuss the design of such instruments and introduce the Organelle, an instrument designed to run software that users can create using popular open-source music environments like Pure Data and Faust.

1:45

3pMU3. Modeling coupled oscillators: Applications for musical sequencing and synthesis. Jazer G. Sibley-Schwartz (Music, Mount Holyoke College, 16 Hooker Ave., Northampton, MA 01060, jazergiles@gmail.com)

There are many naturally occurring systems of coupled oscillators, e.g., the synchronization of firefly flashes, frog calls, and electrical impulses in the brain. The behavior of a large system of coupled oscillators can be approximated using the Kuramoto-Daido model. The model shows that phase calculations for each oscillator can be simplified by comparing each oscillator's phase to the average phase of the collection, eliminating the need to sum over the whole group. In this session, we will explore this algorithm in Max/MSP. We will look at simple systems with a small number of low frequency oscillators, both free-running and rhythmically quantized/Euclidean, as a means of generating rhythmic patterns and sequences. Then we will look at larger collections of oscillators within the audible frequency range and explore how the algorithm can be used to modulate audio signals for timbral effects. This example will delve into how the coupling algorithm affects oscillators that have frequency relationships drawing from the overtone series. We will also see how delaying the averaged signal simulates a proximity effect that more closely resembles the localized synchronization that occurs in natural systems.

2:05

3pMU4. Observation of torsion on bowed strings. Robert Mores (Hamburg Univ. of Appl. Sci., Finkenau 35, Hamburg, Deutschland 22081, Germany, robert.mores@haw-hamburg.de)

Research on bowed string motion focuses on transverse waves and not so much on torsional waves. These are believed to play only a minor role for stabilizing vibrations and no role for perception. Here, torsion is measured on both sides of the bow contact point for a variety of bridge-bow distances on a cello string. Electromagnetical and optical measurements correspond. Every periodic string release is preceded by a reverse torsional motion independent from bowing position or dynamics. Transverse and torsional motions are coupled and there are cases of stabilization, but also cases of perturbation or surrender. Structural and timing analyses of torsional waves suggest that the earlier concepts of differential slipping can be essentially confirmed. Other concepts cannot be confirmed, such as the Schelleng ripples, or are under question, such as the concept of subharmonics.

Contributed Paper

2:25

3pMU5. Sonification of ocean data in art-science. Colin Malloy (Music, Univ. of Victoria, 3800 Finnerty Rd., Music Dept., Victoria, BC V8P 5C2, Canada, malloyc@uvic.ca)

ArtScience is a burgeoning field that promotes bi-directional collaboration between scientists and artists. Artists learn from scientists to inform the creation of new works based on scientific ideas and data. Scientists learn from artists to explore data in new ways and learn new ways of communicating. In 2022, Colin Malloy was the artist in residence for Ocean Networks Canada and created many new musical works based on ocean data. He explored many ways of sonifying ocean data using novel mappings in multiple original solo electroacoustic percussion compositions. Some mappings were more direct—such as using data for wavetables—while others were less so—such as probabilistic trigger mappings based using datasets as probability distributions. This paper presents his methods for sonifying ocean data used for sound production in these musical compositions.

Session 3pNS

Noise, Architectural Acoustics, Underwater Acoustics, Engineering Acoustics, and Acoustical Oceanography: Post Pandemic Soundscapes Part II (Hybrid Session)

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

Invited Papers

1:00

3pNS1. Developing a taxonomy of soundscape design from real-world examples. Xiaochao Chen (Inst. for Environ. Design and Eng., The Bartlett, Univ. College London, Central House, 14 Upper Woburn Pl., London WC1H 0NN, United Kingdom, xiaochao.chen.20@ucl.ac.uk), Francesco Aletta (Inst. for Environ. Design and Eng., The Bartlett, Univ. College London, London, United Kingdom), Cleopatra Moshona (Eng. Acoust., Inst. of Fluid Dynam. and Tech. Acoust., Technische Universität Berlin, Berlin, Germany), Helen Henze (Eng. Acoust., Inst. of Fluid Dynam. and Tech. Acoust., Technische Universität Berlin, Berlin, Germany), Andrew Mitchell, Tin Oberman, Huan Tong (Inst. for Environ. Design and Eng., The Bartlett, Univ. College London, London, United Kingdom), Andre Fiebig (Eng. Acoust., Inst. of Fluid Dynam. and Tech. Acoust., Technische Universität Berlin, Berlin, Germany), Jian Kang (Inst. for Environ. Design and Eng., The Bartlett, Univ. College London, London, United Kingdom), and Brigitte Schulte-Fortkamp (Eng. Acoust., Inst. of Fluid Dynam. and Tech. Acoust., Technische Universität Berlin, Herzogenrath, Germany)

Over the past 20 years, there has been a significant increase in soundscape studies focusing on urban scenarios. Policymakers are increasingly attracted to the concept of soundscapes, as reflected in several documents issued by international agencies that advocate a more user-centred approach when considering the urban acoustic environment. Despite widespread interest in the concept of soundscape, a consensus and systematic review of soundscape design and interventions, as well as a summary of empirical evidence on the benefits of sound methods, are still lacking. The catalogue of soundscape interventions (CSI) project aims to provide a tool for data collection and communication on soundscape practice, with the long-term goal of compiling frequent/recurring soundscape strategies into a "design toolkit" and developing a "design brief" to facilitate communication between local authorities and soundscape consultants and researchers. A platform for collecting examples of soundscape intervention has been published online, and the project is currently in its second phase of populating the example database. When sufficient practices have been gathered, a taxonomy of soundscape design will be developed, which will eventually become a "design toolkit." To ensure that all perspectives are represented, and the taxonomy is based on consensus, stakeholders will be consulted. If a significant increase in the number of soundscape practices is detected, the taxonomy may require revision.

1:20

3pNS2. Soundscape evaluation based on participatory design. Yichun Lu (Dept. of Architecture, National Univ. of Singapore, 4 Architecture Dr., Singapore 117566, Singapore, e0554244@u.nus.edu) and Siu-Kit LAU (Architecture, National Univ. of Singapore, Hong Kong, Hong Kong)

The traditional soundscape assessment method assesses people's perception of a selected acoustic environment using questionnaires or interviews. However, fewer studies have been done to investigate other methods for methodological triangulation. This research aims to propose an interactive approach for soundscape assessment research based on participatory design and apply this method to study the relationship between audio-visual elements and soundscape perceptions in residential areas. The experiment was conducted in a virtual reality (VR) environment. Thirty-two participants were invited to interactively create the audio-visual environment, which they thought was the most eventful or vibrant in outdoor and indoor contexts. The result indicated that the descriptor "eventfulness" could be related to "vibrancy" since participants selected similar audio-visual elements for these two descriptors except for building and greenery. Moreover, participants preferred to model open visual environments and more greenery in vibrant scenes, while they preferred semi-closed buildings and less greenery in eventful scenes. Besides, the result revealed that people in the indoor context might be more sensitive to traffic sound than in the outdoor context. Those results were consistent with those of traditional assessment methods, which means this method may be applied for a methodological triangulation in soundscape evaluation.

1:40

3pNS3. Soundscape study to develop a new noise ordinance. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and Tiago F. Mattos (Brooks Acoust. Corp., Pompano Beach, FL)

A soundscape study was conducted to assist with the development of a new noise ordinance in the City of Delray Beach, Florida. Delray Beach has an active entertainment district, which is popular with tourists and locals. Integrated into the district and the surrounding area are recent residential and business developments. During the pandemic the entertainment district activity was reduced. Since 2021, the activity was resumed. With the intensified activity noise complaints to the City increased. The existing subjective noise ordinance was not effective. A soundscape study was conducted to determine the existing conditions, and to develop new data-driven, perception-based objective standards. The goals for the new ordinance were to create certainty for stakeholders, to balance the interests of businesses and residents, to manage expectations, and to ensure fairness and consistency in application. Long-term and short-term acoustical measurements were made, several public workshops and hearings were held and two soundwalks were conducted. The results of data collection are discussed. At the time of this writing, the draft of the new ordinance is in process, to be presented to the City Commission for consideration. The outcome of this effort will be presented.

2:00

3pNS4. Urban soundscape studies via soundwalks in Nashville. David S. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

Soundscape studies were conducted in Nashville, Tennessee in conjunction with the Acoustical Society fall meeting in 2022. Nashville is a growing city known for its musical entertainment district (Broadway), which post-pandemic attracts numerous tourists year round. It also has other nearby surrounding areas which support many businesses and residences. Both the entertainment district and its surrounding areas were the focus of this study. Three pre-soundwalk acoustical tests were conducted during site surveys to determine the soundwalk path and data collection locations. In December 2022, a soundwalk was conducted to collect participant perception data in accordance with ISO 12913 methods. The results of the physical sound measurements and the soundwalk perception data will be discussed. The effect of transient events on the sound environment are evaluated. In addition, potential interventions to improve existing conditions are proposed

Contributed Papers

2:20

3pNS5. Road traffic noise informational masking with water sound sequences: From laboratory simulation to field study. Jian Li (Dept. of Architecture and Industrial Design, Univ. of Campania "Luigi Vanvitelli," Via San Lorenzo - Abazia di San Lorenzo, SENS-i Lab, Aversa, Campania 81031, Italy, jian.li@unicampania.it), Luigi Maffei, Aniello Pascale, Massimiliano Masullo (Dept. of Architecture and Industrial Design, Univ. of Campania "Luigi Vanvitelli", Aversa, Italy), Minqi Lin (Dept. of Bldg. Environment and Energy Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), and Chi-Kwan Chau (Dept. of Bldg. Environment and Energy Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

Road traffic noise control in urban green space is a big concern for urban designers and public managers. The introduction of water sounds into noisy environment has been proven effective based on the soundscape approach. To extend more effective and applicable strategies for water sound informational masking, the exploration of the spatial settings of virtual water sound playbacks in urban parks were conducted both in the laboratory and field settings. Three different spatial water-sound sequences were added into the virtual noisy environment through an immersive spatial audio system and the real urban green park through the digital audio programming of bluetooth loudspeakers. The mental activities and subjective feelings of two group subjects were evaluated by a portable electroencephalogram (EEG) measurement with a post-doc questionnaire. The better masking effects introduced by the spatial settings of water sounds had been confirmed from the results of more positive emotional feedbacks and more relaxed mental state revealed by the spectral power of alpha band across two experimental conditions. Especially, the two-position switching water sounds brought more attentional network activations. Moreover, more sensory accumulation effects reflected by the mental network activations were observed from the brain activities in the *in situ* measurement compared to laboratory-setting.

2:35

3pNS6. A preliminary field experiment to predict soundscape perception based on the biophilia hypothesis paradigm. Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Faculty of Art, Design and Architecture, Dept. of Interior Architecture and Environ. Design, Ankara 06800, Turkey, semiha@bilkent.edu.tr) and Enkela Alimadhi (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

In a time when biophilic design is investigated mainly with the incorporation of greenery in indoor environments, this study seeks to investigate the relationship between biophilic quality and soundscape perception in oncology polyclinics with and without a garden in Ankara, Turkey. The study, which evolves in an actual environment, claims that the Biophilia Hypothesis paradigm can predict the soundscape perception. Therefore, it aims to examine the audio-visual interaction manifested in polyclinics and its effect on both soundscape perception and biophilic quality using the Biophilic Interior Design Index (BIDI) and soundscape analysis ISO/TS-12913-3 method A. For the objective characterization of the environment, both binaural audio recordings and photographs have been taken, whereas the BIDI has been back-translated by two bilingual experts. In line with the existing literature, the study's preliminary result supports the tendency for polyclinics with a garden to score higher in biophilic quality, therefore soundscape perceptual affective quality is placed on the positive dimension of the circumplex model, whereas polyclinics without a garden score the opposite.

2:50–3:30

Panel Discussion

Session 3pPAa

Physical Acoustics and Structural Acoustics and Vibration: Acoustics and Elasticity of Consolidated and Unconsolidated Granular Materials III (Hybrid Session)

John Yoritomo, Cochair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, D.C. 20375

Richard Weaver, Cochair
Physics, Univ Illinois, 1110 w green, Urbana, IL 61801

James Bittner, Cochair
Michigan Technological University, P.O. Box 22, Houghton, MI 49931

Chair's Introduction—1:00

Invited Papers

1:05

3pPAa1. The effect of stress and saturation on low-frequency nonlinear elasticity in geological materials. Harrison Lisabeth (Geophys., LBNL, 1 Cyclotron Rd., Berkeley, CA 94720, hlisabeth@lbl.gov)

Wave propagation in rocks is typically treated in seismology as a purely linear elastic phenomenon; however, evidence for nonlinear elastic behavior in geomaterials has existed for years. The degree of nonlinearity in fractured material is much greater than that in intact material, resulting in signals that are highly sensitive to the state of the fracture (stress, chemistry, fluids). Elastic nonlinearity can cause a propagating wave to distort, resulting in generation of harmonics and multiplication of waves of different frequencies. One way to exploit these behaviors is to propagate waves of differing frequencies through a material and observe nonlinear wave mixing phenomena, a technique referred to in the literature as nonlinear wave modulation spectroscopy (NWMS). I will discuss the use of NWMS for the study of geologic materials at a range of conditions to highlight the frequency, stress, and fluid dependence of nonlinear parameters in rocks. Analysis of various dependencies can provide insight into the mechanisms of nonlinearity and has the potential to provide methods of monitoring the dynamic state of the subsurface.

1:35

3pPAa2. Control of guided modes propagating in unconsolidated granular layers : Combined effects of local resonators and nonlinear elasticity. Pierric Mora (Lab. GERS / GeoEND, Université Gustav Eiffel, Nantes, France), Mathieu Chekroun (Institut d'Acoustique - Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), and 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)

In this talk, we analyze the attenuation, by a line of local resonators, of guided surface modes propagating in an unconsolidated granular layer. Local resonators are made of dense solid parts added to the granular layer, mobilizing the elasticity of the medium itself for the restoring force and their own mass for the inertia effects. In addition, their weight induces local elasticity changes thanks to the nonlinear elasticity of such unconsolidated granular layers. We show numerically and experimentally that the induced resonance and elasticity change are key ingredients in describing the observed transmission and reflection effects, including mode conversion, for the different guided modes observed. We demonstrate that all these ingredients can be rationally combined to enhance the shielding effect and could be used as strategies to control seismic waves on a larger scale.

2:05

3pPAa3. Finding wave equations for sediment acoustic theories. Sverre Holm (Phys., Univ. of Oslo, P. O. Box 1048, Blindern, Oslo N 0316, Norway, sverre.holm@fys.uio.no), Sri Nivas Chandrasekaran (I), and Sven Peter Näsholm (Informatics, Univ. of Oslo, Oslo, Norway)

The two main theories for wave propagation in sediments, the Extended Biot theory and the Viscous Grain Shearing (VGS) theory, have been formulated in terms of dispersion equations only. Some have considered the lack of a wave equation to be a weakness that may indicate lack of a physical basis for these theories. In the Biot theory, it is the frequency-dependent viscodynamic operator which models viscous boundary effects in pores that poses the problem. It can be formulated as a ratio of Bessel functions for circular pores and as a ratio of tanh-functions for a 2D parallel plane duct. There is no simple equivalent time domain operator for use in the wave

equation. In both geometries, the factor may, however, be approximated well with $(1 + ic\omega)^{1/2}$, where c is a constant. This factor also appears in the Cole-Davidson dielectric theory for complex media. The equivalent time domain operator, the fractional pseudo-differential operator, enables the formulation of wave equations for all three wave modes. The same operator turns out to be central for transforming the dispersion relations of the VGS theory to time-space wave equations. These wave equations enable a closer study of properties of the medium models than that obtained from dispersion relations alone.

Contributed Papers

2:35

3pPAa4. Prediction of acoustical behavior of granular material stacks as measured in a standing wave tube by using a Biot theory-based model. Zhuang Mo (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907-2099, mo26@purdue.edu), Guochenhao Song (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN), Tongyang Shi (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and J. S. Bolton (Ray W. Herrick Laboratories/School of Mech. Eng., Purdue Univ., West Lafayette, IN)

The acoustical behavior of granular materials, such as activated carbon and silica gel, has drawn attention in recent studies, due to their favorable properties such as good low frequency sound absorption. Like other more traditional porous materials, granular materials can also be tested in a standing wave tube for a convenient assessment of their acoustical properties. However, the behavior of granular materials stacked in a standing tube is more complex than that of traditional materials. For example, the response of lightweight glass bubbles reveals a clear dependence on the sound pressure level of the input signal. Also, when tested in standing wave tubes of different diameters, the same type of granular materials displays differences in their behavior. The apparent stiffness of granule pack is also related to the depth of the stack. In the present work, a model based on Biot theory is proposed, together with a consideration of the effect of the change of boundary conditions and the granule stack stiffness in different test configurations. The model is realized by using a finite difference method, and the simulation results are compared with measurements of different types of granular materials.

2:50

3pPAa5. Investigation of ultrasound propagation through dense glass bead packings immersed in water. Arthur Le Ber (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, 1 rue Jussieu, Paris 75005, France, arthur.le-ber@espci.fr), Damien Lépicier (ENS Paris-Saclay, Gif-sur-Yvette, France), Alexandre Aubry, Arnaud Tourin, and Xiaoping Jia (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, Paris, France)

Conventional ultrasound imaging generally relies on a single scattering assumption and a constant sound hypothesis. However, in dense granular sediments where the glass beads diameter is comparable to the wavelength (typically 500 μm), both hypothesizes are no longer valid, resulting in a loss of resolution and contrast, up to the situation where a target to image could totally vanish in the fog. To better characterize this strongly scattering medium, we have conducted several ultrasonic experiments in reflection and in transmission, either on the diffused wave or on the coherent one. Both the transport and scattering mean free paths are estimated in the imaging bandwidth (1–5 MHz), alongside with the phase velocity and extinction length of the coherent wave. These results are compared with an analytical model, i.e. Generalized Coherent Potential Approximation*, which considers a coated sphere buried in an effective medium. Finally, we show a dispersive beamformer which takes into account the measured phase and group velocity. This allows us to obtain a more resolved and contrasted image in such a scattering medium. * Jing X, Sheng P, Zhou M. Phys Rev Lett. 1991

3:05

3pPAa6. Combining granular aerogels with additively manufactured porous structures for broadband sound absorption. L. Paige Nobles (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Dept. of Aerosp. Eng., Wichita, KS 67260, lpnobles@shockers.wichita.edu), Yutong Xue, J. S. Bolton (Midea Corporate Res. Ctr., West Lafayette, IN), and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Our recent investigation [1] showed that the absorption behavior of granular aerogel agglomerates with average particle diameters less than 50 μm is dominated by periodically spaced, high absorption peaks and troughs. While the absorption peaks, especially at lower frequencies, are desirable from an engineering application perspective, overcoming the loss in high frequency performance remains a challenge. Furthermore, the practical application of granular aerogels is limited by the difficulty in handling them. In this work, we propose a new design for a broadband noise absorber achieved by layering the aerogel granules within a 3D printed porous network. The design provides a robust method of incorporating aerogels within noise reduction packages and can be tailored by altering the rigid porous network and the layering arrangement. In this presentation, we present our preliminary findings regarding the fabrication and testing of such hybrid sound absorbers. [1] Dasyam A, Xue Y, Bolton JS, Sharma B. Effect of particle size on sound absorption behavior of granular aerogel agglomerates. Journal of Non-Crystalline Solids. 2022 Dec 15;598:121942.

Session 3pPAb

Physical Acoustics: Ultrasound: Radiation Force and Spectroscopy

Christopher A. Mizzi, Cochair
MagLab, Los Alamos National Lab, MS E536, TA 35, Bldg. 127, Room C108, Los Alamos, NM 87545

Robert Lirette, Cochair
Communications Technology Laboratory, National Institute of Standards and Technology, 306 29th St., Boulder, CO 80305

Contributed Papers

1:00

3pPAb1. Acoustic radiation force and torque on an object near a penetrable interface or impedance boundary. Blake E. Simon (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu) and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

An analytical approach for calculating radiation force and torque on a sphere or spheroid near a rigid or pressure release boundary due to an arbitrary incident acoustic field was developed recently by the authors [Proc. Meet. Acoust. **48**, 045004 (2022)]. In that approach, the linear scattering problem is solved using the method of images along with expansions of the acoustic fields in spherical or spheroidal wave functions. The expansion coefficients can then be substituted into an existing expression to obtain the radiation force and torque on the object. In the present work, the aforementioned theory is extended to include penetrable interfaces and impedance boundaries using an approximation of the interfacial boundary conditions. For fluid-fluid interfaces, the object can be placed on either side of the interface with respect to the incident field. This approximate analytical approach is computationally efficient and compares well with the results from an independent finite element model. In addition, an experiment was conducted with a plastic sphere submerged in water and positioned near a boundary. Results from the experiment are discussed in relation to the analytical model. [BES is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

1:15

3pPAb2. Extraction and manipulation of hydrocarbon droplets by ultrasound radiation. Robert Lirette (Phys. and Astronomy, Univ. of MS, 306 29th St., Boulder, CO 80305, vsr600@gmail.com) and Joel Mobley (Phys. and Astronomy, Univ. of MS, University, MS)

Acoustic tweezers are a method of using sound radiation forces to trap and manipulate objects. Previously, we demonstrated an acoustic tweezer capable of first creating the objects, droplets of CCl_4 from a reservoir, before trapping and manipulating them. We achieved this using a single transducer whose field was modified by a low-profile fraxicon (Fresnel axicon) lens exhibiting specific near-field trapping zones and far-field Bessel beam behavior which enable the trapping process. Here, we extend the method to include another class of fluids, complex hydrocarbons, using conventional SAE30 motor oil in water. Furthermore, with the higher power levels used in this study we demonstrate the discovery of a second trapping region in the field generated by our transducer. The non-contact extraction and manipulation of organic and hydrocarbon fluid samples could have significant biological and environmental applications where a container would contaminate, destroy, or otherwise change the samples under investigation.

1:30

3pPAb3. Acoustophoresis under multiple simultaneous frequencies. David B. Rear (Chemical and Biomolecular Eng., Case Western Reserve Univ., 32000 Euclid Ave., A. W. Smith, Cleveland, OH 44106, david.rear@case.edu) and Donald L. Feke (Chemical and Biomolecular Eng., Case Western Reserve Univ., Cleveland, OH)

The manipulation of suspended particles using ultrasonic fields remains an active area of research due to the many potential applications of acoustophoresis. While prior research has focused primarily on the acoustophoretic behavior of particles in response to an acoustic field of a single wavelength, relatively little work has been done to study the acoustophoretic behavior of subject to two or more concurrent frequencies of sound. The presence of multiple concurrent, resonant frequencies can lead to more complicated particle behavior spatially across the chamber than the simple node/anti-node attraction/repulsion behavior present in single-frequency chambers. In this study, we investigate the behavior of spherical particles (both single phase and core-shell) in multi-frequency chambers in both the Rayleigh (long wavelength) limit and cases where the particle size is the same order of magnitude as the wavelength of the background field. A mathematical description of the phenomena is developed in terms of the ratios of the pressure amplitudes for the individual background frequencies, including the criteria for continuously moveable nodes. Attention will also be paid to the implications of more than one simultaneous frequency on the motion of particles undergoing a change in size.

1:45

3pPAb4. Tailoring nonlinear ultrasound beams using acoustic lenses. Ahmed Sallam (Mech. Eng., Virginia Tech, 495 Old Turner St., Blacksburg, VA 24060, ahmedsallam@vt.edu) and Shima Shahab (Mech. Eng., Virginia Tech, Blacksburg, VA)

Acoustic holographic lenses, i.e., acoustic holograms, can tailor ultrasound beams to precisely construct elaborate focal patterns. Due to their valuable capabilities, acoustic holograms are being rapidly investigated for a wide range of applications in physical acoustics and biomedical ultrasound. Modeling and demonstrating acoustic holograms in linear regimes is well documented in the literature. However, Acoustic holograms have the potential to improve the precision and affordability of therapeutic high-intensity focused ultrasound (HIFU). In this work, we will show the effects of nonlinear propagation on ultrasound fields tailored by acoustic holograms. The intensity-dependent effects of harmonic generation, diffraction, and nonlinear gain on the constructed pressure pattern are discussed. In addition, an accelerated method for simulating high-intensity 3D holographic ultrasound is introduced. We will show how the proposed algorithm greatly reduces the computational load for designing and predicting high-intensity holographic ultrasound patterns. This expedites the procedure of designing the acoustic

lens, which allows for a more rapid and seamless implementation, especially in medical settings.

2:00

3pPAb5. Low frequency ultrasonic absorption in aqueous solutions from variable path through-transmission and pulse-echo measurements. Robert Lirette (Guided Wave Electromagnetics Group, National Inst. of Standards and Technol., 306 29th St., Boulder, CO 80305, vsr600@gmail.com), Małgorzata Musiał, Jason A. Widegren (Fluid Characterization Group, National Inst. of Standards and Technol., Boulder, CO), and Angela C. Stelson (Guided Wave Electromagnetics Group, National Inst. of Standards and Technol., Boulder, CO)

Acoustic absorption spectroscopy can characterize the complex intermolecular interactions within a solution. Currently, there are no wide-spread uses of the method and no commercially available systems exist that measure below 3 MHz. However, chemical processes have been shown to exist at as low as 300 kHz. Here we present measurements on aqueous solutions using two techniques: a variable path through-transmission method operating around 1 MHz and a pulse-echo method operating near 500 kHz. These methods overlap in frequency and their results agreed with each other. We also calibrated the setup using water and measured the sound speed in each sample. Finally, we present preliminary results for the absorption spectra of $MgSO_4$ and other solvents. Accurate measurements of acoustic absorption at low ultrasonic frequencies could prove to be an important tool in the pharmaceutical and chemical manufacturing industries and can lead to better understanding of intermolecular processes in this frequency range.

2:15

3pPAb6. Ultrasonic characterization and beyond: How to select a hydrogel for tissue engineering. Megan S. Anderson (Mech. and Aerosp. Eng, George Washington Univ., 800 22nd St. NW, Washington, DC 20052, andersonm@gwmail.gwu.edu), Kartik V. Bulusu, Michael W. Plesniak, Lijie Grace Zhang, and Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC)

Hydrogels have emerged as a crucial class of materials within the field of tissue engineering. There is growing interest in matching the mechanical properties of hydrogel scaffolds to tissues in the human body and optimizing these properties for cell growth and differentiation. Gelatin methacrylate (GelMA) is a well-accepted, biocompatible hydrogel with tunable mechanical properties. However, the effects of various formulation parameters on its mechanical properties are not well understood. In this study, an array of GelMA scaffold fabrication parameters is evaluated by varying GelMA concentration and ultraviolet light exposure time. Our overarching goal is to characterize the mechanical properties through ultrasound and rheological measurements, providing a framework for GelMA scaffold selection. Pulse-echo ultrasound techniques were used to non-invasively determine the sound speed and attenuation of the scaffolds, revealing significant dependence on GelMA concentration. Steady shear rate and strain- and frequency-controlled oscillatory shear tests using a rotational rheometer (Model: DHR-2, TA Instruments) revealed a range in the levels of shear-thinning as well as viscoelasticity and showed moduli-dependence on both GelMA concentration and light exposure time. Together, this acoustic and rheological characterization can be used to inform the selection of GelMA scaffolds in tissue engineering applications.

2:30

3pPAb7. Accurate determination of elastic anisotropy to detect internal microstructures using resonant ultrasound spectroscopy. Boris Maiorov (National High Magnetic Field Lab., Los Alamos Nat. Lab, MS E536, Los Alamos, NM 87545, maiorov@lanl.gov), Tannor Munroe, William K. Peria, and Christopher A. Mizzi (National High Magnetic Field Lab., Los Alamos National Lab., Los Alamos, NM)

Elastic constants are of great significance, because they are fundamental thermodynamic susceptibilities that connect directly to thermodynamics and electronic structure, as well as to mechanical properties. Resonant ultrasound spectroscopy (RUS) measures non-destructively all fundamental elastic properties by determining the natural frequencies at which a 3D object resonates when is mechanically excited (normal modes). By using an inversion scheme, it is possible to extract the entire elastic tensor of a material with extreme sensitivity in a single frequency sweep using only one, small sample. This makes RUS particularly advantageous to study physical properties of low-symmetry single crystals and textured bulk materials. I will present recent RUS measurement examples of its applications to thermodynamic and materials science problems. I will focus on the effect of texture in elastic constants as a result of mechanical or growth processes and the robustness of results obtained by RUS. In materials such as aluminum that is often considered isotropic but shows clear anisotropy when it is extruded, or 3D printed. Similar but much larger effect is found in additive manufactured stainless steel where the growth process leads to a large anisotropy with a softening of 40%–50% along the build direction.

2:45

3pPAb8. Relaxing free boundary conditions in resonant ultrasound spectroscopy for extreme environments. Christopher A. Mizzi (MagLab, Los Alamos National Lab, MS E536, TA 35, Bldg. 127, Rm. C108, Los Alamos, NM 87545, mizzi@lanl.gov), Tannor Munroe (MagLab, Los Alamos National Lab, Los Alamos, NM), and Boris Maiorov (National High Magnetic Field Lab., Los Alamos National Lab., Los Alamos, NM)

Elastic constants are thermodynamic susceptibilities, which provide symmetry-resolved insight into phase transitions and material anisotropy. Therefore, their importance spans basic to applied science and engineering. Resonant ultrasound spectroscopy (RUS) is a powerful method to determine elastic constants, because RUS simultaneously measures all elastic constants in small, low symmetry samples with high precision and accuracy. Conventionally, elastic constants in RUS are obtained from resonant frequencies assuming the sample freely resonates. However, achieving free resonance is often incompatible with experimental situations, because such configurations are inherently unstable. This limitation has hindered utilizing RUS to study phases arising in extreme environments like strong magnetic fields. To overcome these obstacles, we introduce a variation of RUS in which small, finite-area contact is maintained between the sample and ultrasonic transducers using an adhesive. First, we establish the conditions under which this clamped point contact can be treated as a small perturbation to free resonance. After comparing predictions from this modelling with experiment, we then experimentally demonstrate the ability to extract elastic constants, and their temperature dependence, with comparable accuracy and precision to conventional RUS. Finally, we show we can reproducibly map phase boundaries in temperature and magnetic fields using our new approach.

3:00

3pPAb9. Elastic behavior of additively manufactured 316L stainless steel with changing laser power and speed. Mason Hayward (Univ. of Louisiana at Lafayette, 240 Hebrard Boulevard, Broussard Hall Rm. 103, Lafayette, LA 70503, mason.hayward1@louisiana.edu), Kevin Starlard (Univ. of Louisiana at Lafayette, Lafayette, LA), Erica Murray (Inst. for Micromanufacturing, Louisiana Tech Univ., Ruston, LA), and Gabriela Petculescu (Univ. of Louisiana at Lafayette, Lafayette, LA)

Ultrasonic investigations of additively manufactured 316L stainless steel were performed to reveal correlations between properties and printing parameters. The laser power and scanning speed of the selective-Laser-Melting printing process were varied systematically: 40–100 W in 20-W steps, and 600 mm/s to 1200 mm/s in 200-mm/s steps. Resonant ultrasound spectroscopy and pulse-echo measurements were performed. The extensive data allow for observations of specific behaviors in the elastic properties of the material as a function of VED, the volumetric energy density (VED $\propto$ power/speed). An increase in the values of c_{11} and c_{44} with increasing VED is confirmed, followed by saturation in the expected complete-fusion zone. The saturation values are comparable to those of the wrought ($\sim$250 GPa for c_{11} and $\sim$75 GPa for c_{44}). At low VED, c_{11} is quite small ($\sim$100 GPa). Furthermore, an interesting behavior appears in the incomplete-fusion zone. At a fixed VED, the laser power level affects the elastic properties. As power increases, the porosity decreases, the mass density increases, and both elastic constants increase. The details observed in the material's elastic properties are important for its correct usage in space and biomechanical applications.

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Mitchell L. Day, Chair
Dept. of Biological Sciences, Ohio University, Life Sciences Research Facility 241,
Athens, OH 45701

Chair's Introduction—1:30

Invited Paper

1:35

3pPP1. In search of the missing resolved harmonics: Neural and behavioral studies of pitch discrimination. Bertrand Delgutte (Dept. of Otolaryngol., Head and Neck Surgery, Harvard Med. School, Massachusetts Eye & Ear, 243 Charles St., Boston, MA 02114, bertrand_delgutte@meei.harvard.edu)

The pitch of harmonic complex tones (HCT) plays important roles in speech and music perception and in the perceptual organization of sound. Human listeners rely primarily on the spectral pattern of harmonics individually resolved by the frequency analysis in the cochlea to derive a pitch percept. Yet, evidence for a robust neural representation of resolved harmonics in animal models has been elusive, and some species appear to rely on temporal envelope cues rather than on resolved harmonics to discriminate pitch. We performed both behavioral experiments on pitch discrimination by rabbits and single-unit recordings from the auditory nerve (AN) and inferior colliculus (IC) using HCTs with equal-amplitude harmonics in which the fundamental frequency (F0) was varied over a wide range. We found that rabbits can discriminate the pitch of HCT using the spectral pattern of resolved harmonics for F0s within the range of conspecific vocalizations. Correspondingly, many IC neurons contribute to a rate-place code for the frequencies of resolved harmonics for F0s $>$500 Hz, and this code is more robust to variations in stimulus level than that found in the AN. A remaining challenge is to identify the neural mechanisms underlying enhanced rate-place coding in IC. [Work supported by NIH-NIDCD DC002258.]

Session 3pSA

Structural Acoustics and Vibration, Engineering Acoustics, Computational Acoustics, Noise, and Physical Acoustics: Real World Case Studies for Damping

Robert M. Koch, Cochair
US Navy, 304 White Horn Drive, Kingston, RI 02881

Benjamin M. Shafer, Cochair
PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406

Invited Papers

1:00

3pSA1. Damping in precision vibration control: A review. Vyacheslav M. Ryaboy (Photonics Solutions, MKS, 1791 Deere Ave. Irvine, CA 92606, vryaboy@newport.com)

This paper reviews various damping techniques used in precision vibration control devices such as optical tables and other vibration-isolated platforms for sensitive optical, acoustical, and life sciences experiments, as well as advanced technological processes [1]. These techniques include tunable dynamic vibration absorbers, active damping, and modal damping in isolated platforms, as well as active and passive damping in vibration isolators. The theoretical background, principles of implementation, and test results are illustrated by case studies that include a large tuned-damped optical bench in a petawatt laser facility, multi-beam lithography enabled by active damping, effective damping of vibration-isolated workstation for experiments on animal hearing, and atomic force microscopy assisted by active vibration isolation. [1] V.M. Ryaboy, Vibration Control for Optomechanical Systems, World Scientific, 2022.

1:20

3pSA2. Laser Doppler vibrometry-based measurements on viscoelastic panels for flexural damping properties. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu), Jack Taylor, and Max Miller (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Bending wave propagation is of central importance when enhancing sound transmission losses of sandwiched wall board systems incorporating viscoelastic panels as constrained damping layers. The damping properties of constrained damping layers made of viscoelastic materials can often be characterized by bending wave excitations. However, the multifactor, dispersive nature of the bending waves leads to challenges in reliable dynamic material characterization. To better understand the damping mechanism of the constrained damping layers, an experimental methodology employing complex bending wave theory has been developed to determine these flexural wave properties, including the loss factor and the bending stiffness of highly viscous panels. Relying on a transfer function from the experimentally measured bending velocities between two locations radially away from a flexural wave exciter on the viscoelastic panel under test, this methodology yields the broadband bending loss factor, the bending phase speed, and the bending stiffness. This paper discusses the experimental method for characterizing the above properties from laser Doppler vibrometry-based measurements of bending velocities. This paper also discusses the challenges with this method as well as an approach to mitigate the challenging effects and improve measurement accuracy.

1:40

3pSA3. The acoustics of shrieking perforated panels. Samuel K. Hord (MD Acoust., LLC, 170 S William Dillard Dr., Ste 103, Gilbert, AZ 85233, samuel@mdacoustics.com), Robert M. Pearson, Michael L. Dickerson (MD Acoust., LLC, Gilbert, AZ), and Benjamin M. Shafer (PABCO Gypsum, Tacoma, WA)

Under windy conditions, vortex shedding can cause perforated metal skins to emit undesirable noise. This phenomenon was observed, measured, and replicated. The acoustic spectrum was analyzed and compared to the modes of a vibrating plate. An extensional-damping layer was added to the plates for vibration isolation. The damping layer was applied as a high-loss painted coating to the surface of the plates. The acoustic spectrum was then analyzed again and compared to the spectrum without treatment. This study describes the process used to investigate and solve this noise control problem.

2:00

3pSA4. Modeling nonlinear acoustic damping due to flow separation over a baffle blade. Joseph Day (Univ. of Colorado Springs, 1420 Austin Bluffs Pkwy, Colorado Springs, CO 80918, jday@uccs.edu) and J. M. Quinlan (Univ. of Colorado Springs, Colorado Springs, CO)

Nonlinear acoustic damping has been observed in many high-amplitude acoustic systems as a result of flow separation and shear layer vortical motion, eventually transforming some of the acoustical energy into heat. The amount of nonlinear acoustic damping helps determine the nonlinear limit cycle amplitude, e.g., damping caused by baffle blades in a liquid rocket engine to reduce combustion instabilities. The damping mechanism is dependent on both the location and phase of flow separation. Flow separation is a function of both the boundary layer growth and the acoustically imposed pressure gradient. When the acoustic pressure gradient is adverse, the boundary layer is more prone to separation. Using this as a basis, a model can be created, applicable to general geometry, which can approximate nonlinear acoustic damping in flow over baffle blades. The constructed model will be compared to established cases, such as an orifice in a duct, to validate the model. Once validated, this model can approximate the nonlinear acoustic damping caused by a baffle blade, for both standing and traveling waves, and will be compared to experimental results to test accuracy. This model could result in designing rocket engines based on engine specific damping requirements rather than past successful designs.

Contributed Papers

2:20

3pSA5. Measuring and evaluating damping in 3D printed multi-material compliant joints. Heather L. Lai (Eng. Programs, SUNY New Paltz, 208 Eng. Innovation Hub, SUNY New Paltz, New Paltz, NY 12561, laih@newpaltz.edu) and Bailey Yewchuck (Mech. Eng., SUNY New Paltz, New Paltz, NY)

Fused deposition modeling (FDM) is a standard 3D printing technique used regularly in high performance and consumer level printers. FDM involves extruding heated thermoplastic filament in multiple layers. When two print heads with dissimilar filaments are used in coordination with one another, alternating print layers, multi-material structures can be designed to exhibit tuned dynamic behavior at the flexible / rigid multi-material interface. Joints tailored with specific stiffness and damping properties have applications in a wide range of fields, including vibration and acoustic isolation. While compliant, flexible materials used in multi-material FDM printing have intrinsic material damping associated with the mechanical energy dissipation within each individual material, additional structural damping resulting from poor material interfaces can also arise in these types of applications. Measurement of this damping can provide a non-destructive means of identifying poor interface joints. Appropriate measurement of both forms of damping are important for tuning compliant joints to the desired dynamic behavior as well as identifying flaws within the interfaces. Several different methods for modeling and measuring damping in multi-material compliant joints will be described, and comparison of experimentally measured damping will be presented for material damping alone, and material damping combined with structural damping resulting from different material interfaces.

2:35

3pSA6. Measured noise levels beneath type B flat metal roof assemblies during rainfall. Jon W. Mooney (Acoust. by JW Mooney, 418 11th St., De Witt, IA 52742, acoustics@jwmooney.com)

Flat metal roof assemblies are common in steel building design. Since they can produce high levels of rainfall generated noise, quiet roof assemblies are desirable over noise sensitive spaces. To determine the relative effects of design variations, a pair of fixtures were constructed to measure the noise levels beneath 2×4 ft sections of selected roof assemblies subjected to rainfall. Each test item was assembled from precut 2×4 ft sections of type B metal roof (with or without a concrete layer), cover boards, and insulation, membrane and ballast as required. The fixtures included a tipping bucket to measure rainfall rate. A microphone located inside each fixture and connected to a digital recorder, picked up ambient noise, rainfall generated roof assembly noise, and the tipping bucket signal. To calibrate the recordings, the sound pressure level of the tipping bucket signal was measured at the recording microphone before testing. Recordings are analyzed to calculate indicated octave band sound power levels versus rainfall rate for the various tested roof assemblies. Promising variations for further study are given.

Session 3pSP

Signal Processing in Acoustics, Biomedical Acoustics, Architectural Acoustics, Physical Acoustics, and Underwater Acoustics: Acoustical Imaging, Reconstruction, and Localization III: Biomedical Acoustics (Hybrid Session)

Yangfan Liu, Cochair
Purdue Univ., Ray W. Herrick Laboratories, Purdue University, 177, South Russell Street, West Lafayette, IN 47907-2099

Efren Fernandez Grande, Cochair
Technical University of Denmark, Lyngby, Denmark

Chair's Introduction—1:00

Contributed Paper

1:05

3pSP1. Compressed sensing using a trajectory-based model for ultrasonic tomography in a wooden medium with cylindrical symmetry. Yishi Lee (Metropolitan State Univ. of Denver, Eng. and Eng. Technol. Dept., 1449 7th St., Denver, CO 80204, Denver, CO 80204, ylee24@msudenver.edu)

Stress wave propagation induced by embedded waveguides in circular cross-sections produces radial AW1a, reflected mode (AW1b), and Rayleigh propagations (AW2). Prior studies in this area have developed optimal filtering techniques to extract the characteristics of different arrival components from the time-series waveform. This study exploits the information uncovered in the time-series signal to reduce the sensing requirements for stress wave tomographic inversion. This work employs the algebraic reconstruction technique (ART) and the trajectory estimation from the transient finite element model (FEM) to perform the tomographic inversion. The proposed methodology consists of the following steps: (1) Estimate the arrival trajectories using the time-dependent energy flux propagation vector from the existing FEM-based propagation model. (2) Formulate the pixel contribution matrix based on the estimated trajectories from step 1. (3) Based on a known priori, perform tomographic inversion based on the optimal relaxation and additive iterative techniques. This presentation will discuss the technique and the associated algorithm and present the results of the proposed reconstruction technique with numerical validations. Its broader impact will drastically reduce the traditional through-transmission tomography from 28 measurements around the circular cross-sectional region to 6, providing rapid data collection with comparable efficacy.

Invited Papers

1:20

3pSP2. Optimization of synthetic aperture delay-multiply-and-sum reconstruction. Philip M. Holmes (Mayo Clinic Graduate School of Biomedical Sci., 200 1st St SW, RO_OS_02_2008, Rochester, MN 55902, holmes.philip@mayo.edu), Hyungkyi Lee, and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Synthetic aperture (SA) acquisition allows for focusing at all points in ultrasound images during reconstruction. Both delay-and-sum (DAS) and delay-multiply-and-sum (DMAS) have been used to reconstruct SA images. An integral part of both algorithms is the aperture size used to define the beam shape with the f-number. Although f-number variation has been studied in DAS reconstruction, there has not been a systematic study on this topic for DMAS reconstruction in SA imaging. In this work, we evaluated the effects of f-number in DMAS reconstruction using an imaging phantom (CIRS 040GSE). We reconstructed DAS and DMAS images of hyperechoic, hypoechoic, and anechoic contrast targets, as well as point targets, while varying the transmit and receive f-numbers between 0 and 7 in 0.2 increments. Generalized contrast-to-noise ratio (gCNR) was measured for contrast targets. The full width at half maximum (FWHM) was evaluated using point targets. When imaging contrast targets, the DAS and DMAS algorithms responded very differently when varying the f-numbers. DMAS produced higher gCNR values with low transmit f-numbers and high receive f-numbers, while DAS produced higher gCNR values with high transmit and receive f-numbers. When evaluating FWHM, the optimal receive f-number was close to 1 for both algorithms.

1:40

3pSP3. Localization with deep learning networks for super-resolution ultrasound imaging. Katherine Brown (Bioengineering, Univ. of Texas at Dallas, 800 W. Campbell Rd., Richardson, TX 75080, katherine.brown@utdallas.edu) and Arthur D. Redfern (Comput. Sci., Univ. of Virginia, Dallas, TX)

OBJECTIVE: The objective was to optimize an acquisition methodology and deep learning network to localize a microbubble (MB) contrast agent on a preclinical ultrasound (US) platform with the goal of moving super-resolution ultrasound (SRUS) imaging towards clinical adoption. METHODS: A deep learning network was optimized based on the latest advances in computer vision, convolutional neural networks, and transformer architectures. Synthetic data were produced in an US simulation of tissue with MB at various concentrations flowing in a vascular model. The network was programmed in PyTorch and trained on 12 000 synthetic images. US data were collected with a Vevo 3100 (FUJIFILM VisualSonics Inc) using an MX201 linear transducer. *In vivo* testing images were obtained from hepatocellular carcinoma (HCC) rat liver tumors. After MB injection (Definity), 3000 frames were collected at 90 Hz. RESULTS: The SRUS *in vivo* results reveal a high level of detail. The smallest vessel measured 34 μm in diameter. Compared to conventional methods, the network improved performance by 10x on a CPU. A GPU platform could give an additional boost, by as much 100x. CONCLUSIONS: Deep networks for localization show potential for improving performance of SR-US towards a real-time imaging modality. The use of a pre-clinical focused US platform demonstrates that localization can improve the visualization of vascular detail and aid clinical understanding.

Contributed Paper

2:00

3pSP4. Rapid beamforming of ultrasound chirp signals in frequency domain using the chirp scaling algorithm. Louise L. Zhuang (Elec. Eng., Stanford Univ., 3155 Porter Dr., Stanford, CA 94304, llz@stanford.edu), Jeremy Dahl (Radiology, Stanford Univ., Palo Alto, CA), Howard Zebker (Elec. Eng., Stanford Univ., Stanford, CA), and Marko Jakovljevic (Radiology, Massachusetts General Hospital, Boston, MA)

Linear frequency-modulated (chirp) transmits have been used successfully in the past to increase penetration depth of ultrasound signals in tissue and to improve the signal-to-noise ratio (SNR) in the resulting ultrasound images. However, beamforming chirp signals using delay-and-sum (DAS) can be slow on systems without a GPU. We propose using the chirp scaling algorithm (CSA), originally developed for synthetic aperture radar, as a faster alternative to DAS on CPU that results in similar image quality, especially at larger depths. To perform preliminary comparisons of the beamforming methods, we simulated in FIELD II monostatic synthetic aperture data containing point targets up to 300 mm in depth. The simulation accounted for average signal attenuation in soft tissue of 0.5 dB per MHz per cm. We analyzed the point spread functions and the runtimes of the methods in MATLAB with a single CPU. Beamformed point targets above 80 mm depth had almost identical sidelobe levels and full-width-at-half-maximum between the two methods, while the median runtime for CSA was 9.3 times faster than for DAS. We further extend CSA to multistatic acquisitions with chirp transmits and with curvilinear arrays for more clinical applicability and improved image quality.

2:15–2:25 Break

Invited Paper

2:25

3pSP5. Detecting myocardial perfusion deficit with contrast-enhanced high frame rate ultrasound. Geraldi Wahyulaksana (Biomedical Eng., Erasmus MC, Wytemaweg 80, Rotterdam, Zuid Holland 3015CN, Netherlands, g.wahyulaksana@erasmusmc.nl), Luxi Wei (Erasmus MC, Rotterdam, Netherlands), Maaike Te Lin Hekkert, Daniel J. Bowen, Bernardo Raposo Loff Barreto (Erasmus MC, Rotterdam, Zuid Holland, Netherlands), Antonius F. van der Steen, and Hendrik Vos (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands)

Imaging the myocardial perfusion after a heart attack has a high prognostic value, because it can guide the clinicians for better treatment before, during, or after percutaneous coronary intervention (PCI). Contrast-enhanced ultrasound is a promising imaging modality because it can be performed quickly on bedside within or outside the Cathlab. However, the image quality may generally not be good enough for accurate assessment of the myocardial perfusion. We have developed a technique that uses higher-order singular value decomposition of high frame rate echocardiographic data, in combination with contrast-specific pulsing schemes, to detect flow and perfusion. To assess the sensitivity of our technique to detect the local interruption of myocardial perfusion, we performed *in vivo* measurements on a porcine model. The left anterior descending coronary artery was occluded by using a balloon catheter which mainly interrupted the flow into the apex of the heart. Data were acquired with an ultrasound research scanner and phased array, and processed offline. The results showed that we can localize the blood flow inside the myocardium and detect the interrupt when the vessel is occluded, which shows the first proof of concept.

2:45

3pSP6. Spatiotemporal image reconstruction for photoacoustic computed tomography of tumor vascular perfusion in murine models. Refik Mert Cam (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 1406 W Green St., Urbana, IL 61801, rcam2@illinois.edu), Seonyeong Park, Mark Anastasio (Bioengineering, Univ. of Illinois Urbana-Champaign, Urbana, IL), and Umberto Villa (Oden Inst. for Computational Eng. and Sci., The Univ. of Texas at Austin, Austin, TX)

Monitoring critical physiological processes in murine models like tumor vascular perfusion and its response to prospective anti-cancer treatments is a significant potential use case of dynamic photoacoustic computed tomography (PACT). Previously reported studies of dynamic PACT are based on a frame-by-frame image reconstruction (FBFIR) procedure in which full-view measurement data are assumed to be rapidly acquired. However, many commercial three-dimensional PACT imagers acquire measurements at each tomographic view rotating the object in discrete steps. The time to collect the full-view data is limited by the rotation speed of the object holder and the laser repetition rate. Therefore, FBFIR techniques are not applicable, and there is a critical need for accurate and efficient spatiotemporal image reconstruction (STIR) techniques that can account for spatiotemporal redundancies in the object's features. To address this, we propose a low-rank matrix estimation-based STIR technique in which the sought-after dynamic image is approximated using a semiseparable approximation in space and time. To validate the proposed method, we also develop a virtual imaging framework for PACT employing dynamic numerical mouse phantoms with a physiologically realistic respiratory motion and perfusion model. The studies demonstrated that the proposed method accurately recovers the tumor vascular perfusion and object motion.

3:00

3pSP7. Rapid refocusing of ultrasound images in frequency domain using range-Doppler algorithm. Marko Jakovljevic (Radiology, Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, mjakovljevic@mgh.harvard.edu), Louise L. Zhuang (Elec. Eng., Stanford Univ., Stanford, CA), and Anthony E. Samir (Radiology, Massachusetts General Hospital, Boston, MA)

The ability to maintain image uniformity and high resolution over large depths is important for several clinical applications of ultrasound, including deep abdominal imaging in patients with high BMI. One way to improve image quality in such cases is to use retrospective transmit focusing, which involves combining received data from different focused transmits to improve image resolution and SNR outside of transmit focal zone. Retrospective transmit is typically accomplished using the delay-and-sum (DAS) beamforming, which can be slow on systems without a GPU. As a faster alternative, we treat beamsummed signals from focused transmits as monostatic synthetic aperture data from virtual sources, and apply a frequency-domain beamformer such as RDA to rapidly refocus an RF image. We demonstrate the concept using FIELD II simulated ultrasound signals from a point target phantom over 200 mm depth. The RDA-refocused image shows a uniform point spread function, and the reduction in full-width at half maximum by a factor of 2.5 compared to the original image from focused transmits at f-number 10. RDA-based refocusing also achieves a speedup by a factor of 14 relative to the DAS-based retrospective beamformer.

3:15

3pSP8. Computationally efficient deep learning-based image reconstruction for sound speed estimation in ultrasound computed tomography. Gangwon Jeong (Bioengineering, Univ. of Illinois, 1406 W. Green St., Urbana, IL 61801, gangwon2@illinois.edu), Fu Li (Bioengineering, Univ. of Illinois, Urbana, IL), Umberto Villa (Computational Eng. and Sci., Oden Inst. at the Univ. of Texas at Austin, Austin, TX), and Mark Anastasio (Bioengineering, Univ. of Illinois, Urbana, IL)

Ultrasound computed tomography (USCT) has great potential for breast cancer imaging due to its ability to quantify acoustic tissue properties such as the speed of sound (SOS). Full-waveform inversion (FWI)-based image reconstruction methods can produce high spatial resolution SOS images, but these methods are computationally demanding, especially when formulated in the time-domain. Deep learning (DL)-based methods can be computationally efficient and provide high-quality object estimates as studied in various medical image reconstruction problems. However, there remains a further need to develop and assess the DL-based image reconstruction methods for USCT for use in medical imaging applications. Here, we present a supervised DL approach for USCT image reconstruction that is formulated as an image-to-image translation problem. Specifically, a low-resolution SOS image obtained via traveltime tomography and a high-resolution reflection tomography image are used as inputs to a U-Net, which is trained to estimate a high-resolution SOS map. These input images are complementary, providing both quantitative information about SOS values and information about tissue boundaries. Numerical studies that employ realistic numerical breast phantoms demonstrated that the proposed method produced high-quality SOS estimates based on image quality metrics such as NMSE, SSIM, and PSNR.

Session 3pUW

Underwater Acoustics: General Topics in Underwater Acoustics: Calibration and Laboratory Measurements

David R. Barclay, Chair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, B3H 4R2, Canada

Contributed Papers

1:30

3pUW1. Development and evaluation of primary calibration methods in underwater acoustics in a frequency range from 1 Hz to 100 Hz. Alper Biber (Underwater Acoust. Lab., Tubitak Mam, Gebze, Turkey) and Ata C. Corakci (Underwater Acoust. Lab., Tubitak Mam, Sualtı Akustik Laboratuvari, Kocaeli, Gebze 41470, Turkey, can.corakci@tubitak.gov.tr)

The project entitled "Metrology for Low Frequency Sound and Vibration" ("INFRA-AUV"), currently active within the European Metrology Programme for Innovation and Research (EMPIR). Measurement of seismic and infrasound activity is very important for monitoring natural extreme events. Low frequency sound and vibration monitoring technologies are well-established, but the sensors are often used for specific and local applications only and lack traceability. This project aims to establish both the first primary measurement standards for low frequency sound (in both air and water) and vibration. The aim of the task "Development and Evaluation of Primary Calibration Methods in Underwater Acoustics in a Frequency Range from 0.5 Hz to 100 Hz" is to develop and evaluate candidate calibration techniques in underwater acoustics for use as primary standards in frequency range of interest. These methods will be evaluated in terms of their limitations, uncertainty contributions and calibration results. The pressure coupler reciprocity method is based on the hydrophone, a transducer and projector which are calibrated inside a relatively small coupling chamber where the acoustic field is homogenous. Through a series of electrical measurements of voltage and current, the acoustic sensitivity of the three devices may be calculated traceable to electrical primary units.

1:45

3pUW2. Introducing sound speed variability in laboratory water tank. Alexandra M. Hopps (Phys. and Astronomy, Brigham Young Univ., Eyring Sci. Ctr., N-181, Provo, UT 84606, hoppsa@byu.edu), Tracianne B. Neilsen, and Jay M. Clift (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The variation of temperature in the ocean causes changes in sound propagation. To simulate the naturally occurring variability in an ocean environment, this project quantified the sound speed variation achievable in a laboratory water tank. The rectangular tank (1.2 m $\times$ 3.6 m with 0.5 m depth) has anechoic paneling that minimizes lateral reflections. In this experiment, temperature was measured from four different temperature sensors. The temperature was used to calculate the sound speed in the tank as a function of time while the water was cooled and heated. Sound speed values were compared from four different empirical equations. We found that while the temperature varies in time, the temperature is relatively uniform at different locations in the tank. In experiments involving three hours of heating and adding 100 pounds of ice, there was a 10-degree Celsius difference in temperature, corresponding to a 20 m/s difference in water sound speed. By adding an additional degree of variability to the tank measurements, a portion of the variability seen in the ocean can be replicated. This sound speed variability can then be used to test the robustness of machine learning algorithms.

2:00

3pUW3. Learning source-receiver range in a laboratory tank with variable water temperature. Corey E. Dobbs (Brigham Young Univ., N283 ESC, Provo, UT 84606, coreydobbs205@gmail.com), Alexandra M. Hopps (Phys., Brigham Young Univ., Provo, UT), and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Deep learning has great potential to be useful for ocean acoustics problems. However, uncertainty in an ocean environment must be taken into account, because it affects the ability of trained machine learning models to generalize. Because ocean measurements are costly, we are using a laboratory water tank to help develop our deep learning neural networks. In addition, using a water tank in a lab increases some level of control over the amount of uncertainty in data used to train neural networks. A residual convolutional neural network architecture is trained to identify source-receiver range in an underwater tank environment at room temperature. The model is then tested on data taken with variable water temperature. For comparison, two different source signals are used (sine waves and linear chirps) to analyze the usefulness of broadband sounds. These tests explore how environmental uncertainty impacts source localization and will lead to insights on how to develop robust machine learning for ocean acoustics problems. [Work supported by the Office of Naval Research.]

2:15

3pUW4. Hydrophone calibration between 1 kHz and 10 kHz using elastic waveguides. Robert Drinnan (Oceanogr., Dalhousie Univ., PO BOX 15000, Halifax, NS B3H4R2, Canada, robert.drinnan@dal.ca) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Modern techniques for hydrophone calibration from IEC 60565:2020 typically require sensors to be placed in a free field or within a hydrostatically varying chamber. At mid-frequencies (defined as 1 kHz–10 kHz), the wavelength is too long for free-field conditions in tanks available to most manufacturers and academics. A novel technique to calibrate hydrophones is investigated to address the measurement gap between very-low-frequency (0.1 Hz–1 kHz) and high-frequency (10 kHz–200 kHz) techniques. The measurement environment consists of a 12-meter length of copper tubing that is coiled into a 1-meter diameter helix. Propagation in the elastic waveguide decreases the speed of sound within the apparatus and the wavelength relative to its free-space equivalent. This provides a longer reverb-free time within which to make the calibration measurements using a small ($<$1 m^3) volume. To increase the gain of the system, the recorded files are matched filtered against normalized transmission replicas to determine the signal energy at the receiver. The propagation within the waveguide is studied, including investigating the modal dispersion and channel gain. Calibrations are performed on multiple Ocean Sonics icListen hydrophones, including independently calibrated reference units, using an uncalibrated source to determine the accuracy and precision of the system.

2:30

3pUW5. Using Bayesian optimization to refine laboratory tank sound propagation modeling. Scott P. Hollingsworth (Phys. and Astronomy, Brigham Young Univ., BYU Dept. of Phys. and Astronomy, N283 ESC, Provo, UT 84602, sh747@byu.edu) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Because ocean measurements are costly, a laboratory water tank provides opportunities to take scaled measurements and develop approaches to machine learning that are transferable to ocean acoustic problems. These applications are better achieved if reflections off the side walls are significantly reduced. To reduce these reflections, the side walls of the tank are lined with Aptile SF5048 echo reducing tiles made by Precision Acoustics. This work studies the extent to which two ocean sound propagation models (namely ORCA and BELLHOP) can predict relative transmission loss as a function of source-receiver range and frequency. The modeling parameters of sound speed and density that determine the reflections at the bottom of the tank were refined through a Bayesian optimization algorithm. We found that the optimized parameters improve the model-data agreement and that the models fit the data at certain frequencies corresponding mainly to peaks in the echo reduction for the Aptile panels. This agreement implies that at certain frequencies, sound propagation in our tank acts as it would in a large body of water. Using our tank as a testbed for developing machine learning algorithms is a reasonable approach.

2:45

3pUW6. Turbulence induced self-noise on cylinders in low velocity flow. Roger Oba (Acoust. Div., US Naval Reseach Lab., 4555 Overlook Ave. S.W., Washington, DC 20375, roger.oba@nrl.navy.mil) and Ravi Ramamurti (Labs. for Computational Phys. and Fluid Dynam., US Naval Res. Lab., Washington, DC)

Nominally stationary acoustic sensors, e.g., buoy suspended or bottom anchored, are subject to various environmental flows. These flows in turn generate turbulence along the surface of acoustic enclosures that give rise to self-noise. A continuous-flow water tunnel replicates at-sea flows for evaluation of acoustic enclosures for such self-noise. In preparation for the tunnel experiments, direct numerical simulations of hydrodynamic flow around the acoustic sensor housing have been performed, and show turbulence along the housing surface. Time-space analysis of pressure variation along the surface of the housing shows significant acoustic-like surface interaction at low frequencies, even at very low mean flow velocity. The broad frequency content of the turbulence is due to intermittently generated, irregular, coherent vortex structures that separate from the surface of the housing and advect downstream. These structures drive rapid surface pressure transients that propagate within the housing to the acoustic sensors as noise. The potential for noise reduction using foil shaped housing is explored. This research is supported by 6.2 NRL base program sponsored by the Office of Naval Research. Distribution Statement A: Approved for public release. Distribution unlimited.

Plenary Session and Awards Ceremony

Peggy B. Nelson
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

John Galvin III – For research into speech and music perception with cochlear implants

Kevin Haworth – For contributions to the development of passive cavitation imaging and therapeutic ultrasound methods

Tracianne Neilsen – For applications of machine learning to inverse problems in ocean acoustics

Bogdan Popa – For contributions to active acoustic metamaterials

Sarah Verhulst – For contributions to computational modeling of the normal and impaired auditory system

Robert White – For advancement in the field of acoustic micro-electro-mechanical systems

Introduction of Prize Recipients

2022 Rossing Prize in Acoustics Education to Kathleen E. Wage

2023 Hartmann Prize in Auditory Neuroscience to Bertrand Delgutte

2023 Medwin Prize in Acoustical Oceanography to David C. Barclay

Presentation of Awards

R. Bruce Lindsay Award to Julianna C. Simon

Helmholtz-Rayleigh Interdisciplinary Silver Medal in Biomedical Acoustics and Physical

Acoustics to Vera A. Khokhlova

Gold Medal to Mark F. Hamilton

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. All meetings will begin at 7:30 p.m., except for Computational Acoustics (4:30 p.m.), Engineering Acoustics (4:45 p.m.), and Signal Processing in Acoustics (5:30 p.m.).

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 discussion.

Committees meeting on Tuesday

Engineering Acoustics (4:45)	Northwestern/Ohio State
Signal Processing in Acoustics (5:30)	Purdue/Wisconsin
Acoustical Oceanography	Michigan/Michigan State
Animal Bioacoustics	Chicago F/G
Architectural Acoustics	Chicago A/B
Physical Acoustics	Los Angeles/Miami/Scottsdale
Psychological and Physiological Acoustics	Indiana/Iowa Acoustics

Committees meeting on Wednesday

Biomedical Acoustics	Armitage
Structural Acoustics and Vibration	Chicago H

Committees meeting on Thursday

Computational Acoustics (4:30)	Lincolnshire 1/2
Musical Acoustics	Lincolnshire 1/2
Noise	Chicago C
Speech Communication	Los Angeles/Miami/Scottsdale
Underwater Acoustics	Michigan/Michigan State

R. BRUCE LINDSAY AWARD

Julianna C. Simon

2023

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

ENCOMIUM FOR JULIANNA C. SIMON

…for contributions to the understanding of ultrasound-induced mechanical bioeffects and their clinical applications

10 MAY 2023 • CHICAGO, ILLINOIS

Julianna C. Simon grew up in Leavenworth, Washington. In May 2009, she graduated *Summa Cum Laude* in Bioengineering with a minor in Mathematics from Washington State University, where her freshman Physics professor never stopped recruiting her to Physics. Her doctorate in Bioengineering, December 2013, is from the University of Washington (UW), where she was awarded the Donald W. and Joan P. Baker Endowed Scholarship and the Dean of Engineering Fellowship.

Julianna formed an early and strong passion for space medicine. As an undergraduate in the National Aeronautics and Space Administration (NASA) Lewis Educational and Research Collaborative Internship Program in 2008, Julianna won the Outstanding Student Mentor Award for mentoring a high-school student participant. Her team won the Outstanding Team Award for work on functional near-infrared spectroscopy to improve pilot cognition. As a graduate student, she worked on NASA research developing medical ultrasound for space use. Julianna garnered a 3-year NASA postdoctoral grant entitled "Improving kidney stone detection in space flight analogs" and frequently presented at the U.S. Capitol and NASA facilities. As a testament to Julianna's expertise and respect within NASA, Julianna served on the 2017 joint commission of the National Academies of Science, Medicine, and Engineering to review NASA astronaut health risks and mitigations. After her fellowship, no NASA jobs were available, so she turned to academia.

In January 2017, Pennsylvania State University (PSU) enthusiastically welcomed Dr. Simon as Assistant Professor in the Graduate Program in Acoustics with a courtesy appointment in PSU's Department of Biomedical Engineering. At PSU, Julianna quickly established an experimental ultrasound laboratory called BASiL for Biomedical Acoustics Simon Lab and currently supervises about a half dozen graduate students. She has obtained substantial funding from a variety of sources for diverse projects: a National Science Foundation early CAREER Award, National Institutes of Health (NIH) Trailblazer grant on her first try, a Congressionally Directed Medical Research Program Discovery Award, and an NIH R01 award, which on average are first awarded to researchers 10 years older than Julianna. Likewise, she has published with her students in 20 different journals and given over 100 meeting presentations in her first 5 years at PSU.

Julianna's work at PSU also has included initiating a biomedical acoustics technical affinity group within the Center for Acoustics and Vibration and collaborating with many departments. She has completely reorganized the biomedical acoustics course at PSU to rave reviews and established a new course in research communication, focusing on writing academic papers and grant proposals. Julianna also led the redesign and implementation of the Acoustics Program's scholarship and research integrity training, required of all graduate students. Additionally, Julianna has continued to mentor younger students through outreach as she did at NASA and the UW, where she was always the first to volunteer to teach at the UW Discovery days and the Pacific Science Center Discovery Corps. Julianna recently received PSU funding to develop at-home PAWkits (Penn State Acoustic Wave kits) to give students hands-on experience regarding the fundamentals of acoustics.

Dr. Simon's love of medicine stems from a love of animals with acoustics being a way to solve medical problems. Julianna is an equestrian who had a pony at age 4 and beat 17-year-olds at age 10, trains daily, and competes at the highest levels. Although at first blush Julianna as a student seemed quiet and demure, she showed a great passion for medical research and was not shy in getting involved in every aspect of preclinical safety research and took the lead on these studies. Some referred to student Julianna affectionately in this role as "The Supreme Overlord" because even as a student she directed all aspects of all preclinical studies not just her own. Her work led to the UW's first university-sponsored Investigational Device Exception and human trial. This opened the door to others, and her team won the UW Distinguished Staff Award in 2014 for the accomplishment. This expertise led ASA and the Institute of Electrical and Electronics Engineers to invite Julianna to present tutorials on preclinical testing of therapeutic ultrasound. At the same time, her dissertation work and research with her PSU students created intellectual property useful in other clinical and commercial applications. In recognition of the translational aspects of her work, Julianna received the UW INVENTS award in 2012. At the same time, Julianna has made fundamental discoveries finding liver tissue flows and atomizes like liquid in acoustic fountains, and cavitation nuclei can be seen to be naturally present in the human body. In the latter work, she showed ubiquitous ultrasound imaging could detect these nuclei by placing human volunteers in a hyperbaric chamber to suppress the nuclei and the signal.

Julianna has been active in the ASA continuously since her early student days. She is a highly engaged member of the Biomedical Acoustics, Physical Acoustics, and Education Committees. She has chaired several ASA special sessions, organized and judged student poster competitions, authored an Acoustics Today article, published in society journals, and served on over half a dozen committees. Julianna currently chairs ASA's administrative Committee on Archives and History. As with her research and teaching, Julianna adds substantially to every environment with her intelligence, critical thinking, and pleasant personality. She has a quiet confidence and comfort with herself, and although she is many times the smartest person in the room, she is always happy just to be contributing to the team in whatever way necessary. She has been an excellent collaborator, a passionate independent researcher and leader, and an excellent writer sparing her co-authors much editing. Julianna is one of the nicest and most gracious and genuine human beings we have ever met. It has been a pleasure, Juli.

On behalf of those who have had the privilege of working with Julianna, we welcome this opportunity to celebrate her many accomplishments, and we look forward to her continued growth and future contributions to the field.

MICHAEL R. BAILEY
VICTOR W. SPARROW
YAK-NAM WANG
OLEG A. SAPOZHNIKOV

Helmholtz-Rayleigh Interdisciplinary Silver Medal
in
Biomedical Acoustics and Physical Acoustics

ASA
ACOUSTICAL SOCIETY
OF AMERICA

Vera A. Khokhlova

2023

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
David E. Weston	1998
Jens P. Blauert	1999
Lawrence A. Crum	2000
William M. Hartmann	2001
Arthur B. Baggeroer	2002
David Lubman	2004
Gilles A. Daigle	2005
Mathias Fink	2006
Edwin L. Carstensen	2007
James V. Candy	2008
Ronald A. Roy	2010
James E. Barger	2011
Timothy J. Leighton	2013
Mark F. Hamilton	2014
Henry Cox	2015
Armen Sarvazyan	2016
Blake S. Wilson	2017
Kenneth S. Suslick	2018
Barbara G. Shinn-Cunningham	2019
Michael R. Moldover	2021
George L. Augspurger	2022

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 VERA A KHOKHLOVA

…for contributions to the application of nonlinear acoustics to medical ultrasound

10 MAY 2023 • CHICAGO, ILLINOIS

Vera Aleksandrovna Khokhlova *née* Kovrigina, was born and raised in Moscow, USSR (now Russia). Her parents, Aleksandr Kovrigin and Elina Andrikanis, were physicists, both graduated from the Physics Faculty of the Lomonosov Moscow State University (MSU), which later became the Alma mater for Vera as well. Vera's father strongly influenced Vera's choice of a scientific career. Aleksandr Kovrigin was a member of the scientific group of Rem Khokhlov, a prominent Soviet scientist (Vice President of the Russian Academy of Sciences, Rector of Moscow State University, one of the pioneers of nonlinear optics), who was in the 1960s transforming MSU into a leading international research center for both nonlinear optics and nonlinear acoustics. Vera's father became one of the key experimentalists of the Khokhlov's group and pioneered many discoveries in the field of nonlinear optics.

Besides science, Vera's father was very active in many outdoor sports, and one of his passions was downhill skiing – again, a passion that he transferred to his daughters – Ekaterina and Vera. Vera was especially successful. During her high school years, she even was qualified as a Candidate Master of Sports – the third official level rank in sports in the USSR. As a student at MSU, she became a member of the university downhill skiing club team and participated in numerous competitions. In one of the years at the university, she won the MSU championship. This passion continues to this day as Vera keeps her childhood tradition of ski outings with her family and for members of her lab every winter.

Vera's aptitude and fondness for physics and math became apparent quite early; in the 6th grade of her middle school she transferred from her neighborhood school to the famous "*Moscow Physico-Mathematical School No. 2*." Many of the school graduates afterwards became well-known scientists and tech entrepreneurs. After graduation with the Gold Medal in 1979, Vera entered the Physics Faculty of MSU. There she met her future husband – Dmitry (Mitya) Khokhlov (son of Academician Rem Khokhlov); they were married in 1982. As a student, Vera joined the group headed by another member of Rem Khokhlov's team – Oleg Rudenko. The group was specializing in nonlinear acoustics, which became the field of Vera's scientific activities throughout her life. Oleg Sapozhnikov was in the same student group as Vera since the first year in MSU, and they joined Oleg Rudenko's group simultaneously. This was a start of their lifelong collaboration that included the formation of the Laboratory for Industrial and Medical Ultrasound (LIMU) at MSU.

Vera received the MSc degree from MSU in 1986 and defended her PhD thesis in 1991, which was devoted to statistical properties of diffracting and discontinuous acoustic waves of high intensity. Since then, she has been a faculty member in the Department of Acoustics, Physics Faculty of MSU. In 2012, Vera defended the Doctor of Science thesis, summarizing her studies on shock wave propagation and effects in inhomogeneous media in application to the problems of medical and atmospheric acoustics.

Vera started collaborations with her colleagues from the USA in 1990s. Her first visit was to Austin with Mark Hamilton and David Blackstock in the Mechanical Engineering Department at the University of Texas at Austin in 1993. On a later trip to the US, she stopped off in Seattle to visit Larry Crum's group: Center for Industrial and Medical Ultrasound (CIMU) in the Applied Physics Laboratory (APL) of the University of Washington (UW). This was the start of a collaboration with CIMU colleagues, which has continued for nearly 25 years. Shortly after Vera joined the APL/UW team, Oleg Sapozhnikov followed her to Seattle and they have become an essential part of the CIMU group. Together they write joint proposals with CIMU staff, perform experiments, supervise graduate students, and often bring their students from Moscow to participate in research projects. Vera is delighted to share the podium with Gold Medal recipient Mark Hamilton, and with Julianna Simon, recipient of the Lindsay Award, and one of her graduate students at CIMU, for whom she served as co-advisor.

Vera has become recognized as the leading international expert on nonlinear sound beams used in biomedical applications and in atmospheric acoustics. Her strength is her ability to combine understanding of basics physics with expertise on numerical modeling to tackle important applications of acoustic beams containing shocks. Vera is especially recognized by her studies on High Intensity Focused Ultrasound (HIFU) therapy – one of the most important practical applications of nonlinear acoustics. In particular, she is renowned as an inventor of boiling histotripsy, for which she has numerous publications and several patents (most already licensed by commercial firms).

Vera has been an active member of the ASA for many years. She is an ASA Fellow, was elected to the ASA Executive Council, and is currently the head of the ASA Commission on International Research and Education (CIRE). Vera's ASA service is in addition to her international service that includes two terms on the Board of the International Society for Therapeutic Ultrasound and Board member of the Russian Acoustical Society.

Vera and Dmitry have two daughters, Tatiana (born in 1982) and Maria (born in 1988); both of whom also became physicists. Following the paths of their grandparents and parents, they graduated from the Physics Faculty of MSU and defended their PhD theses there. Now Tatiana is a Research Associate Professor at the University of Washington; Maria is a co-founder of the international tech startup company "*Traceair*." Vera relishes her research visits to Seattle as she can also visit her grandchildren, Peter (11), (little) Tanya (8), and Max (3).

The field of therapeutic ultrasound has become a promising major advance in medicine, with over 150 medical indications being studied, over 100 device manufacturers, and hundreds of treatment centers for patients. Vera Khokhlova has played an important role in the evolution of this field and is a much deserving recipient of the Helmholtz-Rayleigh Interdisciplinary Silver Medal of the Acoustical Society of America.

LAWRENCE A. CRUM
OLEG A. SAPOZHNIKOV
MARK F. HAMILTON

Gold Medal

Mark F. Hamilton
2023

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	Kenneth N. Stevens	1995
Floyd A. Firestone	1955	Ira Dyer	1996
Harvey Fletcher	1957	K. Uno Ingard	1997
Edward C. Wente	1959	Floyd Dunn	1998
Georg von Békésy	1961	Henning E. von Gierke	1999
R. Bruce Lindsay	1963	Murray Strasberg	2000
Hallowell Davis	1965	Herman Medwin	2001
Vern O. Knudsen	1967	Robert E. Apfel	2002
Frederick V. Hunt	1969	Tony F. W. Embleton	2002
Warren P. Mason	1971	Richard H. Lyon	2003
Philip M. Morse	1973	Chester M. McKinney	2004
Leo L. Beranek	1975	Allan D. Pierce	2005
Raymond W. B. Stephens	1977	James E. West	2006
Richard H. Bolt	1979	Katherine S. Harris	2007
Harry F. Olson	1981	Patricia K. Kuhl	2008
Isadore Rudnick	1982	Thomas D. Rossing	2009
Martin Greenspan	1983	Jiri Tichy	2010
Robert T. Beyer	1984	Eric E. Ungar	2011
Laurence Batchelder	1985	William A. Kuperman	2012
James L. Flanagan	1986	Lawrence A. Crum	2013
Cyril M. Harris	1987	Brian C. J. Moore	2014
Arthur H. Benade	1988	Gerhard M. Sessler	2015
Richard K. Cook	1988	Whitlow W. L. Au	2016
Lothar W. Cremer	1989	William M. Hartmann	2017
Eugen J. Skudrzyk	1990	William A. Yost	2018
Manfred R. Schroeder	1991	William J. Cavanaugh	2019
Ira J. Hirsh	1992	Judy R. Dubno	2020
David T. Blackstock	1993	James F. Lynch	2021
David M. Green	1994	Michael J. Buckingham	2022

ENCOMIUM FOR MARK F. HAMILTON

…for contributions to theoretical nonlinear acoustics, education, and for service to and leadership of the society.

10 MAY 2023 • CHICAGO, ILLINOIS

Mark Francis Hamilton was born in Holyoke, Massachusetts. He enrolled in Columbia University in 1974 to study electrical engineering. His senior-year professor, Cyril Harris, pointed Mark toward acoustics, and, with Cyril's encouragement, Mark entered the graduate acoustics program at Penn State University in June 1978. Experimental research at the Applied Research Laboratory's Garfield Thomas Water Tunnel on cavitation bubble noise led to a master's degree. However, Mark found his calling when he took Frank Fenlon's Nonlinear Acoustics course and in 1980 began research on dispersion effects in parametric arrays under Frank for his doctoral topic.

Sadly, Frank died of cancer in June 1981, midway through Mark's research, but at the 1981 Ottawa ASA meeting Mark met David Blackstock from the University of Texas. Lacking suitable faculty at Penn State to supervise Mark, arrangements were made for David Blackstock to serve as Mark's PhD supervisor. Mark moved to the Applied Research Laboratories, University of Texas at Austin (ARL:UT), to complete his research there but remained a student at Penn State, where he defended his dissertation in June 1983. That encounter with David at an ASA meeting however set in motion a decades long partnership that led to UT Austin becoming one of the global leaders in nonlinear acoustics research.

After completing his doctorate Mark was awarded ASA's F. V. Hunt Postdoctoral Fellowship in 1983. He spent the fellowship year with Sigve and Jacqueline Naze Tjøtta at the University of Bergen in Norway with a focus on finite-amplitude sound beams. Mark returned to Texas, first as a research fellow at ARL:UT and then in 1985 he began his teaching career as Assistant Professor in the Department of Mechanical Engineering at UT Austin. Mark is a passionate educator who teaches with energy in a way that engages his students. He has taught many undergraduate and graduate acoustics courses. Mark has supervised or co-supervised 26 PhD students, with another 8 on course. He participated as an instructor at the biennial Physical Acoustics Summer School four times, a course that provides graduate students from around world an intensive week of acoustics education from world class teachers. In addition to exemplary teaching, Mark built a community for the UT students captured by the regular trip to the Posse pub on Friday afternoons.

Mark's academic contribution to the field of acoustics has resulted in more than 100 peer-reviewed publications, the majority of which have been published in the *Journal of the Acoustical Society of America (JASA)*. His contributions have been predominantly in fundamental nonlinear acoustics in air, water, tissue, and solids but also covered topics such as bubble dynamics, metamaterials, thermoacoustics, radiation force, noise underwater produced by offshore wind turbine farms, and mechanical waves in the walls of neurons. His early work focused on finite-amplitude beams that are commonly described using the KZK equation, which captures the physics of nonlinearity, absorption, and diffraction. The equation had been initially developed in the Soviet Union in the 1960s but was challenging to solve in general. In 1995 he published a paper in JASA with his PhD student Yang-Sub Lee, in which they describe a numerical method for solving the KZK equation in the time domain, which was much more computationally efficient than the frequency domain solutions of the time. The paper has garnered more than 500 citations and the on-line version of the code (KZKTexas code) has had thousands of downloads.

In the late 1980's during a visit to the Soviet Union, Mark met with Evgenia (Zhenia) Zabolotskaya—the "Z" in KZK—and Yura Ilinski. After several exchange visits, Zhenia and Yura joined Mark's research group permanently in 1991. They developed and explored a completely new field of nonlinear acoustics: nonlinear shear and surface waves, including Rayleigh, Stoneley and Scholte waves, which have important implications in applications such as surface acoustic waves widely used for filtering of RF signals in mobile devices and landmine detection. The collaboration also resulted in work in the area of bubble physics, in which his research group went beyond the classical problem of a single bubble in an infinite ideal liquid and analyzed multiple interacting bubbles in the presence of surfaces and when in visco-elastic media. The results had implications in applications of bubble physics and biomedical ultrasound in general, e.g., bubbles in blood vessels, bubbles in biological tissue, and bubble clouds produced by lithotripsy in the kidney.

Mark's work has made him one of the pre-eminent experts in the world on nonlinear acoustics. The phrase "he wrote the book" is often thrown around figuratively but in the case of Mark it is literally true. The book in question is "Nonlinear Acoustics," which is edited by Mark and David Blackstock and has become the go-to reference for anybody involved in nonlinear acoustics be they a fresh student or a seasoned researcher.

The Society has recognized Mark throughout his career as a Hunt Fellow (1983), recipient of the R. Bruce Lindsay Award (1989), election to Fellowship, and the Helmholtz-Rayleigh Interdisciplinary Silver Medal awarded in 2014. Mark has also been deeply engaged in the good running and governance of the Society including sitting on 13 committees, acting as chair for five, serving as the vice-president in 2003/4 and the president in 2008/9. He served as chair of the ASA Education Committee (1988-91) and the faculty advisor for the ASA student chapter in Austin since 2011. He has been an Associate Editor of *JASA* for Nonlinear Acoustics, Physical Acoustics and Ultrasonics, and *Acoustics Research Letters On-line*.

Outside of the Society, Mark has also contributed to other professional organizations. For example, he has been involved in multiple committees with the American Institute of Physics, served as Chair of an International Symposium on Nonlinear Acoustics (ISNA), and as secretary of the international body that organizes the ISNA meetings for nearly 15 years. He was the USA representative to the International Commission for Acoustics (ICA) (2013-19; President of the ICA (2019-2022) and currently serves as past-president. Mark has been named Chair of the International Congress on Acoustics to be held in the US in 2025.

In sum, Mark Hamilton has made important contributions to the science of nonlinear acoustics. He has been dedicated to educating and supporting the next generation of acousticians, and he has provided outstanding and passionate service to the running of the Society and many international organizations. The field of acoustics and the Acoustical Society are both so much richer because of Mark's contributions. He is a deserved recipient of the Gold Medal and we send him our warmest congratulations.

ROBIN O. CLEVELAND
LAWRENCE A. CRUM

Session 4aAA

Architectural Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, Physical Acoustics, Noise, and Computational Acoustics: Artificial Intelligence and Machine Learning

Semiha Yilmazer, Cochair
Faculty of Art, Design and Architecture, Department of Interior Architecture and Environmental Design, Bilkent University, Ankara 06800, Turkey

Dick Botteldooren, Cochair
Information Technology, Ghent University, Technologiepark 126, Gent 9052, Belgium

Andrew Mitchell, Cochair
Institute for Environmental Design and Engineering, University College London (UCL), Central House, 14 Upper Woburn, London WC1H 0NN, United Kingdom

Chair's Introduction—8:20

Invited Papers

8:25

4aAA1. Using audio content and emotional response to predict soundscape perception through machine learning. Volkan Acun (Interior Architecture and Environ. Design, Bilkent Univ., Faculty of Fine Arts Design and Architecture, FF Bldg., Rm. 205A, Ankara 06800, Turkey, volkan.acun@bilkent.edu.tr) and Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

This study is concerned with utilizing machine learning techniques for predicting soundscape perception by identifying the audio content of soundscapes and linking it with people's reported emotional responses. This research goal required developing an environmental sound classification model; however, the capabilities of these algorithms have some significant drawbacks. Supervised learning algorithms need a large number of labelled audio samples for each sound category. Given that a model for classifying environmental sound must be trained using a wide range of sound sources, this presents a substantial problem for developing a robust model that generalizes well to different environments. We prepared a convolutional neural network (CNN) based classifier; however, to tackle the limitations, we used musical instruments for the training dataset rather than environmental sound sources and optimized the neural network for this task. Based on how closely the soundscapes' audio content resembled the musical instruments in the dataset, CNN classified the soundscapes' audio content. We then conducted an online soundscape perception survey to evaluate participants' emotional responses to numerous soundscape clips. We prepared a feedforward neural network, which used the sound classification model's audio content output with the survey data to create a model for predicting people's responses to different soundscapes.

8:50

4aAA2. Fast noise mapping: A machine learning approach for predicting traffic noise indicators. Dick Botteldooren (Information Technol., Ghent Univ., Technologiepark 126, Gent 9052, Belgium, dick.botteldooren@ugent.be)

Noise mapping has become a popular tool for assessing the impact of traffic noise on human health and wellbeing. Usually, models for estimating exposure of a population rely on a (standardized) simplification of the noise emission and propagation. Mapping is simplified by considering only the equivalent noise level since it removes all temporal information from the source prediction, the propagation, but unfortunately also from the impact assessment. Yet, it is known that noise events can play an important role in, e.g., sleep disturbance and temporal information are essential in the evaluation of outdoor soundscapes. Models for calculating spectro-temporal levels with a one-second resolution have been proposed, but these quickly become too slow for large scale exposure assessment. Here, a convolutional neural network (CNN) based on environmental characteristics is proposed as a fast alternative. The CNN-model is trained on thousands of detailed physical simulations with varying propagation conditions and traffic intensities on the surrounding roads. To increase the understanding of this machine learning model, Shapley values are used. They show the importance of different features in calculating indicators, e.g., loud events are mainly explained by close-by traffic, L50 by high intensity roads, also at larger distances.

9:15

4aAA3. Deep learning techniques for noise annoyance detection: Results from an intensive workshop at the Alan Turing Institute. Andrew Mitchell (Inst. for Environ. Design and Eng., Univ. College London (UCL), Central House, 14 Upper Woburn, London WC1H 0NN, United Kingdom, andrew.mitchell.18@ucl.ac.uk), Emmeline Brown (Ctr. for Adv. Biomedical Imaging, Univ. College London (UCL), London, United Kingdom), Ratneel Deo (Univ. of Sydney, Sydney, New South Wales, Australia), Yuanbo Hou (Faculty of Eng. and Architecture, Ghent Univ., Ghent, Belgium), Jasper Kirton-Wingate (COG-MHEAR, Edinburgh Napier Univ., London, United Kingdom), Jinhua Liang (School of Electron. Eng. and Comput. Sci., Queen Mary Univ. of London, London, United Kingdom), Alisa Sheinkman (School of Mathematics, Univ. of Edinburgh, Edinburgh, United Kingdom), Christopher Soelistyo (Inst. for the Phys. of Living Systems, Univ. College London (UCL), London, United Kingdom), Hari Sood (The Alan Turing Inst., London, United Kingdom), Arin Wongprommoon (School of Informatics, Univ. of Edinburgh, Edinburgh, United Kingdom), Kaiyue Xing (School of Education, Commun., and Lang. Sci., Newcastle Univ., Newcastle, United Kingdom), Wingyan Yip (Soldo, Ltd., London, United Kingdom), and Francesco Aletta (Inst. for Environ. Design and Eng., Univ. College London (UCL), London, United Kingdom)

Advancements in AI and ML have enabled us to combine automated sound source recognition and deep learning models for predicting subjective soundscape perception. We held a multidisciplinary, cross-institutional Data Study Group (DSG) to investigate how sound source information could be incorporated into deep learning models for predicting urban noise annoyance. We used a large-scale dataset of 2980 15-s recordings paired with 12 210 annoyance ratings (from 1 to 10) and sound source labels. A total of 14 neural networks and 4 conventional ML models were built. The best model was trained to simultaneously predict sound source labels and annoyance rating. It achieved an RMSE = 1.07 for annoyance prediction and AUROC = 0.88 for label classification, while a similarly structured model trained to predict annoyance ratings only (i.e., no sound source information) achieved RMSE = 1.13. Results showed that including sound source labels as a simultaneous training output, rather than as an explicit model input resulted in the best performance. Overall, these models performed very well at predicting both annoyance ratings and identifying sources, providing a starting bed for automated annoyance detection systems. This presentation will provide context for the DSG and present conclusions drawn regarding approaches to applying deep learning techniques to noise annoyance detection.

9:40

4aAA4. A vector classifier for sound similarity classification based on audio features. Cengiz Yilmazer (CSY R&D and Architecture Eng., National Nanotechnology and Material Inst., Ankara 06800, Turkey, censem6@gmail.com) and Semiha Yilmazer (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

This study introduces a vector classifier for Sound Similarity Classification. Sound classification using the known features can be accepted as the prime definition within a limited time. However, the sound is a function not only of spectral composition but also a change of this composition in time Then, it should be characterized with flexible classifier vectors as a function of time segment. The study considers (1) steady-state time segmentation of sound with respect to feature deviations and creating time-marked classifier vectors within the segment made of the know features and (2) using long-time combined segments and histogram base classifiers within the classifier vector. The main subject of the work is the reduction of memory and computation time effectiveness and similarity detection capability.

10:05–10:20 Break

10:20

4aAA5. A retrospective on monitoring noise pollution with machine learning in the Sounds of New York City project. Mark Cartwright (Dept. of Informatics, New Jersey Inst. of Technol., GITC, Rm. 3902E, University Heights, Newark, NJ 07102, mark.cartwright@njit.edu), Charles Mydlarz (Ctr. for Urban Sci. and Progress, New York Univ., Brooklyn, NY), and Juan P. Bello (Dept. of Music and Performing Arts Professions, New York Univ., New York, NY)

The Sounds of New York City (SONYC) project (2016–2022) was a project to monitor and mitigate urban noise pollution using a smart acoustic sensor network, citizen scientists, and collaboration with city agencies. During its lifetime, the project deployed 75 fixed-location sensors and collected over 150×10^6 10-s audio recordings. A key element of the project was the development of machine listening models to detect the sources of noise pollution rather than just the overall noise level. In this talk, we first discuss our initial approach to data collection and machine listening including sensor development and deployment, citizen-science data annotation, self-supervised audio representation learning, and downstream sound-event detection model training. We then discuss analysis results using the outputs of these models, followed by the challenges and limitations of our initial approach. Finally, we discuss our solutions to overcome those challenges, such as citizen-deployed sensors, source-specific loudness estimation, and few-shot sound-event detection.

10:45

4aAA6. Using machine learning to reconstruct room geometry from an impulse response. Alaa Algargoosh (Media Lab, Massachusetts Inst. of Technol. (MIT), 75 Amherst St., Cambridge, MA 02139, alaas@mit.edu) and Nikhil Singh (Media Lab, Massachusetts Inst. of Technol. (MIT), Cambridge, MA)

Machine learning in room acoustics is an emerging field with great potential yet to be explored. Previous research includes predicting acoustic properties such as the impulse response from visual features that include an image of a room. However, one application that can significantly transform the architectural acoustics design process is reconstructing the room geometry based on its acoustic properties. This paper explores applying machine learning in predicting room geometry from an impulse response. The research aims to provide a tool for architects to design a room based on the desired acoustic experience.

11:00

4aAA7. Random forest regression to predict design performance of concert halls. Jonathan Broyles (Architectural Eng., The Penn State Univ., 510 Toftrees Ave., Apt. # 331, State College, PA 16803, jmb1134@psu.edu) and Zane T. Rusk (The Penn State Univ., University Park, PA)

Over the last couple decades, machine learning (ML) and advanced computational strategies have been successful when employed in building physics problems. ML models have known applications in the operations of buildings, including real-time energy monitoring and automated mechanical systems. They can also be used to inform design decisions for complex problems during conceptualization. For acousticians and building designers, trained ML models could provide better insight on expected performance without requiring extensive calculations or room acoustic simulations. This presentation describes a computational method for predicting the design performance (i.e., mid-frequency reverberation time) of various concert halls. More specifically, a random forest (RF) regression model is trained on architectural and measured acoustical data to predict the reverberation time. The RF models are tuned and act as a surrogate model for predicting the mid-frequency reverberation time of concert halls that were not used in model training. The estimated reverberation times are then compared to calculations from conventional equations (i.e., Sabine and Norris-Eyring equations) and room acoustic simulations using ray-tracing methods. This work shows how acousticians and architects can employ ML strategies in the conceptual design phase to improve estimated performance accuracy without expending significant computational resources.

11:15–11:45
Panel Discussion

Session 4aBAa

Biomedical Acoustics, Engineering Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Advances in Elastography I

Thomas Royston, Cochair
University of Illinois at Chicago, 851 S. Morgan Street, Rm. 218 SEO (MC 063), Chicago, IL 60607

Matthew W. Urban, Cochair
Department of Radiology, Mayo Clinic, Rochester, MN 55905

Chair's Introduction—8:00

Invited Papers

8:05

4aBAa1. Quantitative biomechanics using reverberant optical coherence elastography. Kirill Larin (Univ. Of Houston, 4800 Calhoun Rd., 3605 Cullen Blvd, Rm. 2028, Houston, TX 77204, klarin@uh.edu)

In this presentation, I will introduce and discuss recent developments in our effort for quantitative biomechanics of tissues and organs using a multifocal acoustic radiation force source (that combines an ultrasound transducer and a 3D-printed acoustic lens) in reverberant optical coherence elastography (Rev-OCE). An array of plano-concave acoustic lenses, each with an 11.8 mm aperture diameter, were used to spatially distribute the acoustic energy generated by a 1 MHz planar ultrasound transducer, producing multiple focal spots on a target plane. These focal spots generate reverberant shear wave fields detected by the Optical Coherence Tomography (OCT) system. The effectiveness of the multifocal Rev-OCE system in probing mechanical properties with high resolution is demonstrated in layered gelatin phantoms, tissues *ex vivo*, and whole mouse embryos *in vivo*.

8:25

4aBAa2. Quantification of elastic anisotropy of human skin *in vivo* with dynamic optical coherence elastography and polarization-sensitive OCT. Ivan Pelivanov (Bioengineering, Univ. of Washington, 616 NE Northlake Pl, Benjamin Hall Bld, Rm. 363, Seattle, WA 98105, ivanp3@uw.edu), Mitchell Kirby, Peijun Tang, Maju Kuriakose, Gabriel Regnault, Matthew O'Donnell, Ruikang K. Wang (Bioengineering, Univ. of Washington, Seattle, WA), Russell Ettinger (Burn and Plastic Surgery Clinics at Harborview, Univ. of Washington, Seattle, WA), and Tam Pham (Regional Burn Ctr. at Harborview, Univ. of Washington, Seattle, WA)

The complex structure of skin ensures a broad range of biological functions. The loss of a large skin area due to trauma or disease usually requires reconstructive surgery that often involves grafting or flapping. The clinical need for non-contact objective measurements of skin elasticity is necessary to guide grafting procedures to minimize complications such as secondary contracture or hypertrophic scars. Here, we report a non-contact and non-invasive method combining acoustic micro-tapping optical coherence elastography (AμT-OCE) with polarization-sensitive (PS-) optical coherence tomography (OCT) to quantify anisotropic elastic properties of skin. We show that all three elastic constants that define skin's anisotropic elastic deformation and the orientation of collagen fibers in the dermis can be determined from propagating elastic waves over the skin surface. Measurements were performed on healthy volunteers *in vivo*. A nearly incompressible transverse isotropic model of skin elasticity was used to reconstruct the moduli from experimental data after being validated in extensive numerical simulations. Finally, we demonstrate that combining several OCT modalities (structural OCT, OCT angiography, PS-OCT, and AμT-OCE) may provide rich information about skin and demonstrate the potential for complex characterization of scar.

8:45

4aBAa3. Shear wave elastography based on noise correlation and time reversal: From 1d to 3d shear elasticity imaging. Javier Brum (Laboratorio de Acústica Ultrasonora, Instituto de Física, Facultad de Ciencias, Universidad de la República, Igua 4225, Montevideo 11400, Uruguay, jbrum@fisica.edu.uy), Miguel Bernal (Verasonics, Inc., Kirkland, WA), Carolina Rabin, Carlos Negreira, and Nicolas Benech (Laboratorio de Acústica Ultrasonora, Instituto de Física, Facultad de Ciencias, Universidad de la República, Montevideo, Uruguay)

Shear wave elastography (SWE) relies on the generation and tracking of coherent shear waves to image soft tissue's shear elasticity. However, coherent shear wave tracking is not always possible due to scattered or interfering waves that arise from inhomogeneities, muscular activity, heart beating, or external sources. To overcome this limitation, we developed an alternative approach using a complex elastic wave-field. Based on the analogy between time reversal and seismic noise correlation, this complex field is "transformed" into a coherent converging-diverging time-reversal field using spatial-temporal cross-correlations [1]. Using the computed time reversal field,

there are different ways to image the shear elasticity: tracking the coherent shear wave as it focuses, measuring the focus size (Rayleigh criteria) or evaluating the vibration amplitude at the focus [2]. One advantage of this approach is its compatibility with low imaging rates, which led to innovative applications in SWE. Thus, the goal of this talk is to review the major developments in wave-physics for 1D and 2D elasticity imaging using noise correlation of shear waves and to present its latest applications involving passive elastography and 3D elasticity imaging using row-column arrays. [1] Catheline, *Phys. Rev. Lett.* (2008); Brum, *JASA* (2008); Benech, IEEE-TUFFC (2009). [2] Brum, *Front. Phys.* (2021).

9:05

4aBAa4. Transient elastography with focused shear wave beams. John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Medical Ctr., Pittsburgh, PA 15261, jmc345@pitt.edu), Yu-Hsuan Chao (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Branch T. Archer (Chandra Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Kang Kim (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Transient elastography (TE) is a clinically available ultrasound elastography technique in which mechanical vibration of a small piston at the tissue surface generates a short shear wave pulse that propagates into the body. The piston doubles as an ultrasonic transducer; thus, the shear wave propagation is measured with pulse-echo ultrasound along the beam axis, yielding the wave speed and tissue stiffness. The long shear wavelength compared to piston diameter used in current devices results in low shear wave signal that decays rapidly on axis as the shear wave spreads in all directions, thus limiting the signal-to-noise ratio and imaging depth in practice. We present an overview of our new technology of focused shear wave beams, which aims to overcome these limitations by concentrating shear wave energy in TE near to the beam axis with vibration of a concave circular piston. Analytical modeling of focused shear wave generation and propagation is validated against measurements in gelatin phantoms. Results indicate that focused shear wave beams have relatively larger amplitudes and penetration depths compared to unfocused shear wave beams. Ongoing modeling and experimental efforts towards the measurement of elastic anisotropy, such as of skeletal muscle, using focused shear wave beams will be discussed.

9:25

4aBAa5. Shear wave elastography: From dispersion matching to full waveform inversion. Murthy Guddati (NC State Univ., 2501 Stinson Dr., NCSU-Civil Eng., Raleigh, NC 27695-7908, mnguddat@ncsu.edu), Tuhin Roy, Abdelrahman M. Elmeliegy (Civil Eng., North Carolina State Univ., Raleigh, NC), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Shear Wave Elastography (SWE) involves estimating mechanical properties through inversion, i.e., matching measured and simulated propagation characteristics of shear waves in the tissue. The accuracy of the estimated properties depends significantly on the specific characteristics/responses that are being matched. These could range from simple group velocity to dispersion curves and to full-wave response (particle velocity measurements). Using specific applications of arterial, liver, and tumor elstography, we illustrate that effective SWE is performed by resorting to an inversion approach, or combination of inversion approaches, guided by the underlying physics. To this end, we present inversion approaches ranging from matching dispersion characteristics to matching full waveform responses and provide rationale for choosing the appropriate technique(s) depending on the problem at hand.

Contributed Papers

9:45

4aBAa6. Broadband pulses in shear wave elastography. Thomas L. Szabo (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

Usually pulses used for shear wave elastography and ARFI are considered to be monochromatic; however, there are advantages to using broadband pulses and including the absorption characteristics of the tissues. A new broadband formulation for volume acoustic radiation forces is presented that accounts for the pulse shape and the changes it encounters propagating through tissues with frequency power law absorption properties. This equation reduces to the well know formulation for the continuous wave case derived by W. Nyborg. Ideal tone bursts employed in elastography have the same time average intensity tissue as continuous wave intensities, irrespective of length. Tone bursts in viscous media have turn-on and turn-off transients called "precursors" and "postcursors," which can increase pulse amplitudes at the ends of the pulses [Szabo, *JASA* **45**(1): 124–130 (1969)]. By accounting for absorption across a broader bandwidth pulse spectrum, these transient effects can be reduced. Their effects and the potential for pulse shaping are discussed.

10:00

4aBAa7. A Scholte wave based ultrasound elastography method for imaging superficial tissue. Abdullah A. Masud (Dept. of Mech. Eng., Texas Tech Univ., Box 41021, Lubbock, TX 79409-1021, Abdullah-Al.Masud@ttu.edu) and Jingfei Liu (Mech. Eng., Texas Tech Univ., Lubbock, TX)

Pathological changes in tissues are often related to changes in tissue mechanical properties, making elastography an important tool for medical applications. Among the existing elastography methods, ultrasound elastography is of great interest due to the inherent advantages of ultrasound imaging technology, such as low cost, portability, safety, and wide availability. However, the current ultrasound elastography methods, including shear wave elastography, can readily image deep tissue but cannot assess superficial tissue. To address this challenge, we proposed an ultrasonic Scholte-wave-based approach for imaging the elasticity of superficial tissue. The feasibility of the proposed technique was tested using a gelatin phantom with a cylindrical inclusion. In the tests, Scholte (surface) and shear (bulk) waves were simultaneously generated by the same excitation but propagated in the superficial and deeper regions of the phantom, respectively. In this

study, we first demonstrated that the elasticity of superficial tissue could be evaluated by utilizing the generated Scholte wave alone and further showed that a comprehensive elasticity imaging of the tissue extending from the superficial to deep regions can be achieved by combining the proposed Scholte wave technique and the conventional shear wave technique.

10:15–10:30 Break

10:30

4aBAa8. Biaxial prestress and waveguide effects on estimates of the complex shear modulus using optical elastography in a transverse isotropic cornea phantom. Marta Dore (Univ. of Illinois at Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, mdore4@uic.edu), Aime Luna, Michael Sun, and Thomas Royston (Univ. of Illinois at Chicago, Chicago, IL)

Tensile prestress is inherent to the functional role of some biological tissues currently being studied using elastography, such as skeletal and cardiac muscle, arterial walls, and the cornea. Therefore, the impact of prestress coupled with waveguide effects due to small dimensions in one or more directions needs to be better understood. An experimental configuration is designed, fabricated, and experimentally tested using optical elastography. Thin layered isotropic and transversely isotropic phantoms are statically stretched biaxially in plane while simultaneously conducting optical elastography measurements of out of plane motion. Guided by analytical models and numerical finite element simulations, experimental measurements are post-processed to obtain an estimate of the complex (viscoelastic) shear modulus as a function of prestress level and frequency of vibratory motion.

10:45

4aBAa9. A simulation framework for pulse wave and vector flow imaging using fluid–structure interaction and FIELD-II simulations. Pengcheng Liang (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, pl2810@columbia.edu), Nima Mobadersany (Biomedical Eng., Columbia Univ., Washington, DC), and Elisa Konofagou (Biomedical Eng., Columbia Univ., New York, NY)

Pulse Wave Imaging (PWI) utilizes ultrasound elasticity imaging and normalized cross-correlation (NCC) tracking pulse-induced arterial wall distension propagations to perform localized pulse wave velocity and assess arterial compliance mapping for the early detection of vulnerable carotid plaques. At the same time, Vector Flow Imaging (VFI) using a singular value decomposition filter and NCC with axially and laterally shifted axial kernels provides blood flow information. In this study, fluid–structure interaction (FSI) simulations (FEBio) were integrated with FIELD-II ultrasound imaging simulations to optimize both PWI and VFI for simultaneous estimation of compliance and flow in structures of variable vessel geometry and plaque stiffness at distinct ultrasound parameters. The spatiotemporal maps of wall distension rate between FSI ground truth and PWI method were in excellent agreement ($r^2 = 0.968$–0.982) at plane wave angles compounding at ($-1°$, 0, $1°$) and NCC window size of 7 wavelengths in both straight and 87% stenotic vessel geometries. Similar agreement was found in 2D blood flow velocity estimation between FSI ground truth and VFI with a maximum correlation of 0.9464 at the systolic upstroke and r^2 of 0.7426 at the angles of ($-2°$, 0, $2°$) and NCC window size of 60 wavelengths. Overall, the framework presented herein allowed PWI and VFI technique validation and optimization prior to *in vivo* application.

11:00

4aBAa10. Improving group velocity based estimates of arterial stiffness. Charles Capron (Biomedical Eng. and Physiol., Mayo Clinic, 200 1st St. SW, Rochester, MN 55902, Capron.Charles@mayo.edu), Tuhin Roy, Murthy Guddati (Civil Eng., North Carolina State Univ., Raleigh, NC), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Arterial stiffness is a predictor of cardiovascular diseases, the leading causes of death worldwide. Ultrasound-based methods that measure arterial wave propagation have promise for evaluating local stiffness *in vivo*. However, geometric properties of arteries cause dispersion, invalidating typical assumptions underlying the relationship between shear modulus G and group velocity c_g, which clinical implementations of ultrasound shear wave elastography do not consider. Here, we examine the dependence of these estimates on geometry and evaluate alternative approaches. Wave motion in the proximal wall of an artery after application of focused acoustic radiation force is simulated with a semi-analytical finite element (SAFE) model, and c_g is estimated using a time-to-peak algorithm to determine G with several methods. First, the value of G a clinical scanner would report is calculated assuming a bulk medium. Second, a pulse wave velocity (PWV)-based G estimate is calculated by taking $c_g = \mathrm{PWV}$ and applying the Moens–Korteweg equation. Third, we develop an interpolation-based method to provide a corrected G estimate using data generated by the SAFE model. Simulation results show severe geometry-dependent bias with the bulk method, which is partially ameliorated with the PWV method and substantially improved with the interpolation approach. Results are validated using arterial phantoms.

11:15

4aBAa11. Crosstalk analysis of comb detection for measuring shear wave propagation. Hyungkyi Lee (Radiology, Mayo, Rochester, MN, Lee.Hyungkyi@mayo.edu), Philip M. Holmes (Mayo Clinic Graduate School of Biomedical Sci., Rochester, MN), James Greenleaf, and Matthew W. Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Applications, such as shear wave elastography and Doppler imaging, require high frame rates. Plane wave compounding (PWC) is widely used for these applications. Comb detection was proposed to combine the high frame rate of PWC and the high signal quality of focused beam scanning. Comb detection transmits multiple focused beams simultaneously to increase the frame rate. These comb beams are scanned laterally to cover the whole region-of-interest. The simultaneous multiple transmission of focused beams causes crosstalk between neighboring beams. Transmit crosstalk is related to the pressure field of transmit beams, and receive crosstalk is determined by the beam profile of receive beamforming. In this study, we varied f-number (F/N) and apodization window and measured their effects on crosstalk based on the pressure field and an arterial wall simulation using Field II. For transmit design, transmit apodization with a Hamming window significantly reduced crosstalk compared to a rectangular window and transmit F/N had little impact on the crosstalk. Regarding receive beamforming, a Hamming window led to lower crosstalk than the rectangular window and F/N from 1 to 5 suppressed crosstalk to below 30 dB. The effects of crosstalk in shear wave motion data from phantom experiments were also analyzed. Comb detection showed less motion artifact than PWC in phantom experiments.

11:30

4aBAa12. A thermally polarized ^{129}Xenon phantom for MR elastography studies in a ultra-high field MRI system. Irene Canavesi (Biomedical Eng., Univ. of Illinois Chicago, 516 N May St., Chicago, IL 60642, icanav2@uic.edu), Weiguo Li, Dieter Klatt, and Thomas Royston (Biomedical Eng., Univ. of Illinois Chicago, Chicago, IL)

Being second only to cardiovascular problems, respiratory diseases account for five of the 30 most common causes of death worldwide. One way to address this problem is by improving early diagnostic methods. Due to the complex heterogeneous structure of the lung, conventional imaging and elastography techniques based on ultrasound and magnetic resonance (MR) imaging have been limited in their ability to provide regional detail of morphology and mechanical properties, both of which can be altered by respiratory diseases. Hyperpolarized ^{129}Xe(non) MR imaging has shown promise as a means of regionally quantifying ventilation, microstructure, and gas exchange efficiency in the lung parenchymae. MR elastography of the lungs using hyperpolarized gases is also being explored. To advance development, a stable phantom for MR imaging and elastography calibration using ^{129}Xe is needed. The design, fabrication, and experimental testing of a stable thermally polarized ^{129}Xe MR imaging and elastography phantom for a 9.4 T preclinical MR imaging system are presented.

11:45

4aBAa13. Soft tissue characterization by dynamic impact analysis: Results on agar-based phantoms. Anne-Sophie Poudrel, Arthur Bouffandeau, Giuseppe Rosi, Vu-Hieu Nguyen (Multiscale Modeling and Simulation Lab., CNRS, Créteil, France), and Guillaume Haiat (Multiscale Modeling and Simulation Lab., CNRS, Laboratoire MSMS, Faculté des Sci., UPEC, 61 Ave. du gal de Gaulle, Creteil 94010, France, guillaume.haiat@univ-paris-est.fr)

The mechanical characterization of soft tissues is of great importance in different medical fields since a change of soft tissue properties may be of physio-pathologic nature. This study investigates the feasibility of using impact analysis to measure the mechanical properties of soft tissue mimicking phantoms made of agarose. The impact analysis is performed on agar-based specimens with various concentrations by applying small impacts on a beam in contact with the surface of the specimen. The time variation of the signal corresponding to the impact force is analyzed as a function of the agar concentration and the impact force amplitude. A temporal indicator τ is developed, corresponding to the time difference between the initial impact and the rebound of the beam on the hammer. The results show a decrease of τ when the agar concentration increases. The method appears to be sensitive to non-linear phenomena related to the material behavior and/or to contact phenomena between the beam and the specimen. The error on the estimation of the percentage of agar is of the same order of magnitude than the sensitivity obtained using elastography techniques. This study opens new perspective for fast, cheap, and non-invasive mechanical characterization of non-linear soft tissues.

THURSDAY MORNING, 11 MAY 2023 — ARMITAGE, 8:30 A.M. TO 11:10 A.M.

Session 4aBAb

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Making and Using Cavitation Images for Therapeutic Ultrasound I

Meaghan O'Reilly, Cochair
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Michael Gray, Cochair
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Kevin J. Haworth, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

Chair's Introduction—8:30

Invited Papers

8:35

4aBAb1. Desperate times call for disparate measures: Towards harmonization of cavitation data across environments and systems. Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7DQ, United Kingdom, michael.gray@eng.ox.ac.uk) and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom)

As ultrasonic techniques rapidly evolve to provide a host of cavitation-mediated therapies, it is fair to ask: how can we reconcile the data obtained across patients and hardware systems? This talk will describe progress made towards achieving system- and patient-independent cavitation localization and dosimetry. Methods and results will be presented for the characterization, quantification, and correction of biasing factors in the propagation path and receiver system. The talk will conclude with a discussion of limitations and ongoing efforts to demonstrate the full suite of proposed techniques.

8:55

4aBAb2. Time-resolved passive cavitation mapping using the transient angular spectrum approach. Yun Jing (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu), Mucong Li (Thomas Lord Dept of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Juanjuan Gu (North Carolina State Univ., Raleigh, NC), Pei Zhong (Thomas Lord Dept of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), and Junjie Yao (Duke Univ., Durham, NC)

Passive cavitation mapping (PCM), which generates images using bubble acoustic emission signals, has been increasingly used for monitoring and guiding focused ultrasound surgery. This study investigates a transient angular spectrum (AS) approach for PCM. The working principle of this approach is to backpropagate the received signal to the domain of interest and reconstruct the spatial–temporal wavefield encoded with the bubble location and collapse time. The transient AS approach is validated using an in silico model, water bath, and *in vivo* experiments. It is found that the transient AS approach yields similar results to delay and sum, but is considerably faster. The results obtained by this study suggest that the transient AS approach is promising for fast and accurate PCM.

Contributed Papers

9:15

4aBAb3. Investigating the change in point spread function and resolution of random apodization passive cavitation images. Weston P. Gaskins (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, 231 Albert Sabin Way, Cardiovascular Ctr., 3950, Cincinnati, OH 45267, gaskinwp@mail.uc.edu) and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Passive cavitation imaging is an important technique in measuring cavitation dynamics, localizing concomitant bioeffects, and guiding various ultrasound therapies. The canonical algorithm for passive cavitation imaging operates via a delay, sum, and integrate method. With standard diagnostic arrays, this method yields poor axial resolution. We hypothesized that using a random apodization, minimum projection compounding technique could overcome this limitation. The random apodization approach randomly selected only half of the elements to be included in the delay, sum, and integrate algorithm to form a PCI. This process was repeated 30 times using the same data set but a different random subset of half the elements to create 30 images. The final image was obtained via a minimum intensity projection across the 30 images. The improvement of image quality can be tracked by comparing the point spread function (PSF), and the distance at which two cavitation sources can still be resolved, relative to when using the standard algorithm. The PSF reduced by $86\% \pm 9\%$ when the random apodization was applied. Lateral and axial resolution exhibited qualitative changes associated with a lower PSF, but no quantitative change in resolution, based on the Rayleigh criteria, was observed.

9:30

4aBAb4. Contrast-specific imaging of histotripsy: Chirp-coded subharmonic imaging combined with Volterra Filtering. Vishwas Trivedi (Discipline of Elec. Eng., Indian Inst. of Technol. Gandhinagar, MUSE Lab, Gandhinagar 382355, Gujarat, India, vishwas.t@iitgn.ac.in), Emily Wallach, Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, Chicago, IL), and Himanshu Shekhar (Discipline of Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India)

Histotripsy is a non-thermal focused ultrasound therapy under development to ablate tissue mechanically via bubble cloud activity. Real-time ultrasound imaging is used for treatment guidance. Bubble cloud hyperechogenicity is reduced in deep abdominal targets, making contrast-specific imaging an active area of research. Subharmonic imaging with chirp-coded excitation can improve bubble cloud detection, though the contrast-to-tissue ratio (CTR) was still limited to nearly 6 dB. Nonlinear components of a signal can be delineated with Volterra filtering, including those associated with bubble oscillations. In this study, we tested Volterra filtering as a means to enhance bubble cloud detection. Histotripsy bubble clouds were generated in scattering tissue-mimicking phantoms. Imaging of the bubble clouds was performed with a chirped pulse (1.9-μs duration, 7–12 MHz bandwidth). The scattered signals were processed with a subharmonic matched filter, followed by a tuned second-order Volterra filter. Volterra filtering improved the CTR for bubble cloud detection two-fold relative to matched filtering alone (12.5 dB vs 5.6 dB). Further improvement in bubble contrast was observed for third-order Volterra filtering (CTR of 20.3 dB) but at the cost of underestimating the bubble cloud area. Overall, these findings indicate the utility of Volterra filtering as a means to improve histotripsy image guidance.

9:45–10:15 Break

Invited Papers

10:15

4aBAb5. PAM, not spam: Towards quantitative, reproducible, and energy-preserving cavitation imaging. Cameron Smith, Luca Bau, Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, 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)

Since the development of the original passive acoustic mapping (PAM) algorithm for cavitation imaging in 2008, which employed a conventional beamformer adapted from seismology (time exposure acoustics), sequential algorithmic improvements have sought to enhance spatial resolution, reduce computation time and enable increasingly setup-independent reporting of cavitation activity. These enhancements have included the use of frequency-domain rather than time-domain implementations to improve computational efficiency, the introduction of the Robust Capon Beamformer to reduce artefacts, sub-aperture processing to further increase frame rate and the implementation of corrections for frequency-dependent attenuation and for array sensitivity and diffraction to account for probe- and depth-related variability. Once these corrections are taken into account, current PAM algorithms produce accurate estimates of cavitation source strength at a single location, but not in the surrounding region, leading to inaccurate estimates of the total cavitation energy

over the treatment volume. We present an energy-preserving and computationally efficient PAM algorithm based on Lucy-Richardson deconvolution, that enables the calculation of a set-up independent Cavitational Radiated Energy Density (CRED) that ensures cavitation images accurately reflect the total energy of radiated acoustic emissions within the imaging domain.

10:35

4aBAb6. Experimental demonstration of 3D passive cavitation imaging using adaptive beamforming. Barbara Nicolas (CNRS, Univ. of Lyon, Batiment Léonard de Vinci 21 Ave. Jean Capelle, Villeurbanne 69621, France, barbara.nicolas@creatis.insa-lyon.fr), François Varray (Creatis, Université de Lyon, Villeurbanne, France), Jean-Christophe Bera, Audrey sivadon, and Bruno Gilles (UCBL, Université de Lyon, Lyon, France)

Interest in passive cavitation imaging has increased in recent years with the development of cavitation-based treatments. Using cross-spectral matrix formalism, adaptive beamformers, such as Capon beamforming, MUSIC, and p-DAS, have been adapted to 2D passive cavitation imaging (PAM) (Polichetti *et al.*, 2020). More recently, we adapted this formalism in 3D to enhance the resolution of PAM using a commercial imaging array. Three beamformers have been extended in both 3D and the Frequency Domain: the DAS beamformer, the Robust Capon Beamformer (RCB) and the MidWay (MW) beamformer. A random sparse array configuration of a multiplexed commercial 2D array is used to reduce the number of channels used for the acquisition and to reconstruct 3D-PAM without degrading the quality of the reconstructed images. Such an approach allows for achieving a versatile well-resolved PAM using conventional research systems. In an experimental situation, the cavitation is initiated by a HIFU transducer at the tip of a needle and monitored with a high-speed camera confirming the presence of cavitation. This initial work allows us to envisage a possible future for 3D cavitation imaging using adaptive algorithms and sparse probes.

Contributed Paper

10:55

4aBAb7. Comparison of passive beamformers for isolating cavitation activity originating in the spinal canal. Andrew P. Frizado (Medical Biophys., Univ. of Toronto, 230 King St. East, Toronto, ON M5A 1K5, Canada, andrew.frizado@mail.utoronto.ca)

Focused ultrasound application to the spinal cord, through the intact spine, is confounded by the presence of substantial prefocal cavitation, in the posterior soft tissue and musculature. When constructing cavitation images to localize and control intracanal cavitation, the prefocal cavitation zones dwarf the signals from the canal, destructing the localization ability of the imaging inside the sensitive cord tissue—even when conventional phase and amplitude corrections are applied. In this study, multiple beamforming formulations are investigated to mitigate the influence of prefocal cavitation on transvertebral cavitation imaging, while preserving the localization ability in the canal. The performance of a standard delay-and-sum-integrate beamformer (DAS) is compared to delay-multiply-and-sum-integrate beamformer (DMAS), and the DMAS algorithm is tested with a paired multiplicative compounding method (pDMAS) that leverages the dual aperture approach required for transvertebral focused ultrasound sonications. The focal point spread function of the beamformers in passive cavitation images ($t_{int} = 7\ \mu s$) were $7.9 \times 2.2 \times 1.3$ mm (DAS), $6.9 \times 1.7 \times 1.1$ (DMAS), and $6.5 \times 2.9 \times 1.7$ mm (pDMAS) at 800 kHz. These beamforming approaches are evaluated for localization ability and clinical feasibility through cavitation imaging of prefocal and focal sources with *in silico* models and *ex vivo* human vertebrae.

4a THU. AM

Session 4aCA

Computational Acoustics, Structural Acoustics and Vibration, Musical Acoustics, Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, Biomedical Acoustics, and Physical Acoustics: Showcases of HPC Across Acoustics I

Kuangcheng Wu, Cochair
Naval Surface Warfare Center - Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Ralph T. Muehleisen, Cochair
Energy Systems, Argonne National Laboratory, 9700 S. Cass Ave., Bldg 362, Lemont, IL 60439-4801

Chair's Introduction—9:00

Invited Papers

9:05

4aCA1. High-performance computing in phononic materials with applications to nonlinearity and fluid–structure interaction. Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801, kmatlack@illinois.edu)

Phononic materials contain periodic and resonant substructures that enable novel acoustic and elastic wave behaviors, such as band gaps, topologically protected wave propagation, and negative refraction. Phononic materials preferentially take advantage of complex structures and intricate geometries, and even richer dynamics can occur in phononic materials that are nonlinear, multi-physics, or multi-phase. Certain phononic materials are often represented with reduced-order models such as lumped masses and springs, where the underlying physics can be revealed and systematically probed. However, high fidelity and/or performance simulations are critical to, e.g., interpret experimental validations, particularly with additively manufactured samples, analyze realistic structures and loadings, and for systematic optimizations. This talk presents two examples where high fidelity and high-performance computing enables the analysis of new behaviors of phononic materials and structures: (1) strongly nonlinear phononic materials that exploit history-dependent phenomena and effects of a continuum and (2) interactions between phononic materials and a surrounding fluid flow, where coupled nonlinear effects such as fluid–structure interaction are critical. Finally, opportunities for how high-performance computing that can advance phononic materials are discussed.

9:25

4aCA2. Sound generation in the flapping wing flight of insects. John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Holmes 302, 2540 Dole St., Honolulu, HI 96822, alleniii@hawaii.edu) and Kevin O'Rourke (Adaptive Res., Las Vegas, NV)

Flapping wing flight has been topic of recent interest with respect to the maneuverability and agility of insects. Computational fluid dynamics methods have been used to investigate the aerodynamics though typically for incompressible flow. The underlying sound generation mechanism, though of fundamental biological and physical interest, have much less attention. Experimental acoustical and high speed video studies of the Coconut Rhinoceros Beetle (*Oryctes rhinoceros*) and the Oriental Flower Beetle (*Protaetia orientalis*) have motivated large scale simulations accounting for three dimension flow, compressibility, and fluid structure interactions. Computational fluid dynamics simulations were performed using the unsteady compressible flow solver (CAESIM, Adaptive Research, Inc.) using a high resolution (TVD) methodology. Models of the wing flapping motion were accomplished using mesh deformation techniques with the flapping following from rotation with prescribed bending and coupled rotation and translation from the wing's hinge position. Fluid structure interactions with respect to the wing's flexibility are investigated in terms of the wing bending and the leading edge vortex formation.

9:45

4aCA3. High-performance computation toward large-scale underwater acoustics modelling. Noriyuki Kushida (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA, nkushi+whoi@gmail.com) and Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA)

The complex nature of the ocean has long presented a challenge for researchers in the field of oceanography, including in the area of underwater acoustics. As a result, significant efforts have been made to develop accurate numerical models to better understand and study the ocean. Established models such as Acoustic Toolbox and Range-dependent Acoustic Models have proven to be effective for modelling sound propagation. However, these models were designed to run on single-core computers, and there is potential to optimise their performance on modern systems. The use of General-purpose graphics processing units (GPGPUs) and Single Instruction/Multiple Data (SIMD) computing units is one way to achieve this optimization. In addition to improving the computational speed of established models, these optimized models can also contribute to implementing more sophisticated modelling approaches, such as broad-band modelling. In this study, we will discuss the optimized performance of established codes, particularly those based on parabolic equation methods (PE), and also discuss the results of advanced modelling techniques.

10:05–10:20 Break

10:20

4aCA4. Large-scale simulation of high-intensity focused ultrasound with Sierra/SD. Benjamin C. Treweek (Computational Simulation, Sandia National Labs., P.O. Box 5800, Albuquerque, NM 87185, btrewee@sandia.gov), Jacob H. Brody, Alper Erturk (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), S. H. Swift (Photonic & Phononic Microsystems, Sandia National Labs., Albuquerque, NM), Chandler Smith, Cameron A. McCormick, Timothy Walsh (Computational Simulation, Sandia National Labs., Albuquerque, NM), and Nathan W. Moore (Dynamic Material Properties, Sandia National Labs., Albuquerque, NM)

High-frequency simulations of acoustic wave propagation are known to be computationally challenging, and the difficulty is compounded for large domains. For problems like high-intensity focused ultrasound (HIFU) with coupling between piezoelectric and acoustic media, both challenges are present. The larger domain sizes, higher-order elements, and greater levels of refinement necessary in such simulations result in millions of degrees of freedom, large system matrices, and substantial memory requirements, raising the need for parallel high-performance computing (HPC). Sierra/SD is a massively parallel HPC application developed for finite element method simulations in structural dynamics and acoustics. In this work, Sierra/SD is used to simulate an acoustic pulse from a piezoelectric transducer focused on an elastic scatterer in a fluid medium. Three-dimensional simulation results are presented for the acoustic field in the fluid and the stress field in the scatterer, and performance is compared between Sierra/SD and COMSOL Multiphysics for smaller geometries. Finally, to showcase the expanded analysis possibilities afforded by HPC for HIFU, an example is presented using a support vector machine to determine a decision boundary for maximum stress in the scatterer. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

10:40

4aCA5. Big data to streamlined app: Nationwide traffic noise prediction. Mylan R. Cook (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com), Kent L. Gee, Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Shane V. Lympany (Blue Ridge Res. and Consulting, Asheville, NC)

A vast amount of data can be processed using high-performance computing (HPC), but sometimes, the HPC hardware requirements are cost prohibitive. However, by using HPC to create reduced-order models that require much less processing power and data storage, the benefits of HPC can be realized for a wider range of users. For example, detailed traffic noise predictions can be made on small geographic scales using the US Federal Highway Administration's Traffic Noise Model (TNM), but when considering a nationwide scale, TNM becomes infeasible for the typical user because of computational cost. Additionally, though calculating an annually averaged overall level may be within a user's computational capacity, incorporating temporal and spectral variability increases complexity. Using HPC on hourly traffic counts at locations across the country together with published traffic trends and TNM equations, a streamlined app has been made to efficiently predict traffic noise at roads across the nation with temporal and spectral variability. This app, which presently requires less than 700 MB of stored geospatial data and models, incorporates user inputs such as location, time period, and frequency, and gives predicted spectral levels within seconds.

11:00

4aCA6. Update on the Department of Energy Leadership Computing Facilities. Ralph T. Muehleisen (Energy Systems and Infrastructure Anal., Argonne National Lab., 9700 S. Cass Ave., Bldg 362, Lemont, IL 60439-4801, rmuehleisen@anl.gov)

Did you ever say to yourself "Boy I wished I had a bigger computer." Do you have a modeling problem that could benefit from "really" high performance computing but do not know how to get it? Are you trying to create big models but find that cloud computing is the wrong answer but that a large high-performance computer might be the right answer? The Department of Energy (DOE) Leadership Computing Facilities (LCF) are a set of the highest performance publicly accessible computing facilities in the US. ALCF current has Frontier (#1), Summit (#5), and Perlmutter (#8) among the top 10 in the Top500 HPC list. A two exascale floating point computer, Aurora at Argonne National Laboratory, is set to come online as part of LCF in the next year. This presentation will update the previous introductory presentation on the DOE LCF and explain how researchers and industry can gain access and training to use these incredible facilities.

Session 4aED

Education in Acoustics, Physical Acoustics, Noise, and Architectural Acoustics: When Doing it Right Goes Wrong

Kimberly A. Riegel, Cochair
Physics, Farmingdale State College, 2350 Broadhollow Road, Farmingdale, NY 10803

John R. Buck, Cochair
Electrical and Computer Engineering, UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Chair's Introduction—8:00

Invited Papers

8:05

4aED1. Lessons learned from failures. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 48504, dludwigs@kettering.edu)

In certain communities of engineering education, especially where intersections with entrepreneurship shape the discussion, "Learning from Failure" has become a catch phrase. Usually these refer to pedagogical approaches in which students recognize the benefits of trying, failing, reflecting, and trying again. This iterative process also describes a teacher's journey over a decade, learning to implement and practice research-based educational techniques with patience and a focus on the students. However, the stakes can be much higher when teachers try and fail! Examples from an elective course designed for a wide engineering audience, Acoustics in the Human Environment, will trace its evolution. It began as an experiment in a studio classroom format and became an asynchronous online class intended to mirror an engineering work environment. Milestones marking the steps of this journey: parallels to iteration involved in the engineering design process; suggested elements that provide a safety net for teaching failure, and other lessons learned along the way.

8:25

4aED2. A tale of two activities. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

It was the best of intentions, it was the worst of receptions, it was the semester for scientific engagement, it was the semester of sitting silently in the classroom, we had meaningful conversations about the science of sound, we had hushed whispers of hiding information from the course instructor—in short, the semester was like many semesters before it. With apologies to Charles Dickens, I will share the experience of having two sections of the same class, in the same semester, with the same conditions, which ultimately yielded dramatically different levels of student engagement. As an example, the difference in how each section received an in-class activity will be discussed. I will share lessons I learned as a teacher and ways that I hope to grow as an instructor based on this experience.

8:45

4aED3. Project based learning (PBL): Less project, more problem. Kimberly A. Riegel (Phys., Farmingdale State College, 222-05 56th Ave., Bayside, NY 11364, kriegel@qcc.cuny.edu)

Project based learning (PBL) has been shown to be an effective and engaging way of teaching. Though it has been used in many disciplines, it has been specifically employed in physics and engineering often and has been written about in many journals showing the effective outcomes and student growth when this technique is employed. Therefore, when I had a small intimate class of 8 students for introductory physics, the prospect of designing a PBL based course to engage students in thoughtful projects and effective teaching seemed exciting. It took only four weeks before the course structure began to fall apart. By the end of the course, my carefully planned course was completely unrecognizable. In this presentation, I will outline some example projects and what went wrong. I will also discuss how I was able to pivot during the course, the course outcomes and some lessons learned.

9:05

4aED4. Making group-work work: Navigating conflict and encouraging collaborative revision. Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu)

Group work, especially for comprehensive class projects, is ubiquitous in higher education, yet despised by many students. In college, many students simply do not learn how to interact when in groups and how to collaborate effectively. Furthermore, by limiting student assessment of group work solely to a final project grade, students do not have the opportunity to collaboratively reflect on feedback and revise. Inspired by the massive failure of group comprehensive projects in my bioacoustics course, I will share tips I learned to navigate intra-group conflict, provide students with critical feedback on work at 3 key periods throughout the course, encourage collaborative revision, and engage with other groups via peer feedback.

9:25–9:40 Break

9:40

4aED5. Project-based learning goes wrong: the trials, tribulations, and triumphs of managing first-year and fourth-year engineering projects. Martin S. Lawless (Mech. Eng., SUNY Maritime, 6 Pennyfield Ave., S&E 2-38, Bronx, NY 10465, mlawless@sunymaritime.edu) and Kathryn R. Gosselin (Mech. Eng., SUNY Maritime, Bronx, NY)

Project-based learning (PBL) has many advantages for both students and instructors. The students are afforded the opportunity to work on self-driven, meaningful projects that model engineering problems they may see in their future careers. For the instructor, PBL emphasizes problem-solving strategies, requires students to learn relevant technical skills through self-study, underscores the importance of team work, and encourages student buy-in. However, PBL techniques present their own sorts of difficulties, especially when applied in large courses of 50 + students. For example, an instructor must strike a balance between fostering creativity and enthusiasm, while also keeping students on task and focused on the purpose of the project. Managing many small groups or a few large groups can be logistically challenging due to the sheer number of unique projects and students involved. This presentation will focus on two engineering courses, first-year introduction to engineering and fourth-year vibrations, in which project-based learning was employed with partial success. In these courses, there were logistic complications, issues with team dynamics, and problems with final presentations that can hopefully serve as good lessons (or warnings) for others interested in developing project-based courses.

10:00

4aED6. Reflecting on reflections: A case study of disappointment in student writing assignments. John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu)

Several authors and instructors find that reflection essays and other similar journaling exercises help students to self-identify strengths and weaknesses in their understanding of class material. Even better, some students begin developing metacognitive skills identifying higher level strengths and patterns in their study habits. For several semesters, I required students in my undergraduate linear systems classes to submit a short 200–300 word reflection essay with each homework assignment identifying which topics on the assignment they had mastered, and which topics they still found confusing, and why. A vast majority of the submitted essays were disappointing laundry lists: "Problem 1 was about linearity, Problem 2 was about convolution…," completely devoid of any thoughtful self-reflection. I found students' resistance to engage meaningfully in this practice especially frustrating because my own pedagogical development benefitted from a consistent writing practice. This talk will also brainstorm some potential directions or approaches for future iterations of this activity in my classes.

10:20

4aED7. Addressing blank stares and unintended paths in student-centered learning. Jill K. Nelson (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, jnelson@gmu.edu)

Despite decades of research supporting their effectiveness, adoption of student-centered teaching practices is slow, and many instructors try active learning only a few times before reverting to traditional teaching methods. Like most skills, learning to design and facilitate student-centered activities takes practice, and new adopters often encounter frustrations as they begin to implement these approaches. Sharing best practices for smoothing the implementation of active learning can increase the likelihood that instructors will adopt and sustain evidence-based practices. I will describe two common hurdles (and how I address them) when implementing student-centered activities in my undergraduate signals and systems courses. (1) The blank stare: Whether it be resistance to active learning practices, lack of understanding of the content, or just not enough sleep the night before, students often struggle to get started when handed an active learning activity. (2) The path not intended: One of my favorite parts of teaching is designing activities that (I believe) are engaging, relevant, and help students draw connections among important concepts. More often than I'd like to admit, however, the approach I have in mind when designing the activity bears little resemblance to the (equally valid) approach students take when completing it.

10:40

4aED8. A midshipman's experience in leading the underwater acoustics and sonar class through a sound speed versus temperature experiment. Riley G. Plosica (Phys., U.S. Naval Acad., 572C Holloway Rd., Chauvenet Hall Rm. CH040, Annapolis, MD 21402, riley.plosica@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

Oceanography and General Science majors are required to take Underwater Acoustics and Sonar during their junior or senior year at USNA. Our studio classroom has eight work stations where students measure sound speed in water. Author RGP was given the unique opportunity to lead her peers and present the sound speed demonstration in class. In this leadership role, it was necessary for RGP to motivate her fellow students as she demonstrated the pulse-echo experiment and emphasized its significance. First, students observed how

the apparatus functioned and how to take measurements. Next, they broke into groups of four, and conducted the hands-on experiment as RGP communicated each step. Student pairs worked with the apparatus while two others observed. A learned skill during this process was managing the group in the observing role. This was a challenge. Peer-leadership experiences often required the leader to create an atmosphere of respect due to the lack of hierarchy that exists in the typical professor led classroom. Upon conclusion, RGP realized that there was success because students were able to submit accurate experimental measurements. While balancing an upbeat classroom with a constructive learning environment, RGP found the teaching experience instructional and her peer's response gratifying.

11:00

4aED9. When a simple demonstration of resonance in a forced oscillation system proved that resonance does not exist. Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

Students in an introductory musical acoustics course attempted to recreate a simple forced oscillation system to verify its resonant frequency. The procedure followed ideas and instructions found in a YouTube video from Animated Science to show resonance. This was tested by the instructor prior to the experiment, who found it to be straightforward to assemble and to make the appropriate measurements. The calculated frequency was found to be within 2.5% of the experimental value. The students were given two identical springs, each with a spring constant of 30 N/m and a 100 g mass, as well as a function generator and apparatus needed to connect the system. The video shows the mass suspended between the two springs with one end driven by a function generator while the other end is fixed. A meterstick was placed behind the mass spring system to get a rough estimate of the peak amplitude. 6 separate groups of 3 students each tried to assemble the system as shown. Not a single set-up worked as it did in the video. The reasons for this were as varied as the number of groups doing the experiment. The experiment was refined over several years into a well-executed experiment.

11:20–11:50
Panel Discussion

Session 4aNSa

Noise, Computational Acoustics, and Physical Acoustics: Aircraft Noise

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

Contributed Papers

8:05

4aNSa1. On the notional impact of background noise on the human response to UAVs. Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov)

Proponents of UAV-based industries would like to think that their aircraft will be incapable of causing annoyance (among other negative responses) in the communities that they serve due to the noise of their vehicles simply not being heard over the existing soundscape. No measures of noise that are in wide use today for aviation take background sound into consideration and, thus, would not predict this outcome. This work is an attempt to bridge this gap by looking at past results from psychoacoustic research in which background sounds appear to play a significant role (not just for aviation sources). It includes information on the problem of human auditory detection in general—what is the transductive mechanism of the ear, how does one quantify detectability, etc., and it attempts to make the connection between detectability and annoyance. A notional schematic version of how measures/predictions of these two aspects may be incorporated into a single assessment of noise is given. Overall, detectability is shown to be very complex to predict relative to conventional noise metrics that are used to correlate with human response. Given this, the outlook for the use of measures of detectability in scientific, industrial, and regulatory applications is discussed.

8:20

4aNSa2. Towards the environmental noise assessment of Unmanned Aircraft Systems operations. Antonio J. Torija Martinez (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Manchester M5 4WT, United Kingdom, A.J.TorijaMartinez@salford.ac.uk), Carlos Ramos Romero, Nathan Green, and Richard Hughes (Acoust. Res. Ctr., Univ. of Salford, Manchester, United Kingdom)

The rapid expansion of the Unmanned Aircraft Systems (UAS) sector poses a significant challenge for acoustic engineers working on the environmental noise impact assessment of these new aerial technologies. As widely recognised, the noise signatures and operational profiles of UAS differ significantly from conventional aircraft, and therefore, existing measurement procedures and assessment methods might not be able to provide an appropriate assessment of UAS noise. There are specific questions that should be addressed for a wider deployment of UAS operations without compromising communities' health and well-being due to noise exposure: (i) how will communities respond to the unconventional noise signatures of UAS? and (ii) what measurement procedures and metric can ensure an accurate assessment of UAS noise impact? This publication describes the work carried out for the measurement and development of acoustic hemispheres for UAS operations containing sound level, frequency, and directivity information; and how this acoustic data can be used for noise mapping and auralization. Furthermore, a review of the state-of-the-art of noise metrics for UAS is presented, with a focus on existing gaps and further work required.

8:35

4aNSa3. Attention to rattles and a non-equal-energy model are required for proper sonic boom assessment. Paul Schomer (Schomer and Assoc., 2117 Robert Dr., Champaign, IL 61821, p.schomer@gmail.com)

This paper is concerned with the assessment of community response to sonic booms or blasts. It summarizes and analyzes the totality of results from studies in the English language that used real booms or blasts, with subjects in real buildings. In acoustics, we are accustomed to noise sources operating in accordance with the equal-energy principle (a 1 dB increase in amplitude is equivalent to a 1 dB increase in duration). The results show that rattles are the most important attribute contributing to the annoyance engendered by sonic booms/blasts, and that the process is not equal-energy. Rather, the equivalent annoyance generated by a change of 1 dB in the C-weighted boom or blast amplitude is equal to about a 1.5 to 2 dB change in the boom or blast duration where the exchange rate is defined to be 1 over these changes in duration, 0.67 and 0.5, respectively. The exchange rates found in several sonic boom/blast noise studies are given, and as an example, the exchange rate for the historical Oklahoma City study is calculated. The conclusions from the Long-Term Sonic Boom Noise Environments study are examined in relation to the range of exchange rates found in other boom/blast studies.

8:50

4aNSa4. Particle image velocimetry of supersonic jets with micro vortex generators. Mohammad Saleem (Aerosp. Eng. or Eng. Mech., Univ. of Cincinnati, 745 Baldwin Hall, Cincinnati, OH 45221, saleemmd@mail.uc.edu), Omar L. Rodriguez, Aatresh Karnam, Ephraim Gutmark (Aerosp. Eng. or Eng. Mech., Univ. of Cincinnati, Cincinnati, OH), and Junhui Liu (Naval Res. Lab., Washington, DC)

The flow field of supersonic jets with micro vortex generators (MVGs) is investigated using Particle Image Velocimetry (PIV) to shed light on the noise reduction mechanisms associated with such devices. Recently, MVG nozzles have been developed as a new noise reduction technology through the collaboration between the University of Cincinnati and the Naval Research Laboratory. MVGs were implemented on model scale nozzles representative of GE F404 engine nozzles, and noise reductions up to -6 dB have been observed in both microphone measurements and LES simulations. The PIV measurements of the flow field reveal that the MVGs alter the shock cell spacing and strengths in the jet plume and substantially reduce the mean axial velocity at the jet axis. In addition, streamwise vortices generated by MVGs enhance shear layer mixing and redistributes turbulence along the shear layer to narrower regions. The measured flow field quantities are extracted to explain flow field changes on the noise reductions observed in the acoustic far field.

9:05

4aNSa5. Noise exposure of spectators at Utah air show. David C. Lawrence (Dept. of Comput. Sci. and Eng., Culver Academies, Chicago, IL, david.lawrence@culver.org), Kent L. Gee, and Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Spectators are exposed to a variety of loud noises at air shows, which can contribute to noise-induced hearing loss. A previous study [Pääkkönen *et al*., *Appl. Acoust.* **64**, 121–127 (2003)] on three Finnish air shows concluded that their air shows did not seem to exceed damage risk levels. This study investigated noise that spectators are exposed to at the Warriors Over the Wasatch Air and Space show at Hill Air Force Base. Noise dosage, exposure times, and L_{Aeq} for various aircraft were analyzed. The L_{Aeq} ranged from 75 to 95 dB_A and the total noise dosage for the airshow was 118 %, exceeding the NIOSH recommended exposure limits with most of the noise coming from the fighter jets. This shows that spectators should use hearing protection if they plan to stay for the entire air show or at least during portions of an air show that contain fighter jets.

9:20

4aNSa6. Development of gulfstream acoustics simulator. Tongan Wang (Gulfstream Aerosp. Corp., 500 Gulfstream Rd., Savannah, GA 31402, tongan.wang@gulfstream.com)

An acoustic simulator has been developed at Gulfstream Aerospace Corp. to evaluate aircraft interior acoustic environments. Many acoustic simulators developed in the past were based on headphones, but they could not be used to evaluate communication between passengers. Ambisonics and wavefield synthesis are popular methods to recreate sound fields in a 3D space, but they are either limited in the size of the recreated sound field or require great number of speakers. Therefore, a modal-based approach was used to recreate the sound field inside the Gulfstream acoustics simulator. This paper will discuss the construction of the simulator mockup, the analysis to determine the speaker placement, the test to measure the acoustic environment, and the hardware and software systems to control the simulator playback. The acoustic simulator will be used to demonstrate the acoustic environment on different aircraft in different flight conditions, evaluate the effect of improvement options, and develop advanced technologies to create finest aviation experience.

THURSDAY MORNING, 11 MAY 2023 CHICAGO C, 10:00 A.M. TO 11:45 A.M.

Session 4aNSb

Noise and Architectural Acoustics: Flanking Paths: Finding them, Solving Them and Improvement in ASTC When You Do

Bonnie Schnitta, Chair
SoundSense, LLC, 39 Industrial Rd., Unit 6, PO Box 1360, Wainscott, NY 11975

Chair's Introduction—10:00

Invited Papers

10:05

4aNSb1. Visual identification of flanking paths through sound-isolating wall assemblies. Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, samuelunderwood@unomaha.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

The sound isolation performance of wall and floor/ceiling assemblies in the field continue to be inhibited by flanking paths, which may be difficult to identify and remedy. When multiple paths are present, ranking the relative impacts of each path—both the obvious and the obscure—poses a daunting challenge for practitioners. This paper presents a case study of multiple wall assemblies wherein flanking paths were visually identified with camera-based acoustic mapping technology. Apparent Sound Transmission Class (ASTC) ratings measured in the field are compared to published laboratory test ratings and software-modeled performance. Where possible, post-remediation measurements of ASTC rating are reported. Results from this case study highlight the advantages of camera-based visualization and the importance of mitigating flanking paths throughout design and construction.

10:25

4aNSb2. Quantifying the effect of flanking sound transmission on sound insulation and speech privacy metrics. Roderick Mackenzie (Soft Db, 250 Ave. Dunbar, Ste. 203, Montreal, QC H3P 3E5, Canada, r.mackenzie@softdb.com)

In modern commercial office design, sustainable building codes and quality standards often demand certain levels of sound insulation to ensure sufficient speech privacy or freedom from distraction for the occupants. Missing from the minimum requirements of these documents, however, is guidance on eliminating flanking weaknesses that repeatedly and significantly degrade the experienced speech privacy. This paper presents case studies of common acoustical weaknesses found between closed offices and meeting rooms. Through the use of sound intensity measurements and acoustical imagery, the sound power of each weakness is calculated and shown relative to the sound power of the separating partition. By comparing these results to partitions without these weaknesses, the effective reduction in speech privacy is demonstrated using the ASTC, NIC, and SPC metrics. Weaknesses examined and ranked include, open ceiling plenums, lack of or ineffective door seals, door closure pressure, continuous door frames, façade mullions and transoms, interior windows and their frames, lack of acoustical sealant, uninterrupted gypsum on side walls, common door frames at the edges of partitions, modular and operable walls and their junctions, continuous heating elements, thin continuous floor toppings, and ventilation duct crosstalk.

10:45

4aNSb3. Flanking noise transmission in a healthcare facility. Ben Davenny (Acentech, Cambridge, MA) and Jay M. Bliefnick (Acentech, 33 Moulton St., Cambridge, MA 02138, jbliefnick@acentech.com)

Healthcare building constructions have some similarities and some differences compared with those of other building types. This presentation will examine door gasketing problems and perimeter sound isolation leaks found at an outpatient healthcare facility. The observations and test results for the problem conditions will be discussed along with our recommended solutions. Anecdotal feedback from the facility may be available by the time of the presentation.

Contributed Paper

11:05

4aNSb4. Sound transmission through raised access floors. Bryce Lemert (Aercoustics Eng. Ltd., 1004 Middlegate Rd., Ste. 1100, Mississauga, ON L4Y 0G1, Canada, brycel@aercoustics.com) and Eric Salt (Aercoustics Eng. Ltd., Mississauga, ON, Canada)

Raised access flooring systems have become an increasingly popular option across the construction industry, but especially in the office environment. These systems allow for a modular design where electrical conduit and data cables can be easily accessed, and the underfloor air supply can eliminate the need for overhead mechanical systems. While the design has many benefits, the sound transmission through the plenum is a concern as the system introduces an additional flanking path to adjacent rooms. A variety of underfloor baffle options exist to help reduce the transmission through the plenum, each with their own challenges. There is currently limited understanding of the effect of sound transmission through the raised access plenum between adjacent rooms. This paper provides an introduction to the acoustics surrounding the raised access flooring system based on experiences and lessons learned in an as-built environment, both in terms of airborne and structure borne sound transmission. In particular, this paper examines the flanking path that is introduced through the plenum under walls constructed above the raised floor and the effect of underfloor baffles on this sound transmission.

11:20–11:45
Panel Discussion

Session 4aPAa

Physical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration: Acoustic Sensing in Planetary Environments

Andi Petculescu, Cochair
Department of Physics, University of Louisiana at Lafayette, 240 Hebrard Blvd., Broussard Hall, Room 103, Lafayette, LA 70503-2067

Timothy G. Leighton, Cochair
Institute of Sound and Vibration Research, University of Southampton, Southampton, SO17 1BJ, United Kingdom

Chair's Introduction—8:05

Invited Papers

8:10

4aPAa1. Development of balloon-based seismology for venus through earth-analog experiments and simulations. Siddharth Krishnamoorthy (Jet Propulsion Lab. California Inst. of Technol., 4800 Oak Grove Dr., Pasadena, CA 91109, siddharth.krishnamoorthy@jpl.nasa.gov), Daniel Bowman (Sandia National Labs., Albuquerque, NM), Emalee Hough, Zach Yap (Oklahoma State Univ., Stillwater, OK), John D. Wilding (California Inst. of Technol., Pasadena, CA), Jamey Jacob, Brian R. Elbing (Oklahoma State Univ., Stillwater, OK), Leo Martire, Attila Komjathy, Michael Pauken, James Cutts (Jet Propulsion Lab. California Inst. of Technol., Pasadena, CA), and Jennifer Jackson (California Inst. of Technol., Pasadena, CA)

Balloon-based seismology through the study of low-frequency seismo-acoustic signals (infrasound) has gained acceptance as a viable way to study seismic activity on Venus. Balloon-based barometers have the potential to detect and characterize atmospheric waves launched by venusquakes and volcanic eruptions while offering substantially longer instrument lifetimes in the Venus middle atmosphere, where temperature and pressure are significantly more benign (0–100 °C, ∼1 atm) as compared to the surface (>460 °C, ∼90 atm). One of the major challenges in performing balloon-based seismology on Venus is the absence of ground-truth data for event identification and discrimination. To address this challenge, our activities are aimed at building a catalog of terrestrial balloon-recorded infrasound signals of geophysical provenance, using which signal predictions can be extended to Venus and the detectability of events can be analyzed. We will highlight our recently concluded Balloon-based Acoustic Seismology Study (BASS) flight campaign, which served as Earth-analog experiments for Venus balloon-based seismology. Data collected were used to validate seismo-acoustic simulation tools, which are being expanded to include the Venus atmosphere. These tools will used to generate predictions of infrasound signals from geophysical events on Venus. We will also provide perspective on directions for future instrument development for Venus balloon flights.

8:30

4aPAa2. A Martian sonic anemometer. Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu), Don Banfield (NASA Ames Res. Ctr., Mountain View, CA), Rishabh Chaudhary, Tim J. Cheng (Mech. Eng., Tufts Univ., Medford, MA), Anthony Colaprete, Amanda Cook (NASA Ames Res. Ctr., Mountain View, CA), Tara Curran (Mech. Eng., Tufts Univ., Medford, MA), Arwen Dave (NASA Ames Res. Ctr., Mountain View, CA), Julia Huckaby (Mech. Eng., Tufts Univ., Medford, MA), Ian Neeson (VN Instruments, Brockville, ON, Canada), Emmett Quigley, Abraham Rademacher (NASA Ames Res. Ctr., Mountain View, CA), Freidlay Steve (Mech. Eng., Tufts Univ., Medford, MA), and Bruce White (NASA Ames Res. Ctr., Mountain View, CA)

On Mars, we have yet to fully quantify atmospheric turbulent transport [Banfield, *JASA* (2016)]. The key required instrument is a wind sensor that can resolve horizontal and vertical winds, and the perturbations associated with turbulent eddies. On the surface, 10–20 Hz sampling and a sensitivity of 5 cm/s are required (McBean, 1972). Viking, Pathfinder, Curiosity, and Insight used hot film or hot wire anemometers that can be confused by radiative heating and sample at approximately 1 Hz. Sonic anemometry is an alternative high speed, high accuracy approach which uses differences in the acoustic time of flight to determine flow speed. However, operation on Mars can be challenging due to the low acoustic source strength and high attenuation in the thin (6—11 mbar) CO_2 atmosphere, as well as large temperature variations (−100 °C to 20 °C). In this paper, we describe an instrument that addresses these challenges: a 3 axis ultrasonic anemometer using custom low impedance, wide band transducers, low noise electronics, and an optimized mechanical structure. With an earlier prototype in wind tunnel tests in 6 mbar CO_2, we demonstrated resolution better than 10 cm/s, and an accuracy of 3% at flow speeds up to 12 m/s. Data from an instrument flight to 128 000 feet on a stratospheric balloon will be presented, and progress towards a flight qualifiable instrument will be described.

8:50

4aPAa3. The sound of the ingenuity Helicopter on Mars. Ralph Lorenz (JHU Appl. Phys. Lab, 11100 Johns Hopkins Rd., Laurel, MD 20723, Ralph.Lorenz@jhuapl.edu), Sylvestre Maurice (IRAP, Toulouse, France), and Baptiste Chide (Space and Planetary Exploration Team, Los Alamos National Lab., Los Alamos, NM)

The sounds of the Ingenuity Helicopter flying in the Martian atmosphere are among the most notable recordings of the microphone on the SuperCam instrument on the Mars 2020 Perseverance Rover. Distinct acoustic signatures of the helicopter were recorded on the 4th, 5th, 6th, and 8th flights. The detected signatures are around 84 Hz and (occasionally) at 168 Hz, at the blade crossing frequency and its first harmonic. Several higher harmonics were prominent in hover tests in short-range recordings in a test chamber on Earth; these are attenuated by CO2 absorption at the 50 m-plus ranges on Mars. Doppler shift of the 84 Hz signal can be measured and is consistent with the trajectory measured with Ingenuity's navigation camera and inertial navigation unit, and documented by Perseverance's cameras. A striking feature of the sound recordings is an unanticipated deep modulation of the signals with nulls spaced by around 15–20 s, superposed on the simple and expected decline in amplitude with distance. We have evaluated and rejected models of multipath sound interference as requiring implausibly strong near-surface temperature gradients. We find instead that the modulation appears to be the signature of a slight asynchrony between the rotation rates of the two coaxial rotors, such that the blade-crossing azimuth rotates slowly during flight, resulting in a "lighthouse" sweeping of the radiated sound pattern. Analysis of blade orientations seen in the shadow of the helicopter observed in down-looking navigation images supports this model.

9:10

4aPAa4. Modeling acoustic propagation on Mars. Martin Gillier (ISAE-Supaero, Université de Toulouse, 10, Ave. Édouard-Belin, Toulouse 31055, France, martin.gillier@isae-supaero.fr), Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), Naomi Murdoch, Alexander Stott (ISAE-Supaero, Université de Toulouse, Toulouse, Haute-Garonne, France), Xavier Jacob (Institut de Mécanique des Fluides de Toulouse, Université de Toulouse III Paul Sabatier, INP, CNRS, Toulouse, France), David Mimoun (ISAE-Supaero, Université de Toulouse, Toulouse, France), and Sylvestre Maurice (IRAP, Toulouse, France)

It is important to understand how the atmospheric conditions affect the propagation of sound waves on Mars, both for the study of the acoustic sources and of the atmosphere itself. As 3 microphones are already on Mars (2 on NASA/Perseverance and 1 on CNSA/Zhurong) and potentially more to come, a model of the propagation of sound in the Martian surface layer was needed. A first-principle acoustic attenuation model is used to quantify the excess attenuation of the Martian atmosphere. Thanks to inputs extracted from a global circulation model of Mars (the LMD Martian Climate Database), we are able to compare the magnitude of attenuation at different places and times, from 10 Hz to 20 kHz. Propagation effects are also considered by adapting a parabolic equation code to the Martian conditions. This accounts for strong temperature gradients near the surface, as well as gradients of wind speed. The acoustic reflection on the ground assumes a semi-porous medium. The turbulent fluctuations of temperature and wind speed are implemented using the frozen medium approach. The model shows that characteristics of sound waves propagation depend drastically on the local time. It provides a useful framework to interpret acoustic signal recorded on Mars. The model shows that characteristics of sound waves propagation depend drastically on the local time. It provides a useful framework to interpret acoustic signal recorded on Mars.

9:30

4aPAa5. Martian Wind and turbulence heard by the SuperCam microphone on the perseverance rover. Alexander Stott (ISAE-Supaero, Université de Toulouse, 10 Ave. Edouard Belin, Toulouse, Haute-Garonne 31400, France, alexander.stott@isae-supaero.fr), Naomi Murdoch (ISAE-Supaero, Université de Toulouse, Toulouse, France), Martin Gillier (ISAE-Supaero, Université de Toulouse, Toulouse, France), Don Banfield (NASA Ames, Ithaca, NY), Tanguy Bertrand (LESIA, CNRS, Sorbonne Université, Meudon, France), Baptiste Chide (Space and Planetary Exploration Team, Los Alamos National Lab., Los Alamos, NM), Manuel De la Torre Juarez (JPL, Caltech, Pasadena, CA), Ricardo Hueso (Fisica Aplicada, Escuela de Ingenieria de Bilbao, Universidad del Pais Vasco UPV/EHU, Bilbao, Spain), Ralph Lorenz (JHU Appl. Phys. Lab, Laurel, MD), German Martinez (Lunar and Planetary Inst., Houston, TX), Asier Munguira (Fisica Aplicada, Escuela de Ingenieria de Bilbao, Universidad del Pais Vasco UPV/EHU, Bilbao, Spain), Luis Mora Sotomayor, Sara Navarro (CSIC-INTA, Centro de Astrobiologia (CAB), Madrid, Spain), Claire Newman (Aeolis Res., Chandler, AZ), Paolo Pilleri (IRAP, Toulouse, France), Jorge Pla-Garcia (CSIC-INTA, Centro de Astrobiologia (CAB), Madrid, Spain), Nicolas Randazzo (Univ. of AB, Edmonton, AB, Canada), Jose Antonio Rodriguez Manfredi (CSIC-INTA, Centro de Astrobiologia (CAB), Torrejón de Ardoz, Spain), Agustin Sanchez-Lavega (Fisica Aplicada, Escuela de Ingenieria de Bilbao, Universidad del Pais Vasco UPV/EHU, Bilbao, Spain), Michael Smith (Goddard Space Flight Ctr., NASA, Greenbelt, MD), Daniel Viudez Moreiras (CSIC-INTA, Centro de Astrobiologia (CAB), Madrid, Spain), Nathan Williams (JPL, Caltech, Pasadena, CA), Sylvestre Maurice (IRAP, Toulouse, France), Roger Wiens (Purdue Univ., Lafayette, IN), and David Mimoun (ISAE-Supaero, Université de Toulouse, Toulouse, France)

On top of listening to laser shots, rover sounds and the Ingenuity rotorcraft, SuperCam's Mars microphone has recorded over 7 hours of ambient background noise on Mars. These background recordings contain signal due to the Martian wind. Through a comparison to the meteorological data recorded by the MEDA (Mars Environmental Dynamics Analyzer), we can determine the relationships between the microphone data, the wind and the atmospheric stability. Based on these relationships, we have determined a way to estimate the wind speed using the microphone through Gaussian process regression, a machine learning technique. Owing to the sampling rate of 25 000 samples per second, the microphone data can be used to examine Mars' atmospheric dynamics at high frequencies, as yet unexplored on Mars. We will demonstrate how the wind speed estimates from the microphone provide an assessment of turbulence at fine scales, shedding light on the dissipative regime on Mars. One particularly interesting signal recorded by the microphone was a dust devil, which had fast varying winds within the walls of its vortex and signal from dust particles hitting the rover. Combining the microphone data with information from the MEDA sensors and navigation camera (Navcam) images enabled a full parameterization of this event.

9:50

4aPAa6. Modeling support to acoustic sensing at the surface of Mars. Xavier Jacob (Institut de Mécanique des Fluides de Toulouse, Université de Toulouse III Paul Sabatier, INP, CNRS, Toulouse, France, Institut de Mécanique des Fluides de Toulouse, 1, Allée du Professeur Camille Soula, Université de Toulouse III Paul Sabatier, 118 Rte. de Narbonne, Toulouse 31000, France, xavier.jacob@univ-tlse3.fr), Baptiste Chide (Space and Planetary Exploration Team, Los Alamos National Lab., Los Alamos, NM), Sylvestre Maurice (IRAP, Toulouse, France), and Tanguy Bertrand (Laboratoire d'Etudes Spatiales et d'Instrumentation en Astrophysique (LESIA), Observatoire de Paris, Université PSL, CNRS, Sorbonne Université, Univ. Paris Diderot, Sorbonne Paris Cité, Meudon, France)

Acoustic propagation in the atmosphere of Mars is the subject of renewed interest since the presence of microphones on the surface of the red planet. The analysis of the recordings collected by the SuperCam microphone of the NASA Perseverance rover has already shown that the acoustic properties of Mars atmosphere do fit the existing models, and so far, the meteorological data have been available. On this basis, classical atmospheric propagation models have been adapted, including the effects of ground, temperature gradient, wind, and turbulence. A sensitivity study is presented, using two configurations that correspond to the main actual sources and propagation paths experienced on Mars : one similar to the point-like pulsed source that is generated by the expansion of the plasma of the LIBS technique, and the other analogue to the tonal low frequency noise emission of the ingenuity drone. Modeling results show to what extend this two configurations can be use to assess ground or atmospheric properties.

10:10–10:25 Break

10:25

4aPAa7. Acoustic recording of near-surface turbulence on Mars. Sylvestre Maurice (IRAP, 9 Ave. Colonel Roche, Toulouse 31500, France, sylvestre.maurice@irap.omp.eu), Philippe Blanc-Benon (LMFA, Ecole Centrale de Lyon, Ecully, France), Baptiste Chide (Space and Planetary Exploration Team, Los Alamos National Lab., Los Alamos, NM), Tanguy Bertrand (LESIA, Meudon, France), Naomi Murdoch (ISAE-SUPAERO, Toulouse, France), Ralph Lorenz (JHU Appl. Phys. Lab, Laurel, MD), Xavier Jacob (IMFT, Toulouse, France), Didier Dragna (LMFA, Ecole Centrale de Lyon, Ecully, France), Alexander Stott (ISAE-SUPAERO, Toulouse, France), Martin Gillier (ISAE-Supaero, Toulouse, France), Manuel De la Torre Juarez (JPL, Pasadena, CA), Jose Antonio Rodriguez Manfredi (CAB-INTA, Torrejón de Ardoz, Spain), David Mimoun (ISAE-SUPAERO, Toulouse, France), and Roger Wiens (Purdue, Lafayette, IN)

To study Mars chemistry, SuperCam onboard the NASA Perseverance rover fires bursts of 30 laser shots at rocks. Its microphone records the shock wave generated between 2 and 15 kHz by each laser-induced plasma for distances between 2 m and 8 m, at varying local times, from strong thermal turbulence during daytime, to weaker one at dusk or dawn. In each case, the scintillation index (the normalized variance of the intensity), and travel time are calculated. Based on extensive literature on the propagation of a spherical wave through a turbulent medium, there are enough statistics to constrain the field that causes the dispersion of acoustic data. Initial results show that data are well modeled by a Gaussian field. A negative time shift, $\sim$0.5% at 6 m, is observed: the resulting velocity is higher than the mean speed of sound (fast path effect). The variance of arrival times leads to a 15 cm characteristic scale during daytime. Fluctuations of the scintillation index allow the strength of the turbulence field to constrained. Other turbulence models are tested, including the Kolmogorov energy-cascade which typically prevails on Earth for the inertial regime, but has been challenged several times on Mars.

Contributed Papers

10:45

4aPAa8. N-waves propagating through the Mars atmosphere. Baptiste Chide (Space and Planetary Exploration Team, Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, bchide@lanl.gov), Sylvestre Maurice (IRAP, Toulouse, France), Xavier Jacob (IMFT, Toulouse, France), Ralph Lorenz (JHU Appl. Phys. Lab, Laurel, MD), Roger Wiens (Purdue Univ., Lafayette, IN), Philippe Blanc-Benon (CNRS, Ecully, France), Naomi Murdoch, David Mimoun (ISAE, Toulouse, France), and Nina Lanza (Space and Planetary Exploration Team, Los Alamos National Lab., Los Alamos, NM)

One of the particular interest of the SuperCam microphone onboard the NASA's Perseverance rover is that it is coupled with its own reproducible sound source, the blast waves generated by the SuperCam laser-induced plasma expansion. This unique emission-reception pair on Mars is used to investigate the propagation of spark-induced N-waves through the turbulent CO_2-dominated Mars atmosphere. The analysis of waveforms shows large distortions over a burst of 30 subsequent shots, which is consistent with previous laboratory recordings of N-waves through thermal turbulence. Especially, this distortion leads to a scattering of the peak pressure amplitude, whose distribution can be fitted with a generalized gamma function. For bursts recorded farther than 4 m, the distribution gets asymmetric with a shift of the maximum toward the left of the median due to peaks amplified more than twice the median amplitude, likely due to caustics that randomly focus the acoustic wave. On the other hand, over a Martian year, the absolute peak pressure varies by $\sim$50% due to changes in the acoustic impedance and Rankine-Hugoniot conditions, due to the seasonal cycles of pressure and temperature at the surface of Mars.

11:00

4aPAa9. What might we hear on other worlds? A free demonstration programme for planetaria. Timothy G. Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk) and Phil Joseph (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom)

In order to plan missions with acoustical sensors for other worlds, and to design those instruments appropriately, it is useful to predict the nature of the signals they might detect. Such predictions could, for example, identify in advance that a planned single microphone system should be replaced by a 3-microphone system, because the predictions indicate the dominance of uncorrelated non-acoustic (hydrodynamic) pressure fluctuations that need to be separated from the incident sound, allowing (in addition) its the direction of arrival to be estimated. Similarly, calibration errors could also be predicted resulting from the fluid loading experienced by the off-world microphone, or acoustical artifacts resulting from the presence of coupled waves in the structure supporting the microphone. One offshoot of generating such predictions was the ability to build a software package to provide to planetaria the ability to predict certain environmental sounds on other worlds and entertain the public by changing their voices to reflect how they would sound on another world, if they could speak and live. The interest of the public has always been key to ensuring policymaker support for space and extraplanetary exploration. This should be supported by rigorous science, or where guesses need to be made, these should be transparently communicated. This project endeavoured to do this through the aforementioned software package.

11:15

4aPAa10. Seismo-acoustic coupling efficiency at the surface of Venus. Gil Averbuch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Onset, MA), Reyna Houston (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), 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 extreme conditions at the surface of Venus pose a challenge for monitoring the planet's seismic activity using long-duration landed probes. One alternative is using balloon-based sensors to detect venusquakes from the atmosphere. This presentation assesses the efficiency of seismic-to-acoustic energy transfer from Venus' crust to its deep atmosphere. It is, therefore, restricted to immediate neighborhood of the planet's surface. In order to account for supercritical conditions near the surface, the Peng-Robinson equation of state is used to obtain the acoustic wavenumber in the lower atmosphere. The energy transported across the surface from deep and shallow sources is two orders of magnitude larger than on Earth, pointing to a very strong seismo-acoustic coupling. For a more realistic scenario, we simulated the acoustic field generated in the lower atmosphere by the ground motion arising from a vertical array of subsurface point-force sources. The resulting transmission loss maps show a strong epicentral cone accompanied by contributions from leaky surface waves.

11:30

4aPAa11. Acoustics on the planet Mars: Next steps, including sonic booms. Lily R. Hetherington (Penn State, Grad. Program in Acoust., 201 Appl. Sci. Bldg., University Park, PA 16802, lrh5483@psu.edu), Joshua L. Kapcsos (Penn State, University Park, PA), and Victor W. Sparrow (Penn State, University Park, PA)

We congratulate the Perseverance rover team for measuring a dust devil and other sounds on Mars. This is a great achievement, and at the same time just a beginning. The acoustics community needs to apply itself to the situations that might come up when humans visit Mars. Work in the late 1990s and early 2000s set the stage for our knowledge of the outdoor sound environment of Mars. One next step will be to understand how shock waves and sonic booms will propagate on Mars. Because of the high atmospheric absorption, the shock structures of sonic booms are expected to differ on Mars compared to those on Earth. Some of the differences in potentially using a program like NASA's PCBoom suite of tools will be discussed. Sonic booms have been used in the past to probe the atmosphere on Earth, and the prospects seem bright for successful similar applications on Mars. [Work supported by the Penn State College of Engineering.]

11:45

4aPAa12. Considerations of undersea exploration of an extraterrestrial ocean. Grant Eastland (Test and Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

Hypothetical investigations and simulations of the potential effects to acoustic propagation and modal structures in oceans on extraterrestrial planets and moons are presented. Europa, one of the many moons of Jupiter is thought to have a water-based ocean that has a depth of 60 to 150 km and covered by a suspected 15 to 25 km thick ice sheet. Another moon of Jupiter, Ganymede, is the largest moon in our solar system is thought to have an underground saltwater sea 100 km deep. Titan, a moon of Saturn, is also thought to have a water-based sea under an ice sheet in addition to surface, rivers, lakes, and seas of hydrocarbons like methane. Exploration of such exotic environments will require great leaps in technology and a fundamental understanding of the physics of the environment. Understanding the acoustics of such environments will provide the mechanism to develop the advanced sensors needed for the exploration missions. The drive to explore such places could provide evidence of life beyond Earth, since where there is water, life is possible.

Session 4aPAb

Physical Acoustics: General Physical Acoustics I: Time Reversal Technique and Source Characterization

Brian E. Anderson, Cochair
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Sarah McComas, Cochair
3909 Halls Ferry Rd., Vicksburg, MS 39180

Contributed Papers

9:00

4aPAb1. A time reversal metasurface for mimicking the cocktail party effect. Constant Bourdeloux (Institut Langevin - ESPCI Paris - Université PSL, 1 rue Jussieu, Paris 75005, France, constant.bourdeloux@espci.psl.eu), Fabrice Lemoult, and Mathias Fink (Institut Langevin - ESPCI Paris - Université PSL, Paris, France)

The cocktail party effect is the capability to focus one's auditory attention on particular audio sources while ignoring other audio sources. We propose an experimental setup reproducing the cocktail party effect by designing a time dependent metasurface composed of independent active mirrors. Each active mirror is a programmable acoustical unit cell capable of hearing, computing, and emitting acoustic signals: each of them acts as a convolution filter. The proper metasurface temporal filters configuration allows us to establish acoustic communication between groups of individuals immersed in a noisy environment: a MU-MIMO acoustic system. The experiment consists in recording a set of Green's function between N_e emitters and N_j active mirrors, and a second set of Green's functions between the N_j active mirrors and N_r receivers. To increase the spatio-temporal degrees of freedom (Lemoult *et al.*, PRL **103** 2009), we place a forest of steel rods between the active mirrors, the emitters, and the receivers. These data are then used to compute each active mirror's temporal filter using time reversal properties to establish a predefined MU-MIMO configuration. The N_e emitters now emits simultaneously uncorrelated noises. These signals propagate through the disordered medium to the metasurface, then each active mirror hears, convolutes with his filter and emits another signal, which propagates back through the disordered medium towards the N_r receivers. In the end, we establish a correlation matrix by comparing emitted and received signals.

9:15

4aPAb2. Blowing out a candle from a distance with high amplitude time reversal. Jay M. Clift (Phys. & Astronomy, Brigham Young Univ., Provo, UT 84602, j@clift.org), Brian D. Patchett (Phys. & Astronomy, Brigham Young Univ., Orem, UT), and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal is a technique that allows the focusing of sound from a source to a remote location causing an impulsive focusing event. Using more sources and other amplitude-increasing techniques, the focusing of audible sound can reach over 200 dB peak. This can extinguish the flame on a single candle without affecting nearby lit candles. Most methods used to extinguish a flame using sound involve the nearfield and nonlinear airflow effects. Because time reversal focusing can occur far from the sources, the mechanism is different. Using high-speed video footage of the focal event, we can see the finer details of this millisecond scale focus interacting with the flame. Measuring the sound field during the focus event at and near the focal location also sheds light on the mechanism. So far, it has been seen that the focus pushes the flame outward, separating the plasma from its fuel source. The known formation of shock waves provides a possible explanation. Further research is being done to explore the cause of the extinguishing of the flame. This talk will show that not only is it possible to focus high amplitude sound waves but targeted flows of air as well.

9:30

4aPAb3. Modeling free space Mach stem formation in high-amplitude focusing of sound using time reversal. Brian D. Patchett (Phys., Utah Valley Univ. and Brigham Young Univ., 800 W University Pkwy, MS-179, Orem, UT 84058, brian.d.patchett@gmail.com), Brian E. Anderson, and Adam D. Kingsley (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

In acoustics, time reversal processing may be used to focus sound to a selected spatial location. Recently, the nonlinear characteristics of time-reversal focusing at amplitudes as high as 200 dB have been reported [Patchett and Anderson, *J. Acoust. Soc. Am.*, **151**(6), (2022)]. These studies were experimental in nature and suggested that converging waves nonlinearly interact in the focusing of waves, leading to surprising observations of nonlinear amplification. This study investigates the nonlinear interactions and subsequent characteristics from a model-based approach. Utilizing both the k-Wave© package for MATLAB©, and COMSOL Multiphysics©, it is shown that nonlinear interactions between high-amplitude waves leads to free-space Mach-wave coalescence of the converging waves. The number of waves used in both models represents a small piece of the full aperture of converging waves experimentally. Limiting the number of waves limits the number of Mach-stem formations and reduces the nonlinear growth of the focus amplitudes when compared to experiment. However, limiting the number of waves allows the identification of individual Mach waves. Mach wave coalescence leading to Mach-stem formation appears to be the mechanism behind nonlinear amplification of peak focus amplitudes observed in high amplitude time reversal focusing.

9:45

4aPAb4. Sounds produced by subsonic ballistic flow. Steven D. Beck (Beck Audio Forensics, 14101 Hwy. 290 West, Bldg. 1700, Ste. A, Austin, TX 78737, stevendbeck@alumni.rice.edu) and Norman W. Todd (Dept. of Otolaryngol. – Head and Neck Surgery, Emory Univ., Atlanta, GA)

The sounds of passing subsonic ballistic projectiles are receiving more attention yet have limited acoustic characterization. Recent accounts include buzzing sound of a rifled bullet, and the whistling sounds of spherical projectiles produced by police firing non-lethal spherical rubber bullets, paintball, and war re-enactments using muskets. Modern recording equipment with multiple simultaneous channels and high dynamic range is now available to accurately capture subsonic ballistic flow sounds. Spherical projectiles 0.69 in. in diameter were launched from smooth bore replica muskets and high pressure paintball guns, and recorded on multiple microphones. Sounds of vortex shedding at predictable frequencies were produced using

smooth spherical balls with no rotation, fired at velocities between 45 and 152 m/s (150–500 fps), and with Reynolds numbers below 3.75×10^5. The paintballs shed vortices in the 500–800 Hz range (with reported whoosh sounds); while faster musket balls shed vortices in the 2000–2500 Hz range (with reported whistling sounds). Our interest originally arose from historical reports that Erastus "Deaf" Smith, a hero of the Texas Revolution, could not hear musket balls that passed near him; and recent medical speculation that he had a 2000–3000 Hz notch in his hearing spectrum.

10:00

4aPAb5. Noise characterization of unmanned ground vehicles for infrasound array deployment. James Kinnebrew (Information Technol. Lab., US ARMY ERDC, 3909 Halls Ferry Rd., Vicksburg, MS 39180, james.s.kinnebrew@erdc.dren.mil) and Sarah McComas (Geotechnical and Structures Lab., US ARMY ERDC, Vicksburg, MS)

Deployment of infrasound arrays can present humans with challenges including harsh environments and dangerous chemicals. Infrasonic monitoring is the use of subaudible acoustics to monitor something of interest. Sources of infrasound include natural and manmade sources such as volcanic eruptions, large forest fires, and explosions. Robotic infrasound array deployment is being investigated to mitigate the potential challenges associated with human deployment. Robotic deployment would also allow for remote sensor repositioning to improve array response for specific sources. Unfortunately, robotic sensor emplacement will induce noise associated with the robotic platform, and this noise has a potential to mask sources of interest under certain conditions. Understanding to which conditions robotic deployment is applicable is vital to exploring infrasonic monitoring in hazardous environments in the future. This presentation will describe the characterization of noise generated by multiple robotic platforms in both mobile and stationary states. Analysis during each of these states allows for the determination of when robotic mounted infrasonic monitoring is viable.

10:15–10:30 Break

10:30

4aPAb6. A lower frequency signal emitted from tornadoes based on a new mechanism. Bin Liang (Univ. of MS, 145 Hill Dr., Oxford, MS 38677-1848, bliang@go.olemiss.edu), Roger M. Waxler (Univ. of MS, University, MS), and Paul Markowski (Penn State Univ., University Park, PA)

Tornadoes have been shown to radiate infrasound to great distances. After using Lighthill's acoustical analogy to study sound generated by a numerical tornado, we found that there is a significant low-frequency signal between 0.1 Hz and 0.5 Hz. We hypothesized that there is a Kirchhoff vortex-like source at the center of the numerical tornado. Based on vortex sound theory, characteristic frequencies only depend on the strength of the vertical vorticity which can change at different heights of the tornado. Compared to real data analysis, there is a possibility that when a tornado occurs, infrasonic sensors can detect a significant increase in low-frequency signals. This hypothesis is being tested against data collected in the field during the passage of tornado-producing storms.

10:45

4aPAb7. Theoretical investigation on the moving flame–sound interaction in a closed-open combustor. Yiheng Guan (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand) and Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, dan.zhao@canterbury.ac.nz)

Self-excited thermoacoustic instability is highly undesirable for power generation gas turbines, aero-engine afterburners, liquid-fuelled ramjet, and rocket motors. It is typically generated due to the constructive interaction between acoustic perturbations and the flame. It is a general practice to assume that the flame is non-moving and the combustor is typically assumed to be acoustically opened. In this work, we consider a closed-open thermoacoustic combustor with a moving flame. To better understand the physics between the acoustic disturbances and the moving flame, we theoretically investigate the moving flame-sound interaction. We find that the presence of the moving flame can amplitude the gaseous oscillations. Furthermore, the amplification process is shown to depend on the laminar burning velocity on the temperature and density of the combustible gas mixture into which the flame propagates. The sound source is shown to be the quadrupole mode. It radiates very little energy, if the flame dimensions are smaller than the wavelength of the sound produced. The present work shed lights on the sound generation from a moving flame and its interaction with sound.

11:00

4aPAb8. Multi-mode instability interactions in twin jet configurations. Aatresh Karnam (Aerosp. Eng. & Eng. Mech., Univ. of Cincinnati, 2901 Woodside Dr., Cincinnati, OH 45219, karnamah@mail.uc.edu), Mohammad Saleem, and Ephraim Gutmark (Aerosp. Eng. & Eng. Mech., Univ. of Cincinnati, Cincinnati, OH)

Twin jet configurations are prone to a wide range of interactions due to flow instabilities driven by jet self-excitation and cross-excitation of one jet on another. These instability modes and associated phases are sensitive to many parameters chief among which is the exit profile of the nozzle. Extensive studies on Twin Circular nozzles and Large Aspect Ratio (AR) Twin rectangular nozzles have shown various mode coupling but, not much is known about twin jets with moderate aspect ratios ($AR < 3$). This study aims to understand the effect of the exit nozzle geometry on the generation of flow instabilities by comparing a Twin Rectangular nozzle (AR 2) to a Twin Square nozzle (AR 1) with identical design parameters through experimental analysis. High speed Schlieren imaging is coupled with Particle Image Velocimetry (PIV) to explore both qualitative (internal shock cell structure) and quantitative aspects (flow vorticity and turbulence) of these jets. Near field and far field acoustics are also studied to provide a wholistic understanding of the effect of mode coupling exhibited by these jets. Generation and propagation mechanisms of multi-mode instabilities which effect both nozzle orientations are also studied through Spatial and Temporal Fourier analysis.

11:15

4aPAb9. Effect of internal geometry on supersonic jet noise. Kaurab Gautam (Aerosp. Eng. and Eng. Mech., Univ. of Cincinnati, 688 Riddle Rd., Apt. 1400M, Cincinnati, OH 45220, gautamkr@mail.uc.edu), Aatresh Karnam (Aerosp. Eng. or Eng. Mech., Univ. of Cincinnati, Cincinnati, OH), Mohammad Saleem (Univ. of Cincinnati, Cincinnati, OH), and Ephraim Gutmark (Aerosp. Eng. and Eng. Mech., Univ. of Cincinnati, Cincinnati, OH)

The current study discusses the experimental results of the effect of two types of nozzle internal geometries. The first configuration uses a common inlet that diverges the flow into two separate channels that feed twin rectangular nozzles, and the second configuration has the plug flow type system with twin rectangular nozzles drawing air from a common plenum. The nozzles are converging diverging type with design Mach number of 1.5. Flow conditions encompassing overexpanded, design, and underexpanded conditions are tested with acoustic data collected in the farfield and nearfield domains. High-speed Schlieren imaging is used to visualize the effect of internal geometry on the development of the jet shock cell structure and twin jet interaction. The variations in the Overall Sound Pressure Level (OASPL), frequency spectra, and jet phase coupling are used to quantify the effect of the internal geometries. This is supplemented by Spectral Proper Orthogonal Decomposition (SPOD) results obtained from the Schlieren images to quantify the variations in flow development.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Perception Beyond Tones and Speech in Normal and Impaired Hearing: Voice, Emotions, and Music Perception

Deniz Baskent, Cochair
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Tyler K. Perrachione, Cochair
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Virginia Best, Cochair
Speech, Language and Hearing Sciences, Boston University, 635 Commonwealth Ave., Boston, MA 02215

Chair's Introduction—8:00

Invited Papers

8:05

4aPP1. Voice cue sensitivity in children with cochlear implants and with hearing aids. Laura Rachman (Pento Audiol. Ctr., Hanzeplein 1, Groningen 9713 GZ, the Netherlands, l.rachman@rug.nl), Gizem Babaoğlu (Audiol. Dept., Hacettepe Univ., Ankara, Turkey), Leanne Nagels (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Groningen, the Netherlands), Pınar Ertürk, Başak Yazgan (Audiol. Dept., Hacettepe Univ., Ankara, Turkey), Debi Vickers (Cambridge Hearing Group, Clinical Neurosciences Dept., Univ. of Cambridge, Cambridge, United Kingdom), Petra Hendriks (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Groningen, the Netherlands), Gonca Sennaroğlu (Audiol. Dept., Hacettepe Univ., Ankara, Turkey), Etienne Gaudrain (Lyon Neurosci. Res. Ctr., CNRS UMR5291, Inserm U1028, UCBL, UJM, Lyon, France), and Deniz Başkent (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, the Netherlands)

Voice cues, such as fundamental frequency (F0) and vocal-tract length (VTL), allow listeners to distinguish speakers, which can facilitate speech perception in challenging listening conditions. Children with typical hearing continue to develop their sensitivity to F0 and VTL differences throughout childhood, but this is not as clear for children with hearing loss. In prelingually deaf implanted children with cochlear implants (CIs), reduced spectrotemporal details may cause a delay or even a plateau in the development of voice cue sensitivity, while neuroplasticity may compensate for these factors. For children with hearing aids, it is unclear how voice cue perception could be affected by combined factors of neuroplasticity, physiological aspects of hearing loss, and compensatory front-end processing of hearing aids. Here, we will present recent work on F0 and VTL sensitivity in children with CIs and with hearing aids (4–18 years). Our results show a large variability in both CI and hearing-aided children, with some children performing at the level of age-matched normal-hearing children, while others performing lower. Both groups show trends for development as a function of age, but these seem to differ for F0 and VTL. Potential factors that may be contributing to these results will be discussed during the presentation.

8:25

4aPP2. The underlying mechanisms for voice discrimination across the life span. Yael Zaltz (Dept. of Commun. Disord., Steyer School of Health Professions, Sackler Faculty of Medicine, Tel Aviv Univ., Ramat Aviv, Tel Aviv 6997801, Israel, yaelzalt@tauex.tau.ac.il)

Speech understanding in noisy conditions changes across the life span. One strategy that assists listening in multi-talker noise is to track the voice of the relevant speaker, based on his/her fundamental frequency (F0) and formant frequencies (reflecting vocal tract length). Thus, age-related differences in voice discrimination (VD) may explain some of the difficulties that children or older-adults have in speech-in-noise perception. The present experiments aimed to assess VD from childhood to old age and explore the underlying mechanisms for VD across ages. Four experiments assessed difference limens (DLs) for VD, using an oddball paradigm with two-down one-up adaptive procedure in quiet conditions (Two experiments, including 41 children, 36 young-adults, and 15 older-adults), amid speech-shaped noise (16 children, 24 young-adults, and 12 older-adults) and following different testing methods (32 children, and 31 young-adults). Working memory, attention, and speed of processing were also assessed. The poorest VD was shown for the youngest children and the older-adults, with better cognitive abilities and/or improved acoustic accessibility/resolution advancing VD for these groups. children and older-adults may engage more top-down (cognitive) and bottom-up (periphery) resources for VD compared to young-adults, which may partly explain their difficulties in understanding speech in multi-talker environments.

8:45

4aPP3. Examination of audiovisual prosody in cochlear implant recipients. Hartmut Meister (Univ. of Cologne, Geibelstr. 29-31, Cologne 50931, Germany, hartmut.meister@uni-koeln.de), Isa Winter, Moritz Waechtler, Pasale Sandmann, and Khaled Abdellatif (Univ. of Cologne, Cologne, Germany)

Prosody plays a vital role in verbal communication. It is important for the expression of emotions but also carries information on sentence stress or the distinction between questions and statements. Cochlear Implant (CI) recipients are restricted in the use of acoustic prosody cues, especially in terms of the voice fundamental frequency. However, prosody is also perceived visually, as head and facial movements accompany the vocal expression. To date, few studies have addressed multimodal prosody perception in CI users. Controlled manipulations of acoustic cues are a valuable method to uncover and quantify prosody perception. For visual prosody, however, such a technique is more complicated. We describe a novel approach based on animations via virtual humans. Such a method has the advantage that–in parallel to acoustic manipulations–head and facial movements can be parametrized. It is shown that animations based on a virtual human generally provide similar motion cues as video recordings of a real talker. Parametrization yields fine-grained manipulation of visual prosody, which can be combined with modifications of acoustic features. This allows generating both congruent and incongruent stimuli with different salience. Initial results of using this method with CI recipients are presented and discussed.

9:05

4aPP4. Effects of age and hearing loss on talker identification and talker change detection. Virginia Best (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu), Jayne B. Ahlstrom (Medical Univ. of South Carolina, Charleston, SC), Christine R. Mason, Tyler K. Perrachione, Gerald Kidd (Boston Univ., Boston, MA), and Judy R. Dubno (Medical Univ. of South Carolina, Charleston, SC)

While the detrimental effects of age and hearing loss on word recognition and speech comprehension are well-established, surprisingly little is known about how these factors affect the ability to extract talker-related information from speech. Here, we describe two studies that used different behavioral tasks to address this question. In the first study (Talker Identification), listeners were trained to identify talkers by the sound of their voice, and then tested on their ability to identify those talkers in quiet or in the presence of background noise or competing speech. In the second study (Talker Change Detection), explicit talker identification was not required, but instead listeners detected if a change in talker had occurred within a sequence of words. In both studies, listeners were recruited in four groups to include younger/older ages and normal/impaired hearing. In both studies, we found that age and hearing loss had independent effects on performance. We argue that a reduced ability to make use of talker-related information in speech may contribute to the difficulty encountered by older listeners and listeners with hearing loss, when communicating in group situations.

9:25

4aPP5. Effects of audible bandwidth and stimulus modality on emotional responses to sound. Erin M. Picou (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, erin.picou@vanderbilt.edu)

A wide range of emotional responses is typical and important for overall well-being. Adults with hearing loss demonstrate a reduced range of emotional responses to non-speech sounds; their ratings of valence to pleasant and unpleasant sounds are less extreme than those of their peers with normal hearing. The purpose of this project was to explore the role of two potential explanations, limited audible bandwidth and emotional processing differences. Adults with normal hearing and sensorineural hearing loss rated their emotional responses (valence and arousal) to non-speech, affective sounds and pictures (in combination and in isolation). To explore the potential contribution of audible bandwidth, adults with normal hearing rated stimuli that were full bandwidth and also bandpass limited. In addition, adults with hearing loss were specifically recruited to include heterogeneous hearing loss configurations. Differences in ratings between auditory and visual stimuli provide insight into whether reduced emotional responses for adults with hearing loss are specific to audition or generalize to visual stimuli. Results demonstrate the importance of audition and audible bandwidth as key factors in emotional responses to pleasant, non-speech stimuli. These findings have important implications for designing interventions to support the listening needs of adults with hearing loss.

9:45

4aPP6. Utilizing auditory emotion bio-markers, the underpinning of emotion perception improvement in cochlear implant users. Sebastien Paquette (RI-MUHC, McGill, 2900 Edouard Montpetit Blvd, Montreal, QC H3T 1J4, Canada, sebastien.paquette.1@umontreal.ca), Samir Gouin (Dept. of Otolaryngol., McGill, Montreal, QC, Canada), and Alexandre Lehmann (RI-MUHC, McGill, Montreal, QC, Canada)

Cochlear implants (CI) have had tremendous success restoring a sense of hearing in the deaf. However, even after months of intensive rehabilitation, many CI users struggle with appreciating emotive tones in speech and music despite good speech comprehension. Failure to perceive emotional expression can result in maladjusted social behaviour, leading to detrimental socio-economic consequences. Recent advances in automated pattern identification of neuroimaging data can bring empirical support to developing training programs for emotion perception rehabilitation in CI users. We used a machine-learning approach to identify emotion-processing biomarkers in high-density electroencephalograms collected from CI users (22) and matched normal-hearing controls (22). Participants' brain responses elicited by short musical and vocal emotional (happy, sad, and neutral) stimuli were used to train an algorithm to help identify, in each group, the pattern of brain responses that can best predict the presented emotion. Using this approach, we were able to confirm the presence of emotion-specific patterns of brain activity in CI users despite their reported emotion perception deficit. Identifying these patterns brings forward support for implementing a rehabilitation program for emotion perception for this population; if an algorithm can differentiate aurally presented emotions, perhaps CI users can learn to discriminate emotions.

10:05–10:25 Break

10:25

4aPP7. Age and cognitive status as predictors of emotional prosody identification by children and adults with normal hearing or cochlear implants. Monita Chatterjee (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, monita.chatterjee@boystown.org)

In everyday communication, prosody (i.e., the tone and manner of speaking) conveys primary information about a talker's intended emotion. Across the lifespan, emotional prosody identification shows age effects (i.e., development in children and aging in adults) in listeners with normal acoustic hearing as well as in listeners with cochlear implants. Cognitive resources are likely harnessed in the process of mapping acoustic cues to the correct emotion, particularly when the acoustic cues to prosody are degraded as in cochlear implants. We are investigating the dual effects of age and cognition on identification of emotional prosody by school-age child and adult participants with normal hearing or cochlear implants. In school-age children with normal hearing, we observe improved performance with age. In children with cochlear implants, hearing age (years of experience with the device, also correlated with chronological age) is also a significant predictor, along with an additional, interactive effect of nonverbal cognition. In adults with normal hearing, we observe negative effects of age together with a positive effect of working memory for both clean and degraded (cochlear implant simulated) speech. These findings have implications for the processes underlying emotion perception across the lifespan in individuals with normal hearing and with cochlear implants.

10:45

4aPP8. Perception of emotional music by children with cochlear implants reveals developmental plasticity. Karen A. Gordon (The Hospital for Sick Children, Rm 6D08, 555 University Ave., Toronto, ON M5g 1x8, Canada, karen.gordon@utoronto.ca)

Cochlear implants provide poor access to the frequency components needed to accurately identify emotional music and speech. Nonetheless, a series of studies revealed that children with cochlear implants develop unique strategies to identify emotion in music. Although musical changes were difficult for them to detect, children with cochlear implants responded most accurately to rhythmic changes and also relied mainly on temporal information to judge whether music was happy or sad. By contrast, mode (frequency) information dominated perception of musical emotion in typically developing children and was used more clearly by bimodal device users (cochlear implant in one ear with acoustic hearing in the other) than children with bilateral deafness using unilateral or bilateral cochlear implants. Mode became the more dominant cue with increasing residual hearing in the non-implanted ear. Children with unilateral cochlear implants showed prolonged response times relative to a control group of normal hearing peers while judging whether music was happy or sad. Response times in children with bimodal devices and bilateral cochlear implants were more similar to controls. Together, these results demonstrate development of novel strategies for music listening and perception of emotion in music based on available cues in children with cochlear implants and suggest that these strategies require cognitive resource.

11:05

4aPP9. Exploring the impact of hearing loss through music, irony, and rhythm. Chi Yhun Lo (Toronto Metropolitan Univ., JOR 905, Jorgenson Hall, 380 Victoria St., Toronto, ON M5B 0A1, Canada, chi.lo@ryerson.ca), Frank Russo (Toronto Metropolitan Univ., Toronto, ON, Canada), and Gurjit Singh (Phonak Canada, Mississauga, ON, Canada)

Hearing loss is associated with challenges such as listening in noise, misinterpreting emotion, distorted music perception, and poorer quality of life outcomes relative to hearing peers. Despite this, research examining perceptual abilities for individuals with hearing loss has overwhelmingly focussed on measuring a narrow representation of speech perception. These typically include tasks ranging from the perception of simple words-in-quiet to more complex speech-in-noise. However, the full range of human communication extends far beyond this. In this talk, we will present several studies exploring topics that extend beyond traditional and contemporary approaches to perception. First, a music training study for children with hearing loss that enhanced both speech and psychosocial outcomes. Second, a paralinguistic study examining irony comprehension in children with hearing loss. Finally, an examination of rhythm and speech correlates for older adults with hearing aids, and how this can inform the focus of future intervention studies. Findings from these studies underscore the importance of extending our focus beyond traditional perceptual outcomes.

Contributed Papers

11:25

4aPP10. Orthographic information facilitates discrimination of native and foreign-accented speech. Victoria Sevich (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, sevich.1@osu.edu), Emily M. Clausing (Dept. of Otolaryngol. - Head and Neck Surgery, The Ohio State Univ. Wexner Medical Ctr., Columbus, OH), Aaron C. Moberly, and Terrin N. Tamati (Dept. of Otolaryngol. - Head and Neck Surgery, Vanderbilt Univ. Medical Ctr., Nashville, TN)

Previous research on the perception of foreign-accented speech has demonstrated that providing orthographic transcriptions of auditory stimulus words influences accentedness ratings of the talker. Specifically, foreign-accented talkers are rated as having less accented speech in favorable listening conditions when orthographic information is present. Cochlear implant users show poor identification and discrimination of foreign-accented speech relative to listeners with normal hearing. In the current study, we examined whether orthographic information facilitates the discrimination of native and foreign-accented (non-native) speech under cochlear-implant simulation. Participants were presented with native or foreign-accented sentences through an eight-channel noise vocoder with or without matching text. They were asked to determine whether the sentence was spoken by a native or non-native talker. We hypothesized that the presence of matching text would increase identification accuracy and decrease reaction time. Consistent with our predictions, participants more accurately and more quickly identified whether sentences were spoken by a non-native talker in the presence of matching text compared to performance in conditions with no text. These results suggest that the presence of orthographic information may facilitate indexical, non-linguistic perception under cochlear-implant simulation. Future work will determine whether these results may extend to cochlear implant users.

11:40

4aPP11. Arousal but not valence: Music emotion categorization in normal hearing and cochlear implanted participants. Eleanor Harding (Otorhinolaryngology, UMCG, Groningen, the Netherlands, e.e.harding@rug.nl), Etienne Gaudrain (CNRS, Lyon, France), Imke Hrycyk (Otorhinolaryngology, UMCG, Groningen, the Netherlands), Robert Harris (Prince Claus Conservatory, Hanze Univ. of Appl. Sci., Groningen, the Netherlands), Barbara Tillmann (CNRS, Lyon, France), Bert Maat, Rolien Free, and Deniz Başkent (Otorhinolaryngology, UMCG, Groningen, the Netherlands)

Perceiving acoustic cues that convey music emotion is challenging for cochlear implant (CI) users. Emotional arousal (stimulating/relaxing) can be conveyed by temporal cues such as tempo, while emotional valence (positive/negative) can be conveyed by spectral information salient to pitch and harmony. It is however unclear the extent to which other temporal and spectral features convey emotional arousal and valence in music, respectively. In 23 normal-hearing participants, we varied the quality of temporal and spectral content using vocoders during a music emotion categorization task—musical excerpts conveyed joy (high arousal high valence), fear (high arousal low valence), serenity (low arousal high valence), and sorrow (low arousal low valence). Vocoder carriers (sinewave/noise) primarily modulated temporal information, and filter orders (low/high) primarily modulated spectral information. Improvement of temporal- (using sinewave carriers) and spectral content (using high filter order) both improved categorization. Vocoder results were compared to data from 25 CI users performing the same task with non-vocoded musical excerpts. The CI user data showed a similar pattern of errors as observed for the vocoded conditions in normal-hearing participants, suggesting that increasing the quality of temporal information, and not only spectral details, could prove beneficial for CI users' music emotion perception.

THURSDAY MORNING, 11 MAY 2023 CHICAGO H, 10:00 A.M. TO 12:00 NOON

Session 4aSA

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

Hubert S. Hall, Cochair
Mechanical Engineering, Texas Christian University, 2840 W Bowie St., Fort Worth, TX 76109

Contributed Papers

10:00

4aSA1. Multi-fidelity surrogate modeling for structural acoustics applications. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, anthony.l.bonomo.civ@us.navy.mil)

Recently, surrogate modeling methods have been explored for structural acoustics applications. These often involve evaluation of an "expensive" high-fidelity computational model to obtain training data. However, in many applications, models of varying fidelity and computational cost are available. In such situations, one can leverage multi-fidelity surrogate modeling, where the training data from models of varying fidelity are combined and simultaneously used to produce a surrogate model. A particularly popular class of multi-fidelity surrogate modeling techniques is known as co-Kriging, where simulation output from both "expensive" and "cheap" computational models are correlated and a correction process is obtained that maps between the results of these models of varying fidelity. This talk will review co-Kriging and demonstrate its utility on a canonical structural acoustics problem. [Work supported by the Office of Naval Research.]

10:15

4aSA2. Investigation of the radiated sound energy from noise sources using an indirect vibration-based sound power approach. Ian C. Bacon (Dept. of Phys. & Astronomy, Brigham Young Univ., N269 ESC, Provo, UT 84602, ianbacon24@gmail.com), Naomi M. Jensen, Scott D. Sommerfeldt (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

An indirect vibration-based sound power (I-VBSP) method is being developed for acoustic sources radiating energy that cannot be captured using surface vibration measurements. This method involves placing this type of source into a small enclosure that has a single flexible side which the noise source will excite. The vibration-induced side can then be scanned using a scanning laser Doppler vibrometer (SLDV) and calibrated to obtain the free-field energy of each source. Several challenges have been identified during the development of this method, many of which have been addressed to improve the accuracy of calibrating out the effects of the enclosure on the radiation from these noise sources and improve the SLDV measurements. A

4a THU. AM

box-in-box structure was fabricated with a flexible panel to investigate the acoustic response of this new enclosure versus the previous mylar membrane enclosure. Results for the sound power of a source obtained through the mylar and panel structures using the I-VBSP method will be compared with the free-field sound power of the source determined in a reverberation chamber using the ISO 3741 standard. [Funding for this work was provided by the National Science Foundation (NSF).]

10:30

4aSA3. The dependence of sound radiation on position of acoustic source in an enclosure. Naomi M. Jensen (Brigham Young Univ., N269 ESC, Provo, UT 84602, naomi.michelle29@gmail.com), Ian C. Bacon (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT)

In some applications, acoustic sources may be confined to a small enclosure but still radiating sound outwards through the enclosure. However, the sound power that is radiated from the enclosure may potentially be impacted by the location of the source within the enclosure and the properties of the enclosure. The dependence of source position on sound power radiated from the enclosure was investigated using a small rigid rectangular enclosure with a flexible aluminum panel as one of the sides of the enclosure. An acoustic source was moved to numerous locations in the enclosure and sound power measurements were made using the ISO 3741 standard. Results will be shown to numerically quantify the effect of acoustic source position on excitation of the aluminum sheet and radiated sound power. These results are used to develop a calibration curve between the sound radiate by the enclosed source versus the sound radiated by the source in a free field environment. This curve can be used in the development of an alternative method of determining sound power for a source.

10:45

4aSA4. Modal testing on a limited budget: Analysis of instrumented hammer alternatives for impact testing. Hubert S. Hall (Eng., Texas Christian Univ., 1701 W. Marshall Dr., Grand Prairie, TX 75051, H.HALL@tcu.edu) and Curtis Larsen (Eng., Texas Christian Univ., Fort Worth, TX)

In recent years, as the use of numerical modeling has increased, organizations have frequently scaled back their experimental capabilities. Often equipment and expertise are no longer available when modal correlation measurements are required. This presentation is part of a planned series examining a minimalist approach to modal testing. In this presentation, alternatives to a traditional instrumented hammer for impact testing are explored. Comparison testing between a dedicated instrumented hammer procured from a leading structural measurement company and commercial hammers acquired from a hardware store was performed. Two cases of commercial hammers were studied, one featuring a force gauge inserted into the impact hammer face and one with an accelerometer attached to the hammer body. A variety of hammer tips were explored for focused frequency content. Follow-on topics include rudimentary data acquisition options and methods of modal data analysis outside of expensive dedicated modal analysis software platforms.

11:00

4aSA5. Improving transmission through honeycomb panels through modal considerations. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov), Charles M. Reinke, and Anup Parikh (Sandia National Labs., Albuquerque, NM)

Recent efforts centered at Sandia National Laboratories have demonstrated transmission of power and data through continuous metal media using vibration in connection with piezoelectric tiles; however, solid metal construction may be disadvantageous in structural contexts due to the greater mass needed to achieve given stiffness characteristics. Other common structural materials, on the other hand, such as aluminum honeycomb panels, may offer poor transmission possibilities when their characteristics are not appropriated accounted for in the selection of transducers. In this study, COMSOL simulations of the aluminum honeycomb panel materials were used to select transducers with resonance frequencies in proximity to the natural modes of the panel's honeycomb cells. This enabled improved transmission through the honeycomb panels in comparison to less targeted approaches. [SNL is managed and operated by NTESS under DOE NNSA Contract No. DE-NA0003525.]

11:15

4aSA6. Mechanical characterization of additively manufactured polymers using ultrasonic nondestructive testing. Akash Nivarthi (Appl. Res. Labs. - The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, anivarthi@arlut.utexas.edu), Michael R. Haberman, and Christina Naify (Appl. Res. Labs. - The Univ. of Texas at Austin, Austin, TX)

Ultrasonic nondestructive testing (NDT) can be used to relate the print settings of additively manufactured polymers to their macroscopic elastic properties. We present a comparison of the measurement of angle- and frequency-dependent ultrasonic transmission through a flat plate to predictions using a multiscale model that considers infill geometry and constitutive material properties. The experiment is an immersion test that uses a point source and synthetic linear array to measure the transmission coefficient from 0.2 to 1 MHz over a wide range of incident angles. Samples were fabricated using fused deposition modeling (FDM) to print infilled plates with polylactic acid (PLA) filament. Transmission measurements were compared to predictions from a multiscale model consisting of a finite element model to predict the effective anisotropic stiffness based on an assumed infill geometry and PLA material properties and an acoustic reflection-transmission model for an anisotropic elastic plate submerged in water based on the work of Rokhlin and Wang [*J. Acoust. Soc. Am.* **112**, 822 (2002)]. Minimization of the difference between the measured and modeled transmission coefficient for all angles and frequencies by varying model inputs provides an improved understanding of the effects of the print settings on the as-built mechanical properties for 3D-printed materials.

11:30

4aSA7. Lamb waves in thermoplastic polymer plates: An application to monitoring. Jakub Spytek (Institut Langevin, ESPCI Paris, Université PSL, CNRS, aleja Adama Mickiewicza 3, Krakow 30059, Poland, jspytek@agh.edu.pl), Claire Prada, Ros-Kiri Ing, and Julien de Rosny (Institut Langevin, ESPCI Paris, Université PSL, CNRS, Paris, France)

Thin thermoplastic polymer components are widely used in different fields of engineering, including high-technology industries such as automotive or energy. Due to the rapid technological development in these industries, there is an increased demand for polymer components to achieve advanced functionalities, such as the continuous monitoring of their structural health. In this work, we propose to take advantage of elastic guided waves, which are both emitted and measured using arrays of transducers integrated within the structure. This application requires first investigating the Lamb waves in these attenuating and heterogeneous materials. To that end, we estimate the dispersion curves of Lamb waves using a laser vibrometer. A transversely isotropic material model and a fitting algorithm allow for estimating the elastic parameters of the specimens. Then, for each mode, the frequency-dependent attenuation is evaluated. Finally using both theoretical models and experimental analysis, we evaluate the sensitivity of the guided waves to asses various structural changes in the thermoplastic polymer components such as temperature or the presence of contaminants.

11:45

4aSA8. Spatio-temporal evolution and precursory indicators of rock failure processes using acoustic emission tomography. Longjun Dong (School of Resources and Safety Eng., Central South Univ., Changsha, Hunan 410083, China, lj.dong@csu.edu.cn)

The characterization of stress and crack development plays important part in monitoring disasters. To capture the instability enlightenment from the spatio-temporal evolution of rock damage, this paper constructs the velocity field of key nodes in the rock failure process with the active and passive collaborative acoustic emission (AE) tomography method. It realizes the exploration of the micro-crack propagation law and its potential connection with the stress environment, combining the velocity field, the AE density field, and the energy field. Furthermore, from the spatial integrity and directional evolution of the velocity field, the mutation trend coefficient (MTC) and deformation coefficient (DC) of the velocity field are proposed as precursor indicators of rock instability, respectively. Results show that the anisotropy of the velocity field increases with the aggravation of rock damage, and the stress environment controls the spatial heterogeneity of the velocity field. The most important feature of spatial heterogeneity is that there are the continuous decrease and extension of low-velocity regions and the transformation from the high velocity to the low velocity. Real-time acoustic emission monitoring of the velocity field with MTC and DC can early identify rock instability precursors and principal stress direction, which provides vital pre-alarm information for geotechnical engineering.

THURSDAY MORNING, 11 MAY 2023 SALON I, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Speech Production I (Poster Session)

Fenqi Wang, Chair
Linguistics, University of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454

All posters will be on display from 8:00 a.m. to 12:00 noon. Authors of odd-numbered abstracts will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even-numbered abstracts will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSC1. Emphasis spread in Arabic: The domain and trigger. Aziz Jaber (Dept. of English Lang., Yarmouk Univ., Irbid 21163, Jordan, aziz@yu.edu.jo)

Acoustic coarticulatory effects of emphasis, a secondary articulation in the posterior vocal tract simultaneously accompanying a primary one, were investigated in Urban Jordanian Arabic. Nine native speakers of the dialect were recorded reading tri-syllabic monomorphemic and bimorphemic minimal pairs. The minimal pairs contained the voiceless emphatic fricative /s^{ʕ}/ and its plain counterpart /s/ in word initial and word final contexts. The acoustic correlates of emphasis include a raised F1, lowered F2, and raised F3 in the vowels preceding and following the emphatic sound. The results have roughly corroborated our findings in the previous research that the morpheme is, though disproportionately, still a confounding factor of emphasis spread. The most interesting contribution of this research is the perplexing behavior of emphasis spreading when crossing over the morpheme boundaries. Whereas the influence of the emphatic sound is evident on the morpheme falling to its left (e.g., prefixes) is evident, its influence of the morpheme falling to its right (i.e., suffixes) is less clear. This amounts to saying that the morpheme boundary between the stem and the suffix is stronger than that between the stem and the prefix. What this means is that a line of demarcation should be drawn between suffix boundary and prefix boundary.

4aSC2. Discerning race and gender linguistic cues from the acoustic signal. Bryan Ahaneku (Informatics, Univ. of Washington, Seattle, WA) and Rachel Elizabeth Weissler (Linguist., Univ. of Oregon, 1451 Onyx St., Eugene, OR 97403, rweissle@uoregon.edu)

There are multiple factors that contribute to how humans perceive social cues from the speech signal, specifically, cues relating to gender and race. This project sought to find what attributes of the speech signal allow humans to differentiate between speakers across different genders and varieties of American English. To answer this question, 2719 recordings of cis-men and cis women's voices, speaking in AAE (African-American English) and SdAE (Standardized American English) were analyzed. Machine learning methods were applied through WEBMaus, as opposed to Praat, MFA, and FAVE Align, to investigate the relationship between formants and the target variable (gender/English variety). The selection of formants as the focus of study was based upon pre-existing research affirming formants as a source of differentiation between men/women, and AAE/SdAE. Results of this study showed that formants have a minimal impact on perceiving social cues in speech according to WEBMaus. This research encourages further inquiry into other attributes of the speech signal, such as pitch or intensity, for discerning race and gender in the acoustic signal.

4aSC3. Declarative or declarative question?—On the acoustics of English prosody of Hong Kong trilingual children with autism spectrum disorder. Bruce Xiao Wang (Dept. of Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., 11 Yuk Choi Rd., Hung Hom, Hong Kong, brucex.wang@polyu.edu.hk), Si Chen, Fang Zhou (Dept. of Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Jiang Liu, and Cheng Xiao (Dept. of Lang., Literatures and Cultures, Univ. of South Carolina, Columbia, SC)

The current study investigated the production of English prosody (i.e., focus marking) of trilingual Cantonese children with autism spectrum disorder (ASD) and their typically developing (TD) peers (i.e., Cantonese and American English children without ASD) using declarative questions. Speech materials were segmented at word and syllable levels, and word duration, f0, f0 range and intensity were extracted. Acoustic data were fitted using linear mixed-effects models with different explanatory variables followed by a likelihood ratio test. Between group comparison showed that the ASD group had significantly more fluctuating f0 range in post-focus words than the TD groups, which is likely to be an indication of hypercorrection, i.e., over-application of perceived prosodic pattern in English declarative questions. Within groups, the Cantonese children showed different patterns to the English children in terms of the interaction between the acoustic measures and on-focus expansion and post-focus compression. The Cantonese ASD group showed some degree of post-focus compression in terms of

duration and mean f0, while the Cantonese TD group only had such pattern in terms of mean f0. The English TD group had a tendency of on-focus expansion in terms of duration and f0 range, but post-focus words showed significantly higher mean f0 than the pre- and on-focus ones, probably due to the question intonation.

4aSC4. Rhotics in the Salvador, Bahia, dialect of Brazilian Portuguese. Francis Jagiella (Lingusistics, Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 504, Bloomington, IN 47405, fjagiell@iu.edu)

Brazilian Portuguese has two rhotic phonemes: the alveolar flap /ɾ/ and another variable phoneme historically identified as the long version of the rhotic. Across publications, this phoneme has been identified as a velar, uvular, or glottal fricative, or as an alveolar trill or approximant. These phones vary within and across dialects. Deletion, especially word-finally, is also common. This paper examines a poorly documented variety, known for deletion of the rhotic, spoken in Salvador. Thirty-five participants read predetermined stimuli of isolated tokens and sentences (total rhotic phones = 5192). Acoustic and auditory analyses indicate the range of possible surface forms of the rhotic phoneme is more variable than previously cited, with the additional surface forms of palatal and velar fricatives as well as the uvular trill being common and a variety of additional, less common surface forms. When followed by a vowel, word-final rhotics are often neutralized with flaps. Among the possible surface forms, however, the glottal fricatives dominate. While deletion is most common word-finally, it occurs in all environments where the phoneme is found. Exploration of variation across speakers and its connection to demographic variation in the speakers is ongoing and will be reported.

4aSC5. Using automatic acoustic analysis to reveal disruptions to speech articulation in individuals at risk for psychosis. Kasia Hitczenko (Département d'études Cognitives, ENS, EHESS, CNRS, PSL Univ., Paris, France), Yael Segal, Joseph Keshet (Faculty of Elec. and Comput. Eng., Technion–Israel Inst. of Technol., Haifa, Israel), Vijay Mittal (Psych., Northwestern Univ., Evanston, IL), and Matthew Goldrick (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, matt-goldrick@northwestern.edu)

Individuals who are experiencing attenuated psychosis symptoms (placing them at clinical high risk for developing psychosis) exhibit disruptions to cortico-cerebellar circuits that manifest as difficulties in motor control of the face and limbs. These motor dysfunctions are predictive of the onset and progression of psychosis, making them a potential biomarker for this devastating disease. We examine whether these motor abnormalities disrupt speech production, leading to greater variability in speech acoustics. Clinical high risk (CHR) individuals and matched healthy control (HC) individuals produced diadochokinetic speech (rapid, repeated syllable production, e.g., papapa…, pataka…) and read aloud a paragraph. Consonant and vowel onsets and offsets were automatically segmented from diadochokinetic speech using a deep-learning model trained on human annotators. Read speech was automatically segmented at the word level using forced alignment. Vowel segmentation from read speech was based on forced aligner output; stop consonant onset/offsets were segmented by a deep-learning model. The segmentations from each task were then used to estimate speech rate, voice onset time (VOT), vowel durations and formant trajectories. CHR individuals produced more variable VOTs and exhibited greater speech rate variability than HC individuals in both tasks. This suggests that speech acoustics may provide a window into disruptions in this at-risk population.

4aSC6. Extended high frequencies for fricative classification in conversational speech. 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 (Northern Arizona Univ., Flagstaff, AZ)

The current study examines whether the information contained in Extended High Frequencies (EHFs) can improve random forest classification accuracy for fricatives in conversational speech. Prior phonetic research has investigated fricative categorization based on their acoustic characteristics, including spectral, temporal and amplitudinal measures. The spectral measures in these studies have largely been limited to frequency information below 8 kHZ. Only a few studies have examined the contribution of EHFs, energy exceeding 8 kHz. These studies have primarily focused on laboratory speech, so little is currently known about the role of EHFs in more spontaneous, conversational speech styles. Using a corpus of sociolinguistic interview speech from Western Canadian English sampled at 44.1 kHz, we compare classification models with and without frequencies above 8 kHz. We discuss the influence of EHFs on the categorization of fricative identity, and share a cost-benefit analysis of sampling at higher frequencies for speech research corpora.

4aSC7. Voice production in virtual reality: Effects of room size and fullness. 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) and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

This study explores the influence of visual input on voice production in virtual reality with healthy participants. The effects of the size and fullness of various virtual rooms on acoustic voice parameters and self-reported vocal status ratings are examined. Speech samples from healthy participants were recorded in six virtual conditions. After each condition, the participants provided subjective vocal status ratings on visual analog scales. The voice recordings were processed to calculate objective voice acoustic parameters including sound pressure level, standard deviation of fundamental frequency, mean pitch strength, time dose, and cepstral peak prominence smoothed. The effects of the virtual reality conditions on these acoustic parameters and the vocal status ratings were analyzed. The largest virtual room resulted in significantly higher (worse) vocal status ratings. The size of the virtual rooms had statistically significant effects on mean pitch strength and cepstral peak prominence smoothed, among associations with other objective voice acoustic parameters. This study provides evidence that larger and more full virtual rooms contribute to higher (worse) vocal status ratings and changes in objective voice parameters in healthy speakers.

4aSC8. Development of an algorithm for characterizing speech production patterns as context-based Cue Production Profiles. Deborah Torres (Speech Commun. Group, Res. Lab. of Electronics, MIT, 77 Massachusetts Ave., Rm. 36-413, Cambridge, MA 02139, dctorres@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, MIT, Cambridge, MA)

A given word or sound can take on a wide range of acoustic forms in conversational speech, and these patterns of variation are systematic with context. This substantial variability poses a problem for modeling human speech perception, but it can also be highly informative if it can be measured quantitatively; surface phonetic patterns can be used to characterize different speakers, dialects, speaking styles and speech disorders, as well as context-driven variation within a speaker. Adopting the view that individual acoustic cues provide a useful vocabulary for describing these variation patterns, we are developing an algorithm for tabulating the range of acoustic cue production patterns for each phoneme, in a corpus of recorded speech, with the potential for extension to words and phrases. This algorithm will enable the determination of key factors that govern systematic speech variation, including segmental, prosodic and lexical contexts, which can be used to create a profile for a speaker or defined population of speakers, i.e., a Cue Production Profile.

4aSC9. Age and sex effects on sound change: One size does not fit all. Felix Kpogo (Dept. of Linguist., Boston Univ., 621 Commonwealth Ave., Rm. 116, Boston, MA 02215, fkpog001@bu.edu)

This study examines whether speaker age and sex predict production of Advanced Tongue Root (ATR) vowel harmony by urban Twi (Niger-Congo, Kwa) speakers in Ghana. Traditional descriptions of the ATR harmony system state that [-ATR] /a/ is phonetically realized as [+ATR] [æ] before [+ATR] vowels /i, u, o/ and palatal segments. However, recent research indicates that this description is no longer accurate for all Twi speakers: in

contrast to suburban speakers, most urban speakers raise and front [æ] to [e], impinging on phonemic /e/, before /i, u/ and palatal segments (Kpogo, 2021). This pattern suggests a possible sound change (i.e., vowel merger) if suburban speakers are taken to represent a conservative variety of Twi. To explore the source of this innovation among urban speakers, variation in production across speaker age and sex was investigated. Preliminary acoustic data from a picture-naming task suggests that both factors are predictive of the innovative production pattern. Adolescents and younger adults, but not older adults, raise and front [æ] to [e], implicating the younger generation as leading this linguistic innovation. Additional findings suggest that men, but not women, are leading the innovation, a finding inconsistent with the overwhelming generalization that female speakers lead linguistic innovations.

4aSC10. Variable production of voiceless sonorants in Hakha La. Grayson Ziegler (Linguist., Indiana Univ., Bloomington, IN), Stefon M. Flego (Brown Univ., Bloomington, IN), and Kelly H. Berkson (Linguist., Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 852, Bloomington, IN 47405-2201, kberkson@indiana.edu)

Research on Tibeto-Burman voiceless sonorants often centers on nasals, which are standardly described as having two language-specific phonetic realizations: "voiceless unaspirated" nasals have a period of voicelessness and nasal airflow during closure followed by voicing prior to oral release ([m̥^{m}]), while "voiceless aspirated" nasals have a period of voicelessness and nasal airflow optionally preceded by a voiced period, and voicing following oral release ([$^{(m)}$m̥^{h}]) (e.g., Bhaskararao and Ladefoged, 1991; Chirkova *et al*., 2019). Little language-internal inter- or intra-speaker variation has been reported. An exception is South Central Tibeto-Burman (Chin) languages, such as Hakha Lai, in which voiceless nasals reportedly exhibit considerable inter- and intra-speaker variation (Hoffmann, 2018; Ziegler *et al*., 2022). As thorough investigation of this variation is absent from the existing literature, this paper investigates acoustic data from 9 native speakers of Hakha Lai (3M, 6F). Data for all voiceless sonorants (nasals, lateral, and rhotic) is presented, revealing that: (1) voiceless nasals in Hakha Lai indeed show a high degree of variation in the phasing of laryngeal and supralaryngeal gestures, defying the two-type approach; and (2) this variation extends to voiceless laterals and rhotics, which show similar bi-phasic and even tri-phasic variation.

4aSC11. Preschoolers with hearing loss have asymmetries in the rate of initiating conversations. Mark VanDam (Washington St. Univ., 412 E. Spokane Falls Blvd., SHS, HSB 125-X, Spokane, WA 99202, mark.vandam@wsu.edu), Lauren Thompson, David Jeson (Washington St. Univ., Spokane, WA), Aleah Brock (Univ. West Georgia, Carrollton, GA), Elizabeth Wilson-Fowler (Eastern Washington Univ., Spokane, WA), Sandie Bass-Ringdahl (Univ. Georgia, Athens, GA), and Paul De Palma (Gonzaga Univ., Spokane, WA)

Recent work has shown that preschool children initiate more conversations than their mothers and fathers. It has also been shown that boys and girls initiate conversations at about the same rate. It is not known if children with hearing loss initiate conversations at comparable rates to their typically-developing peers, or whether boys and girls with hearing loss differ. In this work, we collected daylong audio recordings from preschoolers with hearing loss using a body-worn audio recorder. We used automatic speech processing routines on the 7600 + hours of audio to identify talkers and conversational turns in their natural family settings. We examined how children with hearing loss initiate conversations compared with their typically-developing peers. We found no difference in rate of conversation initiation by hearing status or sex. We did find that regardless of hearing status or sex, children consistently initiate more conversations with their mothers than with their fathers. Results bear on the role of conversational exchanges, joint attention, and the auditory experiences during speech and language development in children with hearing loss.

4aSC12. Effects of utterance length on word and phrase durations. Seung-Eun Kim (Linguist., Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, sk2996@cornell.edu) and Sam Tilsen (Linguist., Cornell Univ., Ithaca, NY)

This study examined the effects of utterance length on word and phrase durations to investigate how the planned length of an utterance influences the rate of speech. An experiment was conducted in which the visual stimuli cued the production of sentences with one, two, or three subject noun phrases (NPs)—e.g., *"Nine green rhinos and eight red weasels and eight blue llamas live in the zoo."* In addition, a novel, delayed stimuli condition was tested in which the visual stimuli that cued non-initial NPs (*"eight red weasels and eight blue llamas"*) were presented after participants started production, and thus sentence length changed after utterance initiation. Word and NP durations were analyzed. NP durations were longer in sentences with more NPs, and moreover, durational increases were primarily observed in the rightmost words of the NPs. These findings are important first because they show that the previously observed effects of number of words on word durations extend to phrases, and second because they show that phrase duration effects are localized to phrase ends. When the utterance length changed after the start of production, participants lengthened the end of the initial NP and the following conjunction to plan for the newly presented NPs.

4aSC13. Evidence for raw acoustics as the target of phonetic imitation. Ivy Hauser (Linguist., Univ. of Texas Arlington, 701 Planetarium Pl., Box 19559 — 132 Hammond Hall, Arlington, TX 76019, ivy.hauser@uta.edu), Emily Graham, and Xinwen Zhang (Linguist., Univ. of Texas Arlington, Arlington, TX)

Phonetic imitation (also called convergence or accommodation) occurs when talkers alter their production towards speech they hear, even in lab settings without explicit instruction to imitate. Though general evidence for imitation is robust, much of the existing work does not distinguish between convergence towards a linguistic target versus convergence towards an acoustic target. This study presents a direct test case of these hypotheses through spontaneous imitation of the English alveolar sibilant /s/ in a delayed shadowing task. Participants first produced /s/-initial words, then were exposed to model speech with either enhanced or reduced spectral mean (SM) on /s/, and finally produced /s/-initial words again post-exposure. The model talker had higher than average baseline SM, so raw acoustic values were higher than those of most participants. All participants exposed to the enhanced stimuli raised SM, converging towards both the raw acoustics of the model talker and the linguistic pattern of enhancement. However, participants exposed to the reduced stimuli also raised SM. This diverged from the linguistic pattern of reduction but converged towards the raw acoustics of the model talker, whose SM was still higher than participants' even when reduced. These results suggest raw acoustics can be the target of phonetic imitation. Possible interactions with phonological contrast, implications for the perception-production link, and methodological considerations for future imitation studies are discussed.

4aSC14. Acoustic analysis of voicing errors in children with childhood apraxia of speech. Michelle T. Swartz (Speech Lang. Pathol., Thomas Jefferson Univ., 901 Walnut St., 23rd Fl., Philadelphia, PA 19107, michelle.swartz@jefferson.edu), Laura Koenig (Haskins Labs, New Haven, NY), and Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., Bloomfield, NJ)

Childhood apraxia of speech (CAS) is a complex neurological speech sound disorder (SSD) that involves impaired speech motor planning and programming. Speech characteristics of CAS include difficulty sequencing speech movements in the absence of muscle weakness resulting in segmental and suprasegmental speech deficits. Past work indicates that children with CAS are delayed in acquisition of voicing contrasts, and are often perceived to produce errors or distortions of voicing. However, not all past studies employed acoustic analyses to assess voicing characteristics in CAS. This study will use voice onset time (VOT) to analyze the voicing contrast in children with CAS (aged 8;0 to 15;11) compared to peers with residual speech sound disorder (RSSD) and typical development (TD). VOT of initial consonants from twelve children (CAS = 4, RSSD = 4, TD = 4) will be assessed in words of varying length, stress pattern and target complexity. Given that children with CAS are known to experience greater articulatory breakdowns in phoneme sequences of increasing length and complexity, we expect that group differences will increase with task complexity. Overall, based on past work, we also predict that children with CAS will demonstrate shorter and/or more variable VOTs for plosives than children with SSD and TD.

4aSC15. Gender-specific covariation of voicing cues in stops: Girls are leading the sound change. Lian J. Arzbecker (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 37 Pressey Hall, Columbus, OH 43210, arzbecker.1@osu.edu), Riley Goebel, Ewa Jacewicz, and Robert A. Fox (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Multiple cues to stop coda voicing in English can be distributed over larger domains, covarying at a long distance within a syllable (Hawkins and Nguyen, 2004). Our recent work uncovered systematic temporal relationships between voiced and voiceless consonants in the coda and the syllable-initial /b/, indicating that information about coda voicing—phonetic detail cueing the phonological voiced/voiceless contrast—is already available at the onset (Jacewicz *et al*., 2021). These systematic long-distance cues to coda voicing are further altered by regional dialect variation that defines the relative durations of /b/-closure and (un)broken voicing in this closure (extending from the preceding sonorant). However, systematic dialect effects were found in adult females and males (Arzbecker *et al*., 2022) but not in girls (Jacewicz *et al*., 2021). Here, we present the corresponding acoustic data from 8 to 12-years old boys (n = 47). The results show that boys have retained the pronunciation of men: They produced more voicing in the closure (83%) than girls (55%), and the closures of Southern boys were almost fully voiced, following the dialect-inherent pattern. Adding the missing link, the current study provides evidence that girls are leading the sound change and the covariation of voicing cues is gender specific.

4aSC16. Modelling the intensity difference of Spanish alveolar taps with finite mixture models. Scott J. Perry (Linguist., Univ. of AB, 3-28 Assiniboia Hall, Edmonton, ON T6G 2E7, Canada, sperry1@ualberta.ca), Matthew C. Kelley (Linguist., Univ. of Washington, Seattle, WA), and Benjamin V. Tucker (Linguist., Univ. of AB, Edmonton, AB, Canada)

The production of stops has been documented to vary considerably in several languages. The difference between the minimum intensity during the consonant and the maximum intensity of the surrounding vowels is an acoustic correlate of taps, whose realizations include plosives, approximants, and deletions. We investigated how lexical, phonetic, and predictability-related factors are associated with changes in the intensity difference of Spanish taps. We conducted an acoustic analysis using a force-aligned corpus of conversational Spanish, with ten percent of tokens hand-corrected to evaluate performance. Model checking with generalized linear models indicated that one distribution could not adequately account for the observed data. As such, we analyzed variation in intensity difference using finite mixture models comprising two skew-normal distributions, which provided a substantially better fit. The results of our modelling approach require a more nuanced interpretation than standard linear models. Our model indicates that Spanish tap production is a complex system where some variables, like frequency, are related to categorical shifts between two potential realizations, and other variables are related to gradient intensity changes within each potential realization of the tap. We also find that articulatory factors like speech rate are responsible for larger intensity changes than lexical properties like frequency.

4aSC17. Acoustic variability of examiners' productions in language assessment. Elizabeth Ancel (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE115 Shevlin Hall, Minneapolis, MN 55455-0279, ancel014@umn.edu), Lizbeth Finestack, and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Speech-Language Pathologists and other professionals 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 in assessment is sentence repetition, in which a child repeats a live-voice production of a sentence. Sentence repetition tasks necessarily reflect the unique prosody used by experimenters, and in particular their rate of speech and f0 patterns. This study investigates examiners' rate and f0 variation as they administer a sentence repetition task in order to quantify variability within and between the examiners. To do this, we analyzed recordings of 25 different speakers administering the Sentence Repetition subtest of the Clinical Evaluation of Language Fundamentals (CELF) to children aged 3 through 10 years old. For each production of a sentence, we measured the mean pitch, pitch range, and speaking rate. The results indicate the extent to which each examiner varied these prosodic features based on the assessment item and the age of the child undergoing the assessment. This variability is compared across examiners to capture a possible point of inconsistency that may affect children's performance on these assessment tasks.

4aSC18. Speech directivity patterns generated from a high-fidelity speech corpus. Allison Trine (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, at17@illinois.edu), Margaret Miller (Boys Town National Res. Hospital, Omaha, NE), Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), G Christopher Stecker (Boys Town National Res. Hospital, Omaha, NE), and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

Human talkers are directional sound sources—a phenomenon that has consequences for speech perception in multi-talker environments. Directivity patterns for speech showing frequency- and angle-dependent radiation reveal that speech generally becomes more directional toward the front of the talker as frequency increases. Differences in physical attributes can lead to individual variability in directivity patterns across talkers. Here, we examine individual variability in speech directivity using frequency-dependent directivity indices and directivity maps. Speech directivity was examined in the horizontal plane using a corpus of simultaneous multi-channel full-bandwidth (48-kHz sampling rate) recordings of the Bamford-Kowal-Bench (BKB) sentences recorded in an anechoic chamber. Thirty subjects (15 female) were recorded. The long-term average speech spectrum was utilized to calculate directivity indices in 1-ERB (equivalent rectangular bandwidth) bands. Gender differences in directivity indices were evaluated using a linear mixed-effects model. There was no main effect of gender. There was a main effect of ERB band with higher-frequency bands tending to have higher (i.e., more directional) directivity indices, however there was a nonmonotonic relationship between average directivity indices and frequency. Directivity maps demonstrated individual differences in speech radiation. [Work supported by NIH under Grant No. R01-DC019745.]

4aSC19. Using a production-center methodology to probe syllabic constituency. Yu-Jung Lin (World Lang., Literatures, and Cultures, College of the Holy Cross, Worcester, MA) and Kenneth J. de Jong (Dept. of Linguist., Indiana Univ., Ballantine Hall, Bloomington, IN 47405, kdejong@indiana.edu)

Research on speech production in the P-center paradigm (Rapp, 1972) explores how speakers produce various items with temporal regularity, as guided by a metronome. Studies find a regular pattern in which, with small offsets created by different types of onset consonants, and by material

appearing after the vowel, the acoustic onsets of stressed vowels are aligned with the metronome. The current paper examines the extent to which language structure, in the form of syllabic constituency, affects speech in metronome alignment. Previous work (Lin, 2020) finds strong evidence that Mandarin speakers associate approximants with neighboring vowels, contra English wherein labial approximants (as in *quick*) appear to associate with the previous onset consonants. In the current work, 6 speakers of Mandarin produced syllabic forms in time to a metronome. Stimuli included items with and without vowel-like approximants before the vowel. Mandarin productions show strong evidence for metronome alignment with the onset of these approximants, as would be the case if a vowel complex forms the main unit of alignment and approximants are part of this vowel complex. Analysis of the parallel English productions is currently underway, and will be reported.

4aSC20. The effect of phrasing, speech rate, and information structure on tonal coarticulation in spontaneous Cantonese. Xin Gao (Linguist., Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104, kauhsin@sas.upenn.edu) and Mark Liberman (Linguist., Univ. of Pennsylvania, Philadelphia, PA)

Pre-low raising is a tone coarticulation phenomenon in which the pitch of a high-tone syllable is higher before a low tone than before a high tone. Lee and Xu (2016) found that speech rate influences the degree of this pre-low raising, in recordings of read disyllables in carrier phrases. In the present study, we examine this phenomenon in spontaneous Hong Kong Cantonese using the CantoMap Map Task dataset (Winterstein *et al*., 2020), also exploring the possibility of similar effects on the realization of other tones. Along with speech rate, we also consider the possible effects on this tonal dissimilation of phrasing (relations between a syllable and the subsequent syllable), the relative location of a syllable in a sentence, and information structure (the role of the word in the map task interaction). We expect that all of these factors will influence tonal coarticulation in general, and pre-low raising in particular.

4aSC21. Duration impedes loss of the palatal lateral in Languedocien as compared to Gascon. Kaitlyn Owens (Indiana Univ. Bloomington, 355 North Eagleson Ave., GA 3151, Bloomington, IN 47405, kaitowen@iu.edu)

Previous acoustic evidence suggests loss of /ʎ/ in the Gascon dialect of Occitan is due to contact with French, however this study probes acoustic evidence for the notable retention of /ʎ/ in the Languedocien dialect. Both dialects of Occitan have been in intense long-term contact with French and maintain /ʎ/ in positions where it was lost in the development of French. We analyze 181 tokens of /ʎ/ in oral narratives performed in the contemporary dialects of Gascon and Languedocien in the *OcOr Corpus* by using mixed-effects linear regression to predict the durations of the lateral and glide segments in word-medial positions in tokens of [ʎj], the most frequently realized variant of /ʎ/ in both dialects. By examining the durations of both the glide-like transition and lateral segment in these tokens, we find that the lateral segment of these variants is longer in Languedocien than other segments in either dialect ($p = 0.0009$). We propose that the increased duration of the lateral segment renders the lateral feature of /ʎ/ more salient in Languedocien than in Gascon. Thus, this increased duration aides in impeding the loss of the lateral segment due to contact, given that synchronic perception may influence diachronic sound change.

4aSC22. Neural networks' posterior probability as measure of effects of alcohol on speech. Ratree Wayland (Linguist., Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, ratree@ufl.edu), Kevin Tang (English Lang. and Linguist., Heinrich-Heine-Universität Düsseldorf, Düsseldorf, Germany), Fenqi Wang (Linguist., Univ. of Florida, Gainesville, FL), Sophia Vellozzi, and Rahul Sengupta (Comput. and Information Sci. Dept., Univ. of Florida, Gainesville, FL)

Intoxication has a well-known effect on speech production. Lester and Skousen (1974) reported that the place of articulation for /s/ is retracted and /tʃ/ and /ʤ/ are deaffricated (i.e., substituted by a non-affricate segment) in drunken speech. Zihlmann (2017) further established the robustness of deaffrication as it cannot be consciously suppressed under intoxication. Using these prevalent speech errors as test cases, this study extends a phonologically-informed neural network approach to the study of intoxicated speech. The approach has success in measuring pathological speech and lenition patterns in healthy speakers. Degrees of place retraction for /s/ and deaffrication of /tʃ/ and /ʤ/ are estimated from posterior probabilities calculated by recurrent neural networks trained to recognize [anterior], [continuant] and [strident] features. When applied to a corpus of alcohol English speech, preliminary results suggested that sober versus drunken state could be reliably predicted by the three posterior probabilities. The directions of the effects are largely in line with previous studies. For example, /tʃ/ and /ʤ/ are more fricated (higher strident and continuant probabilities), and /s/ is more retracted (lower anterior probability) in drunken compared to sober speech. The results suggest that the intoxicated speech can be reliably quantified by this new approach.

4aSC23. Comparing segmental and suprasegmental features in speaker gender perception: An acoustic distance approach. Brandon Merritt (Rehabilitation Sci., The Univ. of Texas at El Paso, 1101 N. Campbell, El Paso, TX 79902, bmmerritt@utep.edu)

Segmental (e.g., articulation) and suprasegmental (e.g., intonation) speech features provide meaningful perceptual cues to speaker gender. Yet, the relative perceptual importance of these features remains poorly understood. Further, proposed methods for quantifying these features often require labor-intensive measurement by human judges. This study used an acoustic-only method to quantify segmental and suprasegmental features among 60 speakers with varied gender identities (e.g., non-binary, transgender and cisgender men and women) and compared these acoustic measures to human perceptual judgements of speaker gender. Thirty listeners rated confidence in speaker gender and masculinity/femininity for each speaker. To obtain acoustic distance among speakers, three conditions were created that manipulated which acoustic features were present in the speech utterances. Acoustic distance was calculated in each of the three conditions between each speaker and a cisgender man and woman who were perceptually rated as most gender prototypical. Acoustic distances for the three conditions were fit to regression models to determine which was most predictive of listeners' perceptual judgements. Both segmental and intonation features predicted perceptual measures. However, segmental cues better explained listener's ratings, suggesting their greater perceptual weighting in listeners' judgements of speaker gender.

4aSC24. Lexical tone and vowel duration in San Sebastián del Monte Mixtec. Jae Weller (Lang. Sci., Univ. of Wisconsin–Madison, 1168 Van Hise Hall, 1220 Linden Dr., Madison, WI 53706, jdweller@wisc.edu), Jeremy Steffman (Linguist., Northwestern, Chicago, IL), Félix Cortés (Proyecto Club Alma Mixteca, Huajuapan de Leon, Oaxaca, Mexico), and Iara Mantenuto (English, California State Univ., Dominguez Hills, Carson, CA)

Vowel duration has been shown to vary as a function of lexical tone (Gandour, 1977; Gordon, 2001; Yu, 2010). Here we explore the relationship between tone and duration in a dialect of Mixtec. San Sebastián del Monte Mixtec (SSM) has three tones which link to individual moras: H(igh) M(id) L(ow). These can be combined in a long (bimoraic) vowel, in some cases creating contour tones: HH MM LL HL ML LM (Cortes *et al*., forthcoming). Audio recordings of 14 native speakers (9F, 5M) were collected, producing 264 utterances with target words in phrase-medial position. The corpus includes all tones, as well as all vowels /i,e,a,o,u/ across four possible word shapes (with several lexical gaps). With mixed effects modeling, we find that duration varies as a function of tone with vowel and onset consonant controlled for (1) L tones are shorter than M and H tones in monomoriac syllables, and (2) tones in bimoraic (long vowel) monosyllabic words vary in duration as well: {MM,HH,LH} > {ML,LM,LL}. First vowels are longer, consistent with stress on the initial syllable. We further discuss the relation between duration, mean F0, and F0 range, and the influence of vowel features and preceding consonant on duration.

4aSC25. Acoustic characteristics of variation in vowels in Assamese. Jupitara Ray (Linguist., Boston Univ., 1384 Commonwealth Ave., Apt. 30, Boston, MA 02134, jupitara@bu.edu)

This study provides an overview of acoustic characteristics of the vowel system in three regional varieties of Assamese. Recordings were made of 24 speakers producing the vowels /i/, /e/, /ɛ/, /ɑ/, /u/, /ʊ/, /o/ and /ɔ/ from a wordlist of 128 tokens. The speakers came from three locations in Assam, India: namely Lower Assam, Upper Assam, and Guwahati, and were balanced for region, sex, age, and education background. Acoustic analysis was conducted in Praat. First, the formant settings for male and female speakers were adjusted accordingly: 5500 Hz for female speakers and 5000 Hz for male speakers, where the vowel onsets and offsets were identified perceptually in the waveform, coinciding with the spectrogram in Praat. First and second formants were measured at the midpoint of the vowels for each of the 24 talkers, for a total of 9216 formant measurements. Formant measurements were then plotted on an F1xF2 chart using R. Findings show that Upper Assamese speakers had a high /ʊ/ production, while Lower Assam speakers had a low /ʊ/ in the acoustic space, while within Guwahati, more educated speakers produced /ʊ/ that was higher compared to speakers with less education.

4aSC26. Automatic detection of nasalization in speech via cue-based analysis. Blisse Kong (Speech Commun. Group, Res. Lab. of Electronics, MIT, 50 Vassar St., Cambridge, MA 02139, blisse@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, MIT, Cambridge, MA)

Modern speech recognition systems are powerful, but it is not simple to understand the mechanisms by which they model speech, or precisely how they make use of acoustic cues. Moreover, a system for the explicit detection of individual acoustic cues may be useful not only for speech recognition but for understanding the principles that govern sub-phonemic patterns of surface-phonetic variation, and how specific cue patterns may reflect speech disorders. This report describes a transparent, acoustic cue-based nasalization detection module, that can be used to find not only nasal consonants but also nasalization wherever it appears, e.g., within a vocalic region. We lay out a framework for extracting measurements that are key to detecting nasalization, in particular, the nasality-related spectral peaks P0 and P1, and confirm the utility of these measurements for nasality detection in spoken words. In particular, we present a pre-processing and cue-value extraction framework, and propose a Gaussian Mixture Model-based approach to detect the regions of nasalization in a speech signal. This work on a detection module for nasalization is part of a larger effort to develop a speech recognition system that is based on landmarks and other acoustic cues.

4aSC27. Speech perception and children with speech sound disorder: An assessment of non-errored speech sounds. Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., 1515 Broad St., Attn: Elaine Hitchcock, Bloomfield, NJ 07003, hitchcocke@montclair.edu) and Laura Koenig (Haskins Labs, New Haven, NY)

Previous studies assessing speech perception in children with speech sound disorder (SSD) suggest inconsistent, if any, differences from typically-developing peers (TD). However, others suggest that children with SSD perceive inaccurate productions as acceptable variants of a sound. Currently, limited research explores speech perception of non-errored sounds in children with SSD compared to TD peers. This study assesses how distributional characteristics of naturally produced child speech stimuli, collected from six 2–3-year-old English-speaking children, might impact listening behavior of children with SSD for *non-errored sounds*. Stimuli included six exemplars per speaker and VOT categories of short-lag /b d/, short-lag /p t/, long-lag /b d/, long-lag /p t/. For each POA and VOT group, /b d/ and /p t/ VOTs were bimodally distributed (shorter for voiced targets), separated by a minimum 5 ms gap. We will compare perception results for TD children and children with SSD (ages 6;0–10;11). Earlier findings showed high accuracy in adults and TD children for targets with appropriate VOT values with adults slightly outperforming children. We hypothesize that children with SSD will demonstrate similar accuracy levels between groups when VOT values are appropriate for the target, and slightly greater response variability and group differences for inappropriate VOTs.

4aSC28. Acoustic properties of vocal emotions in American English and Mandarin Chinese. Fenqi Wang (Linguist., Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, fenqi@ufl.edu) and Ratree Wayland (Linguist., Univ. of Florida, Gainesville, FL)

Previous studies have indicated that different vocal emotions in American English and Mandarin Chinese have distinct acoustic profiles, but the acoustic profiles of vocal emotions have not been established for each language. This experiment analyzes recorded sentences produced in five emotions by 10 native English speakers and 10 native Mandarin Chinese speakers from a novel emotional speech database (Zhou *et al*., 2021). To better understand the acoustic characteristics of the emotional utterances in both languages, this study used the feature extraction toolkit openSMILE to extract 6373 features (Schuller *et al*., 2016). Principal component analysis was applied to the extracted features for feature selection. Linear mixed-effects regression models were performed to determine the effect of emotions on the selected features. Bayesian multinomial logistic regression models were performed to examine the effects of acoustic features on emotions. The results suggested that American English and Mandarin Chinese exhibit different acoustic patterns of vocal emotions, but some features, particularly pitch-related features, are used by both languages to express vocal emotions. However, pitch variation may be more restricted in the vocal emotions of Mandarin Chinese relative to American English.

4aSC29. Categoricity and gradience: A case study of Sakha lateral liquids. Hassan Munshi (Linguist., Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104, hm679@sas.upenn.edu)

I report on a recent acoustic study on the lateral liquid /l/ variation in Sakha (Turkic). I investigated /l/ darkening in Sakha in multiple lexical (including mono- and bi-morphemic) environments. In particular, I show that the effects of neighboring vowels on /l/ darkening are quite remarkable, to the extent that one is tempted to conclude that /l/ is participating in vowel harmony. I do not necessarily make this claim, but it is quite clear that adjacent vowels play a major role in Sakha/l/ darkening. The results show that /l/ variation in this language, like English, is both categorical and gradient. Backness of vowels determines which /l/ to appear. Therefore, the major finding of the study is that, rather than syllable structure, it is primarily vowel backness that determines /l/ variation in this language. This, in turn, indicates that variation of this consonant is categorically affected by vowel harmony. At the same time, I also discuss how one could potentially argue for a completely phonetic (i.e., gradience) approach. Still, based on the investigation of /l/ variation in several environments, I conclude that categoricity and gradience each plays a distinct role.

4aSC30. Vocal level and vocal effort variability across phrase types. Lady Catherine Cantor Cutiva (Com Sci. & Disord., Michigan State Univ., East Lansing, MI, cantorcu@msu.edu), Mark Berardi (Univ. Hospital Bonn, Bonn, Germany), Sarah Gundry, Hannah MacDonald, and Eric J. Hunter (Com Sci. & Disord., Michigan State Univ., East Lansing, MI)

Vocal effort is the most common symptom that in patients with voice disorders and decreasing vocal effort is often a therapeutic goal. Previous studies indicate that a vocal effort goal can elicit voice production with very similar acoustic parameters. The purpose of this study is to clarify the effort and acoustic parameter dependencies on different types of phrases elicited with different goals. Three types of speech (automatic, read, structured spontaneous) were elicited at using three communication goals (confidential, conversational, raised) with vocal effort quantified using a Borg CR100 scale. Participants' vocal level reliably changed between the three elicited communication goals. Further, there was some trending differences depending on the length and type of speech. Further details will be presented. The results indicate the utility of the Borg CR100 in tracking effort in voice production that is repeatable with respect to vocal level (dB). Additional acoustic parameters as well as speech differences will be discussed.

4aSC31. Gender, but not speech material, affects extended high-frequency levels in the speech spectrum. Vahid Delaram (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901, S 6th St., Champaign, IL 61820, vahidd2@illinois.edu), Margaret Miller (Boys Town National Res. Hospital, Omaha, NE), Rohit M. Ananthanarayana (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL), G. Christopher Stecker (Boys Town National Res. Hospital, Omaha, NE), Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

Current evidence suggests extended high-frequency (EHF) speech cues support speech perception. Audibility of these cues likely depends on speech spectral levels at EHFs. These levels may vary across genders and different speech materials. In this study, we investigated the effect of talker gender and speech materials on EHF levels in speech. A group of 30 (15 female) native speakers of American-English was recruited to participate in this study. A three-minute spontaneous narrative was recorded for each participant along with a subset of the Bamford-Kowal-Bench (BKB) sentences. An ERB-scaled long-term average speech spectrum was calculated for the narrative and for the BKB sentences for each subject. Linear mixed-effects models were used to test intersubject and intrasubject variability in 8 EHF ERB bands. There was a significant effect of gender with female EHF levels ~4 dB higher than male EHF levels. Within-subjects comparison of BKB sentences and narratives revealed no significant difference in EHF levels between speech materials. These findings highlight the possibility that EHFs could play a more prominent role in female speech perception than male speech perception, and suggest that EHF levels are relatively stable across speech materials for a given talker. [Work supported by NIH under Grant No. R01-DC019745.]

4aSC32. Acoustic speech parameter relationships with voice disorders and phrase differences. Sai Aishwarya Ramani (Com Sci. & Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824-3405, ramanisa@msu.edu), Eric J. Hunter, and Lady Catherine Cantor Cutiva (Com Sci. & Disord., Michigan State Univ., East Lansing, MI)

While acoustic speech analysis is non-invasive, the utility has been mixed due to the range of voice types. For vocal health practitioners to efficiently and quickly assess and document voice changes, knowing which voice parameter would be sensitive to vocal change is crucial. Using a database of 296 individual voices including 8 voice pathology types and typical voice samples, the sensitivity of a range of acoustic speech parameters to differentiate common voice pathology types was investigated. Both traditional and contemporary acoustic speech metrics were estimated for the samples using a custom MATLAB script and PRAAT (e.g., jitter, shimmer HNR, CPPS, Alpha ratio, PPE). Analysis then evaluate the predictability value of the metrics to discriminate pathology type. From the pool of parameters, 11 were able to identify pathological voices from normal controls and several of the parameters were more sensitive to some pathology. For example, CPPs and jitter values could discriminate neuropathological voices whereas HNR and Shimmer cold discriminate muscle-based pathologies. These results indicate how the sensitivity of acoustic speech metrics to the voice pathology types can allow for the identification of individual metrics (or combinations of metrics) which could be used to track changes in vocal health.

4aSC33. Pitch elevation and vocal loudness are related to swallowing safety. A proposal to improve their performance. Adrián Castillo-Allendes (Com Sci. & Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824-3405, casti208@msu.edu) and Eric J. Hunter (Com Sci. & Disord., Michigan State Univ., East Lansing, MI)

The ability to perform high-pitched and loud tasks have been associated with lower risks of penetration and aspiration in people with oropharyngeal dysphagia (OD). Previous studies suggest that these two vocal tasks may improve swallowing-related kinematics; laryngeal elevation and glottic closure. This study aims to identify which types of vocal exercises can maximize pitch elevation and vocal loudness. This ongoing project involved five subjects with OD who were asked to perform three vocal exercises: effortful pitch glide, straw phonation, and water resistance therapy (WRT). Fundamental frequency (*f*o) and maximum dB SLP were obtained during their performance; moreover, the vocal effort was quantified using a Borg CR100 scale. Descriptive statistics were used to characterize the differences between the three proposed exercises. WRT was the exercise that allowed to obtain higher pitch elevation values with less vocal effort (median *f*o = 603 Hz), over EPG(594 Hz), and then straw phonation (512 Hz). Furthermore, EPG allowed reaching a higher vocal loudness during its performance. These preliminary results indicate that WRT could be a potential therapeutic adjunct to improve swallowing-related biomechanics. Further details will be presented. Future studies could incorporate accelerometry to obtain more accurate *f*o and dB SPL measures during vocal exercise performance.

4aSC34. Prosodic variation and rich linguistic environments in a preschool setting. Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu) and Kimberly Nesbitt (Dept. of Human Development and Family Studies, Univ. of New Hampshire, Durham, NH)

Children require rich linguistic environments (RLEs) to support their development in early education settings (Bratsch-Hines *et al.*, 2019; Justice *et al.*, 2018). RLEs are defined by high-quality language, teacher tone (prosodic/fundamental frequency-f0 variation), and interactions between the teacher and children (increased turn-taking, contingent responses, open-ended questions) (Jones & Rowland, 2017). As we support enhancing the quality of interactions, we lack a quantifiable way to assess teacher language to provide clearer feedback. Our aims are (1) to quantify the measures used to assess the quality of teacher language, (2) to examine teacher prosodic variation across a typical day, and (3) to explore any connections between teacher (quality) language and prosody. Audio data were recorded from preschool classroom teachers (teachers of children aged 3 to 4-years-old; n = 5 to date). Each site had two days of recording, 4 to 8 hours per day. Teachers completed a background survey and reported a log of daily activities. The recordings are transcribed to analyze language and speech: measures of f0 variation, types of open-ended questions, word usage diversity, number of turns taken between the teachers-students, and presence of contingent responses. Data from this study help us to understand RLEs and provide a coding system to systematically capture teacher language in a more nuanced manner.

4aSC35. Within- versus between-speaker acoustic variability in Thai. Yoonjeong Lee (Linguist., Univ. of Michigan, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeol@umich.edu) and Jody Kreiman (Head and Neck Surgery, UCLA, Los Angeles, CA)

This paper continues our studies of acoustic variability in voice between and within speakers. Previous work indicates that acoustic variability is characterized by the balance between high-frequency harmonic and inharmonic energy in the voice (measured using cepstral peak prominence) and by formant dispersion, regardless of the speaker's sex, native language, or speaking style. Our recent investigation of the language effect on voice variability surveyed three languages, English, Korean, and White Hmong, which form an organized subset with respect to their linguistic use of F0 and the phonological status of phonation quality. The cross-linguistic comparisons revealed a second tier of language-specific variability that reflects features of the phonology of each language. The present paper adds a tone language, Thai, to the mix. Samples of read speech from fifty Thai speakers (33F, 17M) were evaluated as in our previous studies. Results revealed that variability in Thai voices is primarily accounted for by the same biologically relevant measures observed globally, and further shaped by the language-specific use of F0. These data further support our hypothesis that acoustic variability in voice is governed first by biological factors, secondly by language-specific, cultural factors, and finally by idiosyncrasies of an individual's anatomy and personal speaking style.

4aSC36. Acoustical analysis of Taylor Swift's speech and singing throughout her career. Miski A. Mohamed (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, moha1180@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Taylor Swift has changed geographic environments, speaking communities, and singing communities throughout her career, inviting question into changes in her voice acoustics. Capitalizing on the availability of interviews and multiple song recordings, we tracked changes in vowel formant trajectories and other vocal properties from different points in her life to see how her voice might have been influenced by these changes in community (e.g switching from country to pop style, moving from Philadelphia to Nashville). Comparing vowel space from 2012 to 2019, we observe higher tongue position (lower frequency F1) for numerous vowels, especially for /eɪ/ and /aʊ/. Rounding for /ɔ/ was reduced, and /æ/ was pronounced farther back in the mouth. We also observed reduced distinction between the varieties of /ɑɪ/ before voiced and voiceless consonants (e.g., ride versus write), as well as greater front-back movement for /u/. Studio re-recordings show remarkable consistency of word timing and intensity. Conversely, Swift's live singing voice includes some variation from the recording, including fewer silences between words. Live versions feature extended duration for words that are emphasized for emotion, and/or ends of phrases, compensated by shorter durations for other segments so that the overall song length was very consistent.

4aSC37. Individual differences in acoustic variability and perceptual sensitivity to subphonemic variation. Sarah Harper (Neurological Surgery and Weill Neurosci. Inst., Univ. of California, San Francisco, 1651 4th St., San Francisco, CA 94158, sarah.harper@ucsf.edu)

Individual differences are robustly attested in both speech production and perception, with interspeaker variation in production frequently observed in the same domains in which extensive variation is observed across listeners in perception. This study extends existing research by examining whether the relationship between production variability and perceptual sensitivity previously observed for vowels (e.g., Perkell *et al*., 2008; Franken *et al*., 2017) is also observed for consonants. 55 speakers of American English completed two online tasks designed to evaluate individual differences in the production and perception of /s/ and /ɹ/: a sentence production task and a 4IAX perceptual discrimination task. Preliminary comparison of acoustic variability measurements from the production task and response accuracy measurements from the perception task suggest that individual differences in these two domains are sometimes, but not always, related across speakers. Specifically, production variability and perceptual sensitivity were correlated for phonologically meaningful acoustic dimensions in the liquid phoneme /ɹ/, but not for any measured dimensions in the fricative phoneme /s/.

4aSC38. Effects of bilateral cochlear implants on vocal intensity control. Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, 901 South Sixth St., Champaign, IL 61820, simins2@illinois.edu) and Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Bilateral Cochlear Implants (BiCIs) can improve many perceptual tasks, including speech perception in noise, when compared to unilateral implantation. However, evidence suggests that BiCI users may perform worse on some vocal tasks, such as F0 production, than when using their better ear alone. This research aimed to determine if this detrimental effect on vocal control with BiCI devices extends to the control of vocal amplitude variations. Ten BiCI users were asked to produce a sustained vowel vocalization using both CIs and one CI separately. Their recordings were analyzed acoustically to determine long-term control of amplitude variations using the Variation of Peak Amplitude (vAm). The preliminary findings showed that using both CIs resulted in less control over vocal intensity for eight participants when compared to using one ear alone. The other two participants either showed no difference between the conditions or had a higher vAm with the unilateral ear than with the bilateral condition. The result suggests that BiCI use might detrimentally affect controlling long-term vocal intensity variation beyond any effects of unilateral CIs. Further investigation is needed to address how the vocal intensity control in BiCI users is affected.

4aSC39. Acoustics of the creaky phonations in Yateé Zapotec. Yuan Chai (Linguist., Univ. of California Los Angeles, 3125 Campbell Hall Box 951543, Los Angeles, CA 90095, yuanchai@g.ucla.edu), Adrian Fernández, and Briseida Mendez (Los Angeles, CA)

Yateé Zapotec (YZ) is a variety of Zapotec spoken in San Francisco Yateé, Oaxaca, Mexico. The language has three phonation types: modal, checked, and rearticulated. Checked and rearticulated phonations both involve glottalization realized on vowels but differ in the phasing of the glottalization in vowels. Checked phonation has late-phased glottalization whereas rearticulated phonation is characterized by mid-phased glottalization. Previous studies on the phonetics of Zapotec phonations seldom compare the phonations in the time domain. In this study, we compare the spectral tilt, degree of noise, amplitude of voicing, and duration of the three phonations over the time course of vowels. Using multi-dimensional scaling analysis, we also find that the amplitude dip in the middle of the vowels (induced by glottalization) differentiates rearticulated vowels from modal and checked vowels, while the shorter duration differentiates checked vowels from modal and rearticulated vowels. We conclude that in YZ, mid-phased glottalization is the defining feature of rearticulated phonation whereas duration is the defining feature of checked vowels. Future studies can test whether YZ listeners are also more sensitive to amplitude dip and short duration when perceiving rearticulated and checked vowels.

4aSC40. Acoustic correlates of aspirated consonants in Maranao. Jason W. Lobel (Linguist., Univ. of Hawai'i at Mānoa, Honolulu, HI), Erik R. Thomas (Dept. of English, North Carolina State Univ., Box 8105, Raleigh, NC 27696-8105, erthomas@ncsu.edu), Jeff Mielke (English, North Carolina State Univ., Raleigh, NC), and Labi H. Riwarung (Mamitua Saber Res. Ctr., Mindanao State Univ., Marawi, Philippines)

Aspirated obstruents are rare in Austronesian languages, one exception being the southern Philippine language Maranao, as reported by Lobel and Riwarung [*Oceanic Linguist*. **48**, 403–438 (2009)]. In Maranao, aspirated consonants occur as a reflex of a cluster of a former voiced stop and a homorganic obstruent (*bp > p', *dt > t', *ds > s', *gk> k '). The most obvious correlate to non-Maranao speakers is a dramatic raising of the following vowel, which also occurs after voiced obstruents, but not after historic single voiceless obstruents—e.g., /təkaw/ [təkaw] 'startled' (earlier *təkaw) vs. /tək'aw/ [təkʰɤw] 'thief' (earlier *təgkaw) However, native Maranao speakers regard the raising as a property of the consonants, not the vowels. We examined the correlates of the apparent aspiration. The vowel raising is realized robustly and consistently, with some overlap in F_1/F_2 space among contrastive vowels. However, aspirated and unaspirated stops also show differences in VOT and in measures of breathiness of the following vowel, albeit with somewhat less consistency. Differences between /s/ and /s'/ were not evident except for realizations of following vowels. We explore the role of pharyngeal expansion due to voicing in the development of these Maranao segmental realizations.

4aSC41. Gemination variation in three varieties of regional Italian. Veronica Miatto (Linguist., Stony Brook Univ., SBS Stony Brook University, Stony Brook, NY 11790, veronica.miatto@stonybrook.edu), Amber Leon, Tia Rosales, and Kaitlin Stephen (Linguist., Stony Brook Univ., Stony Brook, NY)

Consonant gemination in Italian is phonologically distinctive, being acoustically realized with consonant closure duration and duration of the preceding vowel as primary cues. Southern varieties of Italian have been claimed to geminate more than Northern varieties. The purpose of these study is to analyze variation in gemination in two Northern varieties of Italian (Veneto Italian and Friuli-Venezia Giulia Italian) and Neapolitan Italian, spoken in the South. Moreover, generational variation will be analyzed for Venetian speakers. The study involved 40 speakers: 10 young speakers per region and 10 older speakers from Veneto. The speakers read aloud bisyllabic Italian words with voiceless stop geminates or singletons. Results

show that there is no significant difference in gemination between younger (VY) and older Venetians (VO). This is surprising, as VO are more proficient in Venetian, a non-geminating language. Moreover, while VY and Friulian (F) speakers geminate similarly, a significant difference was found with how Neapolitan (N) speakers geminate, but not on whether N speakers geminate more than VY and F speakers. In fact, while VO and F geminate by lengthening the consonant closure more than shortening the preceding vowel, N speakers appear to shorten the preceding vowel much more than lengthening the consonant closure.

4aSC42. Acoustic properties of subtypes of creaky voice. Patricia A. Keating (Linguist., UCLA, 3125 Campbell Hall, UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu), Marc Garellek (Linguist., UC San Diego, La Jolla, CA), Jody Kreiman (Head and Neck Surgery, UCLA, Los Angeles, CA), and Yuan Chai (Linguist., UCLA, Los Angeles, CA)

"Creaky voice" is a term that covers multiple kinds of voicing, and there is no single defining acoustic property shared by all subtypes of creaky voice. Here we explore the distinct characteristics of each subtype. We identify three main properties of creaky voice: low f0, irregular f0, and constricted glottis (as shown by electroglottography). Prototypical creaky voice has all of these properties; other subtypes are characterized by different subsets of properties. We will describe, with reference to previous literature, multiply-pulsed creak (a special case of irregular f0, along with low f0), open-glottis creak (low and often irregular f0, but unconstricted glottis), vocal fry (low but regular f0, constricted glottis), and creak with such irregular pulsing that no f0 can be recovered. Building on our previous work [Keating *et al*. 2015 Proc. ICPhS], we show how various acoustic measures pattern for each subtype. Results from parametric speech synthesis provide support for our acoustic observations.

4aSC43. The status of lexical stress in understudied Kabyle Tamazight-Berber: "Acoustic evidence from Kabyle dominant speakers." Dehbia Gaoua (Linguist., Boston Univ., 1071 Beacon St., Brookline, MA 02446, dalilag@bu.edu)

Until now, the classification of the prosodic systems of some languages (including Berber languages) is not conclusive and open to alternative interpretations. The question is: are these languages better classified as lacking stress or more controlled and systematic studies of these languages are needed? This study asks three main questions: (1) Is there lexical stress in Kabyle?, (2) what are the acoustic properties that manifest stress (if it turned out "it exists")? and (3) which syllable in the word is prominent? Results from a closely controlled acoustic production study of 6 male Kabyle-native speakers, showed some evidence of lexical prominence in root words, which was manifested by intensity, duration (for vowel /æ/), and vowel quality (higher and more peripheral prominent syllable for /æ/ vowels). Louder, and higher and more peripheral vowels were observed in initial syllables while longer vowels (/æ/) were found in penultimate syllables. Surprisingly, based on the present data we did not see strong evidence of F0 being a meaningful correlate; however, we observed acoustic patterns consistent with an enhancement of the initial syllable corresponding to the focus condition, which led us to suggest that the pitch results looked like an intonational pitch accent that is location-sensitive. Implications and further work are discussed in the paper.

4aSC44. Towards improving automatic speech recognition for underrepresented dialects with data augmentation. Sarah Bakst (SRI Int., 333 Ravenswood Ave., Menlo Park, CA 94025, sarah.bakst@sri.com), Emre Yilmaz, and Diego Castan (SRI Int., Menlo Park, CA)

Speech technology harnessing information from a speaker's voice promises to enhance security and assist in everyday tasks. Automated speech recognition (ASR) converts spoken words into text, facilitating interaction with electronic devices. ASR is also increasingly used in education in programs that assess students' learning through interaction with computers. However, ASR may not work equally well for underrepresented accent groups. Multiple studies over the last several years (e.g., Tatman 2017, Koenecke *et al*., 2020) have shown that ASR performs particularly poorly on African American English (AAE). This performance drop is likely due to imbalances in accent representation in training data. Here, we assess vocal tract length adjustment as a data augmentation method for increasing representation of AAE speech in the training data, with the aim of improving ASR performance on AAE. We compare this data augmentation method to standard data augmentation methods (e.g., environmental).

Session 4aSP

Signal Processing in Acoustics, Biomedical Acoustics, Architectural Acoustics, Physical Acoustics, and Underwater Acoustics: Acoustical Imaging, Reconstruction, and Localization IV: Underwater Applications (Hybrid Session)

Yangfan Liu, Cochair
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Efren Fernandez Grande, Cochair
Technical University of Denmark,

Chair's Introduction—8:30

Invited Paper

8:35

4aSP1. Imaging ocean water columns by acoustic contrast reflection signals in existing marine seismic data. Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38655, zhang@olemiss.edu)

Marine seismic surveys conducted for oil and gas exploration use a source-receiver system consisting of airgun sources and hydrophone receiver arrays. The collected data were commonly used to image subsurface beneath the seafloor. The surveys had also collected signals of acoustic reflections from ocean water columns due to acoustic impedance contrast by temperature and salinity. These signals can be processed to provide high-resolution images of the water-column structures such as eddies and internal waves, forming a discipline termed as seismic oceanography in the literature. Acoustic contrast reflection signals in a three-dimensional seismic survey, collected by multiple parallel receiver arrays, can even be used to explore movements of water columns {see our recent study in [Zou, *et al*., *Ocean Sci*. **17**(4), 1053–1066 (2021)]}. Nevertheless, because of the low acoustic impedance contrast in oceans, these water-column reflection signals are extremely weak, e.g., about 2–3 orders weaker than the seafloor reflections. Meanwhile, these signals consist of acoustic multipath features that require an appropriate filtering of the signals (see our recent study in the study by Zou and Zhang [*JASA* **150**(5), 3852–3860 (2021)]. Understanding of the signal features and appropriate applications of acoustic signal processing methods are inspired to improve the imaging quality.

Contributed Paper

8:55

4aSP2. Reducing ambiguities in single-hydrophone matched-field processing by exploiting source motion. Margaret Cheney (Mathematics and ECE, Colorado State Univ., Fort Collins, CO) and Ivars Kirsteins (NUWC, 1176 Howell St. Bldg. 990, Newport, RI, ivars.p.kirsteins@us.navy.mil)

Matched-field processing for localizing an underwater acoustic source in range and depth from power-spectrum measurements obtained at a single hydrophone receiver often suffers from high side-lobes and ambiguities when the signal is narrowband or signal bandwidth is small. In this paper we review how source motion can be utilized to reduce side-lobes in depth and range via an incoherent synthetic-aperture-like approach and explain the mechanisms responsible for side-lobe reduction relative to the height of the main-lobe by using a normal-mode expansion for the pressure field. We also derive an approximation for the depth main-lobe width when the true source range is known for the ideal rigid-waveguide case. Numerical results are presented corroborating the analytical analysis along with some matched-field localization ambiguity surface examples for (a) an ideal shallow-water waveguide with a pressure-release top boundary and a rigid bottom boundary and (b) a more realistic shallow-water Pekeris environment, to demonstrate how side-lobes and ambiguities are reduced when source motion is exploited in the matched-field processing.

Invited Paper

9:10

4aSP3. Near-field acoustic holography for underwater moving surfaces. Nicolas P. Valdivia (Acoust. Div., Naval Res. Lab., Code 7130, Washington, DC 20375, nicolas.valdivia@nrl.navy.mil)

Near-field acoustic holography has been the standard technique to accurately image static structure vibration from nearby acoustic pressure measurements. The classical application bases the method on the stable solution of an integral surface representation of the acoustic pressure. In this work, we will be concerned with the extension of NAH to image a vibrating structure moving underwater. In an underwater medium, even at slow speeds, the action between the flow and structure produces many radiation sources that could severely limit the accuracy of the integral surface representation used for static NAH. Using the Lighthill acoustic analogy, we use an integral surface representation for the structure's radiated sound and a quadrupole source that models the corresponding flow-induced sound. Finally, we will justify the validity of the proposed method for COMSOL numerically generated pressure data that results from the internal excitation of a cylindrical structure moving over an underwater medium. This work has been supported by the Office of Naval Research.

Contributed Papers

9:30

4aSP4. Active sonar detection with acoustic spiral waves: experiment and modeling. Michael A. Bisbano (ATMC/ECE, Univ. of Massachusetts Dartmouth, Fall River, MA) and David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsDB@gmail.com)

Active spiral-wave active localization is a technique for determining location of acoustic targets [after Dzikowicz *et al.*, *JASA* **146**, 4821–4830 (2019)], which relies on measuring the phase difference between a reference and spiral wavefronts. This research presents a monostatic model of spiral detection and tracking for multiple targets in sparse noise limited environments. Simulation and experimental results from an acoustic spiral wave beacon in a small underwater test tank will be presented. Results show good agreement to resolve targets in horizontal and vertical dimensions with a compact aperture. [Work supported by ONR 321MS.]

9:45

4aSP5. Creating acoustic spiral waves with quadrature and tertiary phase-biased dipoles. Avery A. Jackson (ATMC/ECE, Univ. of Massachusetts Dartmouth, Fall River, MA) and David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsDB@gmail.com)

Acoustic spiral waves have been demonstrated by exciting orthogonal modes of vibration in cylinders with just one electroacoustic transducer [*J. Acoust. Soc. Am.* **132**(6) (2012)]. Applications including navigation, active sonar, and imaging are emerging. We demonstrate several practical engineering approaches to create constituent signaling with either quadrature or tertiary phase bias to excite spiral waves and compare their effectiveness. Approaches using CODECs, DSP, and flip flops are designed and characterized then tested on underwater transducers in realistic operating conditions. Tradeoffs of source level, effective coupling, bandwidth, and radiation patterns are summarized. [Work supported in part by BTech Acoustics LLC and ONR 321MS.]

Invited Papers

10:00

4aSP6. Numerical modeling of the 2D hologram of the low-frequency moving source by using horizontal antenna in a shallow water waveguide. Sergey A. Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation), Sergey Tkachenko, Nikolai Ladykin, and Dmitry Sotnikov (Mathematical Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

The 2D holographic signal processing method by using a horizontal linear antenna (HLA) in shallow water is considered in the paper. The offered method of moving broad band source sound field hologram formation by using HLA is based on holographic signal processing of each element of HLA. The sound source moving in shallow water waveguide creates a stable interference pattern (interferogram) in the frequency-time domain on each HLA element. The 2D-FT (2D Fourier Transformation) of the interferogram is antenna element hologram (AEH). AEH allows to coherently accumulate the sound intensity of the interferogram in a small area as focal spots for each HLA element. It is shown in paper that the focal spots localization in AEH of HLA depends upon the source range, velocity, and motion direction [S. Pereselkov and V. Kuz'kin, *JASA* **151**(2), 666–676). The output HLA hologram is formed by superposition AEH. Relationship between focal spot coordinates on the HLA hologram and the source range, velocity, and motion direction are derived in the paper. The results of the numerical experiment of HLA hologram formation for low-frequency (100–400 Hz) sound field of source moving in shallow water are presented. The results presented in the paper significantly expand the efficiency of interferometric signal processing in shallow water. [This study was supported by RFBR 19-29-06075, MK-4846.2022.4.]

10:20–10:35 Break

10:35

4aSP7. Experimental formation of the 2D hologram of high-frequency source by a vector receiver in shallow water. Sergey A. Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation), Sergey Tkachenko, Nikolai Ladykin, Dmitry Sotnikov, and Alexey Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

The results of a high-frequency experimental formation of the 2D hologram of a moving noise source using a cylindrical small-sized vector-scalar antenna are presented. The purpose of the experiment is to test the efficiency of the interferometric method in the high-frequency domain (1–15 kHz) for noise source localization. The experiment was carried out in a shallow water area with a water layer depth 87 m. The source localization method is based on processing of its interferogram. The 2D-FT (2D Fourier Transformation) of the interferogram is hologram of moving source. 2D hologram allows to coherently accumulate the sound intensity of the interferogram in a small area as focal spots in hologram domain. The focal spots localization in hologram domain depends upon the source range, velocity, and motion direction. The time dependences of source range, its velocity and motion direction are estimated on based of holographic signal processing. The comparative analysis of theoretical dependences and experimental data is carried out. [This study was supported by RFBR 19-29-06075, MK-4846.2022.4.]

Contributed Papers

10:55

4aSP8. Data-model comparison of elastic target scattering for a shallow water sonar system for buried UXO detection. Jason Philtron (Penn State Univ., ARL Bldg., University Park, PA 16802, jhb186@psu.edu), Daniel C. Brown, Shawn Johnson, Kyle S. Dalton, Geoff Goehle, and Thomas E. Blanford (Penn State Univ., State College, PA)

Successful buried unexploded ordnance (UXO) detection in shallow water (<3 m) has been demonstrated using a sonar system deployed from a shallow-draft surface vessel. Data was collected over several years and during different seasons in a freshwater reservoir near Howard, PA, USA. The target test field contains a variety of science (e.g., sphere, cylinder) and other objects. Previously, a point-based sonar scattering model that simulates calibrated time-series sonar data due to sediment surface and volume scattering effects was shown to provide simulated data of the environment. Recently, object scattering models have been added to the simulation. The results of a model-data comparison of objects in the environment will be presented. Realistic simulation results containing objects will increase understanding of detection capability in different geometries (e.g., depth) and environments (e.g., sediments, layering).

11:10

4aSP9. Array invariant-based range-only target motion analysis for ship noise measured by two vertical line arrays in shallow water. Hoseok Sul (Marine Sci. & Convergence Eng., Hanyang Univ. ERICA, Sci. ans Technol. Buliding 1, 237, 55, Hanyangdaehak-ro, Sangnok-gu, Ansan-si, Gyeonggi-do 15588, South Korea, sulhosuk@hanyang.ac.kr), Moon Ju Jo (Marine Sci. & Convergence Eng., Hanyang Univ. ERICA, Ansan, South Korea), Taek Lyul Song (Dept. Electron. Systems Eng., Hanyang Univ. ERICA, Ansan, South Korea), and Jee Woong Choi (Marine Sci. & Convergence Eng., Hanyang Univ. ERICA, Ansan, South Korea)

A novel approach for underwater acoustic localization by using passive line array sonar will be proposed in this talk. In the case of the received signal having a finite pulse duration, the target location can be easily estimated using the time-difference-of-arrival (TDOA) method between sensors. However, it is difficult to use the TDOA method for signals that exist for a relatively long time, such as ship noise. An array invariant method has been proposed to measure the source-receiver distance, but it does not accurately estimate the target location. In this talk, target localization will be performed with the range-only target motion analysis method using non-linear target tracking after estimating the target range by the array invariant method. The proposed method is tested using the ship noise of the R/V *Onnuri* measured by two vertical line arrays during the Shallow-water Acoustic Variability Experiment (SAVEX-15), and the result will be discussed in this talk. [This research was supported by the Ministry of Oceans and Fisheries of Korea (PM63010, 2022).]

11:25

4aSP10. Maximum likelihood estimation for leak localization in water distribution networks using in-pipe acoustic sensing. 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), Stan Fong (Digital Water Solutions, Guelph, ON, Canada), Dirk Friesen (Univ. of Waterloo, Waterloo, ON, Canada), and Sriram Narasimhan (Civil and Environ. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

With drinking water resources rapidly depleting with time, water conservation efforts have received special emphasis, especially in arid regions like California. One of the major sources of unused water expenditure is inconspicuous leaks in underground water distribution networks (WDN), making it highly essential to quickly detect and localize them. The leak detection and localization problem has been widely studied for a straight pipeline system, however, estimating the leak location in a pipe network remains largely unexplored. In this study, we measure the acoustic pressure signals inside a pipe network at multiple locations using state-of-the-art hydrophone-enabled devices. To localize the leaks in pipe networks, we propose maximum likelihood estimation, which has previously shown high efficacy in localizing mobile devices in a cellular network. In this approach, the cross-correlation of the filtered signals from different sensor pairs yields multiple time delays corresponding to multiple acoustic paths traversed by the leak noise in the pipe network, which is more difficult to solve compared to a straight pipe system. The leak location is then identified by maximizing a conditional probability distribution function of the distance between the sensor and the leak location.

Invited Paper

11:40

4aSP11. Low-frequency holographic signal processing for moving sources separation in shallow water waveguide. Sergey A. Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Sci. Ctr. for Wave Res., General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation), Nikolai Ladykin, Dmitry Sotnikov, and Alexey Pereselkov (Mathematical Phys. and Information Technol., Voronezh State Univ., Voronezh, Russian Federation)

The results of numerical modeling of the interferometric signal processing for broadband sources (200–300 Hz) moving in shallow water waveguide is presented in the paper. Each source creates in receiver point a stable interference pattern of the intensity distribution (interferogram) in the frequency-time domain. The 2D Fourier Transformation (2D-FT) is applied to analyze the superposition of several interferograms of the sources moving in waveguide. 2D-FT of interferogram—hologram is superposition of the holograms of the sources moving in waveguide. The source hologram allows to coherently accumulate the sound intensity of the source interferogram in a relatively small area as focal spots. The sources can be separated when the sources holograms spots are not overlapped and thus can be filtered. The information about each source is in separated hologram. So, separated hologram allows to estimate source range, its velocity and direction (S. Pereselkov and V. Kuz'kin, *JASA* **151**(2), 666–676). Within numerical experiment the separation of three noise sources with different intensities moving in different directions with different velocities is considered.The error of estimation of parameters of each source is used as a criterion of resolving efficiency of proposed method. [This study was supported by RFBR 19-29-06075, MK-4846.2022.4.]

4a THU. AM

Session 4aUW

Underwater Acoustics, Acoustical Oceanography, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics: Acoustic Methods for Unmanned Mobile Platforms: Sensing, Localization, and Communications I

Martin Siderius, Cochair
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Aijun Song, Cochair
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Electrical and Computer Engineering, Georgia Insitute of Technology, 85 Fifth Street NW, Atlanta, GA 30332

Chair's Introduction—8:30

Invited Papers

8:35

4aUW1. Acoustic receiving on glider-type autonomous underwater vehicles: Advantages and challenges. Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu)

An ocean glider is a specific type of Autonomous Underwater Vehicle (AUV) that operates using a buoyancy engine rather than traditional propellors or thrusters. This method of propulsion enables a glider to be deployed for weeks or months and is relatively quiet, making the ocean glider a desirable platform for persistent acoustic monitoring. A glider typically remains submerged for several hours, during which time it can be challenging to localize the vehicle precisely. Gliders are low-power platforms that operate in the mid-water column, and therefore subsea navigation technologies implemented on other types of AUVs, such as inertial navigation systems aided by Doppler velocity logs, may not be suitable. Acoustic signals from fixed broadband sources have been used for subsea localization of Seaglider, a commercially available glider platform. Measurements of the multipath acoustic arrival structure received on Seagliders at ranges up to hundreds of kilometers from transmitting sources were used for vehicle localization in both temperate and polar underwater sound propagation environments. Methodology and results for subsea localization of the Seaglider platform will be presented and placed in the context of a broader discussion of the advantages and challenges specific to the glider as a moving acoustic receiving platform.

8:55

4aUW2. Localization of an acoustic autonomous underwater vehicle using multi-channel back-propagation methods. Natalie Kukshtel (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, nkukshtel@whoi.edu), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanographic Inst., Woods Hole, MA)

Autonomous underwater vehicles (AUVs) are extremely useful tools for studying the acoustics of complex ocean environments due to their ability to detect environmental changes with greater spatial resolution than fixed moorings. During the New England Shelf Break Acoustics (NESBA) experiments in May 2021, an AUV system was deployed to collect acoustic data for investigating the local biological, physical, and geological oceanography. This acoustic AUV system was comprised of a modified REMUS 600 vehicle, a hull-mounted 3.5 kHz transducer, and a towed multi-channel hydrophone array. Along mission profiles where the AUV is fully submerged but too shallow for bottom-lock navigation, one challenge is accurate localization of the AUV. Localization was performed in post-processing using multi-channel back-propagation methods applied to AUV source signals received at mooring hydrophones in the NESBA network as well as ship-towed sound source signals received at the AUV-towed array. Uncertainty in the localization estimates due to spatiotemporal sound speed changes was investigated, and hydrophone mooring tilt angle was determined by minimizing the localization uncertainty. Following localization, this AUV acoustic data was used to investigate local seafloor sub-bottom properties and the acoustic effects of biological scattering layers and varying physical oceanography. [Work supported by the Office of Naval Research.]

9:15

4aUW3. Acoustic beacon aided underwater vehicle navigation: Observability analysis on vehicle trajectory. Jae Won Choi (Univ. of Illinois Urbana-Champaign, 1308 W. Main st., Urbana, IL 61801, choi223@illinois.edu), Girish Chowdhary, and Andrew C. Singer (Univ. of Illinois Urbana-Champaign, Urbana, IL)

In the GPS-denied underwater environment, unmanned underwater vehicle navigation commonly depends on integrating the outputs of an inertial measurement unit (IMU) and Doppler Velocity Sonar (DVS) for recursive self-localization. However, dead reckoning navigation based on only an IMU and DVS suffers from the accumulation of errors over time. Analogous to using a GPS signal from a satellite, an acoustic signal from a source from a known location can be used to measure time-of-flight (distance) and motion-induced Doppler (relative velocity) to correct for these errors. However, due to the non-linear relationship between the pose and acoustic measurements, analyzing the observability of the positioning system in a traditional sense is nontrivial. For this work, we aim to analyze and design the trajectory of a vehicle that operates in order to ensure a form of observability. This approach provides a sufficient condition for the existence of a one-to-one relationship between the state of the system and the observed series of measurements. By analyzing this form of observability over possible trajectories, our goal is to find a vehicle path plan that can yield a unique pose based on measurements from the onboard sensors and received acoustic signals.

9:35

4aUW4. Ship noise quantification of fisheries vessels on the Great Lakes. Steven Senczyszyn (Great Lakes Res. Ctr., Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, sasenczy@mtu.edu), Andrew Barnard (Acoust., Penn State, University Park, PA), Erik Kocher (Great Lakes Res. Ctr., Michigan Technol. Univ., Houghton, MI), Thomas M. Evans (DNRE, Cornell, Ithaca, NY), Lars G. Rudstam (Natural Resources and the Environment, Cornell Univ., Bridgeport, NY), Suresh A. Sethi (Cornell Univ., Ithaca, NY), Susan E. Wells (U.S. Fish and Wildlife Service, Ithaca, NY), Scott R. Koproski (U.S. Fish and Wildlife Service, Alpena, MI), S. D. Hanson (U.S. Fish and Wildlife Service, New Franken, WI), David M. Warner (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI), Benjamin Turschak (Great Lakes Sci. Ctr., U.S. Geological Survey, Charlevoix, MI), Daniel L. Yule (Great Lakes Sci. Ctr., U.S. Geological Survey, Ashland, WI), Steven A. Farha, Timothy P. O'Brien (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI), Kevin N. McDonnell (Great Lakes Sci. Ctr., U.S. Geological Survey, Alpena, MI), James M. Watkins (Cornell Univ., Ithaca, NY), Mark R. Dufour, Patricia M. Dieter (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI), James J. Roberts (Great Lakes Sci. Ctr., U.S. Geological Survey, Sandusky, OH), and Peter Esselman (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI)

Prey fish abundances in the Great Lakes are a driver for several agencies' commitments to the Council of Lake Committees to support fisheries management. These management decisions have profound economic and social impacts within the Great Lakes region. Fisheries estimates done by echosounders or trawling may be biased due to the propagated noise from large fisheries vessels. In the first two years of a four-year collaborative study, crewed fisheries vessels and uncrewed Saildrone vessels were used to compare abundance estimates between "loud" and "quiet" vessels. To quantify the effects of ship noise on prey fish abundance estimates, a mobile ship noise measurement system was designed and deployed to measure radiated acoustic signatures of several ships in the Great Lakes. This talk will discuss the deployment of a mobile underwater acoustic test range and show results of ship noise measurements from the first two years of the program. In addition, plans for future data generation, analysis, and comparison between crewed and autonomous systems will be discussed and an overview of the effects of ship noise on fish avoidance will be discussed in the proceeding talk.

9:55–10:10 Break

10:10

4aUW5. Quiet uncrewed surface vessel assesses fish avoidance of motorized survey ships in the Great Lakes. Thomas M. Evans (DNRE, Cornell, Fernow Hall, Ithaca, NY 14853, tme33@cornell.edu), Lars Rudstam (DNRE, Cornell, Ithaca, NY), Suresh A. Sethi (NY Coop Fish and Wildlife Res. Unit, USGS, Ithaca, NY), Andrew Barnard (Acoust., Penn State, University Park, PA), Scott R. Koproski (Alpena Fish and Wildlife Conservation Office, USFWS, Alpena, MI), S. D. Hanson (Green Bay Fish and Wildlife Conservation Office, USFWS, New Franken, WI), Daniel L. Yule (Lake Superior Biological Station, USGS, Ashland, WI), David M. Warner (Great Lakes Sci. Ctr., USGS, Ann Arbor, MI), Susan E. Wells (Green Bay Fish and Wildlife Conservation Office, USFWS, Ithaca, NY), Benjamin Turschak (Charlevoix Fisheries Res. Station, Michigan DNR, Charlevoix, MI), Steven A. Farha, Timothy P. O'Brien (Great Lakes Sci. Ctr., USGS, Ann Arbor, MI), Kevin N. McDonnell (Alpena Fish and Wildlife Conservation Office, USFWS, Alpena, MI), James M. Watkins (DNRE, Cornell, Ithaca, NY), Mark R. Dufour, Patricia M. Dieter (Great Lakes Sci. Ctr., USGS, Ann Arbor, MI), Erik Kocher (Dept. of Civil and Environ. Eng., MTU, Houghton, MI), James J. Roberts (Great Lakes Sci. Ctr., USGS, Sandusky, OH), Steven Senczyszyn (Mech. Engineering-Eng. Mech., Michigan Technol. Univ., Houghton, MI), and Peter Esselman (Great Lakes Sci. Ctr., USGS, Ann Arbor, MI)

Acoustic surveys of fish are a foundational component of many fisheries monitoring programs, including surveys in the Great Lakes. These surveys are conducted with traditional crewed and motorized vessels, but fish avoidance of these types of platforms has been reported in multiple studies, potentially biasing estimates. A quiet uncrewed surface vessel, Saildrone, was equipped with a 120 kHz Simrad EK80 transducer and deployed in Lakes Huron, Michigan, and Superior in the summers of 2021 and 2022. The drone was then overtaken by numerous motorized vessels using transducers with the same frequency. The average target depth, target strength, and nautical area scattering during overtakes were compared. We looked for a fish behavioral response with General Additive Models using distance from the Saildrone to the vessel as the predictor. We also compared the effectiveness of acoustic surveys from each platform with analyses of variances over 2 km sections of the overtake. Fish showed a limited response to approaching vessels, and acoustic estimates of fisheries relevant measures were similar between Saildrone and motorized vessels. Findings from this work will inform interpretation of acoustic data in the Great Lakes and provide the largest scale testing of fish avoidance to acoustic surveys to date.

10:30

4aUW6. Performance study of ocean acoustic tomography methods in the upper-ocean environment using autonomous platforms. Etienne Ollivier (Mech. Eng., Georgia Inst. of Technol., 1040 Hemphill Ave. NW, Atlanta, GA 30318, eollivier3@gatech.edu), Karim G. Sabra (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), and Nicholas Durofchalk (School of Phys., Naval Post-Graduate School, Monterey, CA)

Accurate knowledge of the spatial and temporal evolution of the three-dimensional ocean sound speed profile (SSP) is crucial for underwater source localization. Ocean acoustic tomography (OAT) methods aim at reconstructing SSPs variations based on acoustic measurements between multiple source-receiver pairs (e.g., eigenrays arrival times). This study investigates the estimation of range-dependent SSPs using a classical model-based OAT method (i.e., ray-based ocean acoustic tomography), for various configurations of source and receiver configurations using autonomous platforms in a highly-dynamic upper ocean environment. A regional oceanographic simulation of the De Soto Canyon circulation in the Gulf of Mexico is used to construct 3D sound speed variations spanning a month long which exhibits significant sub-mesoscale variability. Two main aspects affecting OAT performance in the presence of high 3D SSPs variability are investigated: (1) The influence of the input acoustic data (i.e., source-receivers configuration and platform motion, arrival-times accuracy...) and (2) the actual implementation of the OAT scheme (i.e., selection of complexity reduction basis, linearized forward model assumptions as well as the use of iterative solvers) on SSPs estimations errors. Practical implications for the design of OAT experiments in the dynamic upper ocean will be discussed. [Work supported by the Office of Naval Research.]

10:45

4aUW7. Bayesian navigation in shallow water using passive acoustics. Junsu Jang (Scripps Inst. of Oceanogr., UC San Diego, 9679 Caminito Del Feliz, San Diego, CA 92121, jujang@ucsd.edu) and Florian Meyer (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

Autonomous underwater vehicle (AUV) navigation relying on active acoustic sources causes noise pollution, while dead reckoning leads to a localization error that increases with time. Therefore, AUV navigation based on passive acoustics is appealing. However, for AUV navigation, extracting location information with passive acoustics is a challenging signal processing task. Due to the small form factor of an AUV as a sensing platform, only a single hydrophone or a small aperture hydrophone array can be used as an acoustic sensor. Furthermore, the acoustic signals originate from uncooperative sources. Here, we propose a Bayesian navigation approach for an AUV that exploits acoustic signals generated from sources of opportunity (SOOs) in a shallow water environment. The waveguide invariant (WI) parameter is estimated from cross-correlation coefficients of non-linearly transformed tonal signals of an SOO. It is assumed that the location information of the SOO is transmitted by an automatic identification system. Additionally, the range rate is inferred using the spectrum of cross-correlated acoustic fields over a time interval. The WI parameter estimate, the range rate estimate, and inertial measurements are fused in a Bayesian parameter estimation approach. The navigation capability is demonstrated using simulated and real data from the SwellEx-96 experiment.

11:00

4aUW8. Velocity estimation using a compact correlation velocity log. Thomas E. Blanford (The Penn State Univ., State College, PA 16804, teb217@psu.edu), Daniel C. Brown, and Richard J. Meyer (The Penn State Univ., State College, PA)

Unmanned underwater vehicles require bottom-referenced acoustic navigation aids to maintain long-term positional accuracy without surfacing. When these platforms are small, they create new design constraints for acoustic navigation aids because of the limited available space and power. Traditional acoustic navigation techniques, such as Doppler Velocity Logs, are unsuitable for use on small platforms because of the power required to maintain adequate signal to noise ratio when they are scaled in size. A compact correlation velocity log (CVL) is an alternative approach that can meet the power, space, and accuracy requirements for an acoustic navigation aid on such platforms. This device uses a single projector, a sparse receive array, and estimates platform motion using a multi-dimensional fitting algorithm over an ensemble of 3 or more pings. This presentation will discuss the theory of operation, simulation, and experimental results for a 300 kHz compact CVL that is 4×8 cm^2. [The authors want to acknowledge Lockheed Martin Rotary and Mission Systems for their financial support of this work.]

Session 4pAA

Architectural Acoustics, Noise, Engineering Acoustics, and ASA Committee on Standards: Sound Data for Sound Design

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

Evelyn Way, Cochair
Maxxon Corp, 920 Hamel Road, Hamel, MN 55340

Chair's Introduction—1:10

Invited Papers

1:15

4pAA1. Perspective on publishing acoustic data: Case studies, limitations, and solutions. Evelyn Way (Res. & Development, Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, evelynway@gmail.com)

There are many considerations for manufacturers publishing acoustic data. Varied audiences, test variability, building codes, liability, and convoluted standard test methods put competing interests on product testing and data publishing. Discussion will use case studies of floor/ceiling isolation data to provide clarity and insight to help the practitioner navigate a complicated data landscape.

1:35

4pAA2. Discussion of impacts of measurement error in architectural acoustics performance specifications. Steve Rittmueller (R&D, Masonite, 1955 Powis Rd., West Chicago, IL 60185, srittmueller@masonite.com)

Customers for acoustic products need manufacturers and professionals to provide performance specifications to allow informed design decisions. Acoustic professionals rely upon standard tests and special labs to quantify products. Data published by the ASTM for E90 shows that an STC rating for a single construction can have a variability range of up to 7 points. Being logarithmic, this equates to a sound pressure range of more than 2×, larger than 100%. The log nature of many acoustical performance metrics hides this from most. This large variability causes challenges for the architectural acoustics community. The primary negative impact: customers do not trust the acoustical ratings manufacturers provide. This is not an issue of the manufacturers providing inaccurate information, it is an issue of under-reported and incomplete understanding of the uncertainty in the metrics themselves. For perspective, NRC measured the Young's Modulus of OSB material to have a mean of 6.8×10^9 and a standard deviation of 1.5×10^8 (Ref: NRC IR-766, Table 22). This yields a measurement error of 3.8%. It is simple to account for this in the design process. If the metric has an uncertainty as large as the acoustic metrics, the standard paradigm for accounting for error not only breaks down, it incentivizes bad actors: cherry picking, lab shopping, etc. More open acknowledgement of the measurement error in architectural acoustic metrics would increase the confidence customers have in them and would allow designers to make more informed decisions.

1:55

4pAA3. Lessons in large-sample laboratory sound transmission loss testing. Benjamin M. Shafer (PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Thousands of laboratory sound transmission loss tests, in accordance with the ASTM International Test Method E90, reveal some trends that challenge multiple aspects of building sound isolation. Some notions about sound transmission through building assemblies are accepted industry-wide that these current studies bring into question. The applicability of the Sound Transmission Class (STC) rating system also becomes challenging when observing trends in large and varied data samples. From preliminary small-sample studies to measure uncertainty across laboratories to large-sample observations of assembly test data, this presentation offers guidance for both the development of standards in the building sound isolation industry and for education in the building design process.

2:15

4pAA4. Adventures in uncertainty: A former consultant's view of manufacture-driven acoustical testing. Michael Raley (PAC Int., 2000 4th Ave., Canby, OR 97013, mraley@pac-intl.com)

Acoustical consultants commonly rely on laboratory testing of wall and floor/ceiling assemblies to inform their designs and to evaluate noise control solutions. Today, much of the available laboratory testing comes from manufacturers of noise control products. Laboratory tests are taken as indicative of assembly and product performance, but there is significant uncertainty inherent in the ASTM E90 and E492 test methods. When evaluating test results to design assemblies or compare products, this uncertainty is often given insufficient attention and test results are taken as de facto indicators of assembly performance. Manufacturers are faced with how to handle this uncertainty when deciding what to test, where to test, how to test, and ultimately what results to make public. Manufacturers are often accused of "cherry-picking" test data, but the reality is often more complicated than that. In this presentation, the author will discuss his experience working at product manufacturers and deciding what to test and what data to make public. The presentation will also look at some examples of how the uncertainty in laboratory testing could affect the outcome of the tests. Finally, the author will discuss other ways of presenting test data that aim to reduce some of the uncertainty and make it easier for acousticians to make good design decisions.

2:35

4pAA5. Which number is correct? Making sense of acoustical test results. Andrew Schmidt (USG Corp., 700 N Hwy. 45, Libertyville, IL 60048, ASchmidt@usg.com) and Austin Phillips (USG Corp., Libertyville, IL)

Acoustical consultants and designers rely on sound data published by manufacturers to make informed design decisions. Even when manufacturers make efforts to communicate these data clearly, acoustical test values can be misinterpreted or misapplied. Nuanced sample mounting methods, specific test system configurations, and variability inherent in laboratory test methods are all factors that can affect test results and how or what data are published in manufacturer literature. Since at least the early 1950s, United States Gypsum (USG) has tested and published sound data for its products and systems. A brief history of USG's acoustical testing and sound-related publications will be presented, along with examples of variability of results, influencing factors, and how this information is communicated to different audiences.

2:55

4pAA6. Laboratory perspectives on the business of acoustical materials testing. Eric P. Wolfram (Riverbank Acoust. Labs., 1512 S Batavia Ave., Geneva, IL 60134-3302, eric@riverbankacoustics.com)

Eric Wolfram will share experiences from 10 years as Laboratory Manager for Riverbank Acoustical Laboratories in Geneva, IL. He will start with an overview of the typical process for acoustical materials testing, important details to consider when reviewing test reports, and common concerns raised by manufacturers regarding current industry practices. Eric will share perspectives on how acoustical test data is used for product development and marketing.

3:15–3:30 Break

3:30

4pAA7. A discussion addressing the challenges to making laboratory acoustic data accurate, usable, and specifiable. Kenneth W. Good (Armstrong World Industries, Inc., 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

One might think that accurately measuring and reporting acoustic data would be well established, and in many cases this is true. However, laboratories, manufactures, and other's often face unique situations that present difficulties producing meaningful data and communicating in a useable way. A recent example involves sound absorption for discrete absorbers. This paper will discuss observations used to face these challenges.

3:50

4pAA8. Challenges in tuning the measurement performance of a reverberation chamber. Aldo Glean (Saint-Gobain Res. North America, 9 Goddard Rd., Northboro, MA 01532, aldo.glean@saint-gobain.com), Stanley D. Gatland II (Saint-Gobain North America, Malvern, PA), and Jerry G. Lilly (JGL Acoust., Inc., Issaquah, WA)

Round-robin testing determined that a high degree of reproducibility exists between laboratories that conduct sound absorption measurements in accordance with ASTM C423. Factors that include room dimension, source location and diffuser type have been identified as sources for differences in measurement results. Such lab to lab testing variation presents a challenge when reverberation chambers are replaced with new design and construction features that are not identical. An iterative process is required to produce measurements that match historical results. The presentation will describe the design approach utilized to construct a new reverberation room with different dimensions, diffuser types and materials. Experimental data will be presented showing the results of modifications made to the chamber after achieving initial qualification metrics. The iterative process provided an opportunity to study the impact of diffuser type and surface area and other properties on reducing the performance gap between the two reverberation rooms.

4:10

4pAA9. Acoustic ceiling systems inside buildings: What we were taught and should now know instead. Gary Madaras (Acoust., Rockfon, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com)

This presentation is an executive summary of how modular, suspended, acoustic, ceiling systems perform inside buildings as presented at meetings of the Acoustical Society of America and noise control engineering conferences between 2015 and 2022. It bridges the gap between laboratory testing of ceiling panel metrics and how complete ceiling systems perform inside buildings when combined with other building elements such as lights, air diffusers, floor slabs, plenum barriers and mechanical devices in the plenum. Ceiling system performance will be discussed from the perspectives of complying with minimum sound absorption, minimum sound isolation and maximum background noise level requirements in building design standards and guidelines. Corroboration with foundational testing conducted by the National Research Council of Canada and ASHREA will be integrated. Standards with prediction methods that should be used instead of ceiling panel metrics will be reviewed. Citations to studies will be provided for more detailed information.

4:30

4pAA10. Repeatability in fan sound ratings: Challenges faced in a certification program. Tim Mathson (Lab., AMCA Int., 30 West University Dr., Arlington Heights, IL 60004, tmathson@amca.org)

The Air Movement and Control Association International (AMCA International) maintains a certified ratings program for published air and sound performance for fans. Sound test results can come from AMCA 300 reverberant room testing but are also accepted from manufacturers' accredited labs using other international test standards covering in-duct, enveloping surface, and sound intensity test methods. Observed repeatability between these test methods and labs will be discussed. AMCA 301 covers sound calculation methods, including extrapolation to non-tested sizes and speeds. Requirements and practical recommendations for this conversion will be discussed. Surveillance for the program comes through periodic check tests on manufacturers' products. The check test process provides an overall view of measurement and calculation variance, as well as inevitable manufacturing tolerances and their impact on sound. Finally, the results of round robin testing conducted on centrifugal, tube axial, and vane axial fans at various accredited sound labs will be reviewed.

4:50

4pAA11. Sound rating of catalog air handlers. Derrick P. Knight (Trane Technologies, 2313 20th St. South, La Crosse, WI 54601, Derrick.Knight@TraneTechnologies.com) and Stephen J. Lind (LindAcoustics LLC, Onalaska, WI)

Publishing sound power data for air handlers involves providing data across millions of variations related to operating conditions and unit configurations. AHRI 260 provides both the measurement and modeling guidelines used by most HVACR companies in the United States. However, many building designers are unfamiliar with the details of how a finite set of measurements taken in accordance with AHRI 260 becomes published sound rate data used to design quiet buildings. A case study based on a real product release is used to clearly define this process.

5:10

4pAA12. Dangers of applying fan manufacturer data to equipment sound ratings. Paul F. Bauch (Appl. Equipment, Johnson Controls, 100 JCI Way, York, PA 17406, paul.f.bauch@jci.com) and Roger L. Howard (Appl. Equipment, Johnson Controls, York, PA)

Equipment sound ratings are a valuable tool for consulting engineers; however, the basis of these ratings is not always clear. Custom AHU manufacturers most commonly utilize fan manufacturer data in accordance with AMCA 301 Methods for Calculating Fan Sound Ratings from Laboratory Test Data, and the AMCA 311 Certified Ratings Program. This paper will compare fan manufacturer data to test data collected in accordance with the mapped sound rating approach detailed in AHRI Standard 260, discuss possible sources of error in applying this data, and propose a best practice when comparing ratings from multiple custom AHU suppliers.

Session 4pAB

Animal Bioacoustics: Contributions of Expert Subjects to Animal Bioacoustics

Dorian Houser, Cochair
National Marine Mammal Foundation, 2240 Shelter Island Drive, Suite 200, San Diego, CA 92106

James Finneran, Cochair
Code 56710, NIWC Pacific, 53560 Hull St., San Diego, CA 92152

Chair's Introduction—1:00

Invited Papers

1:05

4pAB1. Heptuna: The most outstanding subject. Patrick Moore (Bioacoustics, National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92101, patrick.moore@nmmf.org)

One of the best experimental collaborators I have ever worked with was HEP, a bottlenose dolphin (*Tursiops truncatus*). HEP was instrumental in leading the way for subsequent dolphin investigations. In the first study on dolphin sound localization, HEP was the willing subject [D. L. Renaud and A. N. Popper, "Sound localization by the bottlenose porpoise *Tursiops truncatus*," *J. Exp. Biol.* **63**, 569–585(1975)]. Thus began a series of critical experiments that set the foundations of cetacean hearing and echolocation. I had the pleasure to partner with Whit Au and HEP to explore the many aspects of dolphin hearing and echolocation. HEP carried the experience of his training and testing to new experiments. If he saw just one response manipulandum, he seemed to know it was a go/no go response requirement, or if two, a forced choice response. HEP was the first animal to use "jaw phones" to test hearing, and interaural time and intensity differences. He demonstrated he could control his echolocation clicks, he provided measures of his transmitting and receiving beam patterns, and much more. Unfortunately, we lost HEP in 2010 after a 40-year history with the U.S. Navy Marine Mammal Program.

1:25

4pAB2. The life semi-aquatic: Harbor seal sprouts and milestones in marine bioacoustics. Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, 115 McAllister Way, Santa Cruz, CA 95060, coll@ucsc.edu)

Harbor seal Sprouts (*Phoca vitulina*) spent 31 years contributing to a wide variety of scientific studies, supporting educational opportunities for students of all ages, and training generations of future scientists, veterinarians, animal care specialists, and teachers. He participated in research projects on associative learning, sensory biology, communication, and physiology, but was perhaps best known for his contributions to marine mammal acoustics. His good-natured cooperation advanced knowledge of sound production and ontogeny, revealed the true amphibious hearing capabilities of seals, and improved our ability to predict the harmful effects of human-generated noise. Sprouts contributed to 34 empirical publications (thus far), many reviews of marine mammal cognition, communication, and sensory biology, and national policy guidelines for marine mammals. Sprouts taught us many things, but perhaps the most important lesson is about the value of individual subjects. While science and conservation often happen at the level of populations, species, and ecosystems, Sprouts showed us that one individual matters and can have a significant impact. He reminds us to find delight and discovery in this complicated, extraordinary life in science.

1:45

4pAB3. Temporary threshold shifts in Boris the bottlenose dolphin (*Tursiops truncatus*), varying noise duration and intensity. T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Marine Res. Facilty 227 (MS# 50), Woods Hole, MA 02543-1050, amooney@whoi.edu) and Paul E. Nachtigall (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kanehoe, HI)

There is much concern regarding increasing noise levels in the ocean and how it may affect marine mammals. Early on, there was little information regarding how sound affects marine mammals and no empirical data on the relationship between noise intensity and exposure duration. Working with Boris the bottlenose dolphin, *Tursiops truncatus*, we explored the effects of octave-band noise, and sonar signals on odontocete hearing and noise-induced temporary hearing threshold shifts (TTS). Sound pressure levels (SPLs) and exposure durations were varied to measure the effects of exposure duration and amplitude. Hearing thresholds were initially measured behaviorally, and later using auditory evoked potentials before and after sound exposure to track TTS and recovery. These efforts showed that threshold shifts were not linear but rather shorter sounds required greater sound exposure levels to induce TTS; a contrast to some early published literature. Further, sound exposure levels of sonar signals required to induce TTS were high, supporting the notion that relatively short sounds must be of relatively high intensity to induce threshold shifts. From these data and this remarkable animal, a novel algorithm was written that predicts the physiological effects of anthropogenic noise if the intensity and duration of exposure are known.

2:05

4pAB4. Forty-plus years of Navy research by the dolphin BLU. James Finneran (U.S. Navy Marine Mammal Porgram, Naval Information Warfare Ctr. Pacific, 53560 Hull St., San Diego, CA 92152, james.j.finneran.civ@us.navy.mil)

The dolphin "BLU" has had an amazing career spanning over 40 years of Navy fleet support and scientific research. Her research career began in the late 1970s with Sam Ridgway, studying lung-collapse and intramuscular nitrogen levels in diving dolphins. For the next several decades, she served with the Navy's Fleet mine-hunting systems and was deployed to locations all over the world. In 2003, at age 37, BLU returned to research, where her intelligence and unflappable demeaner quickly made her a star. Over the next few years she participated in a number of studies using behavioral and auditory evoked potential techniques to measure hearing thresholds and temporary threshold shift (TTS). TTS data from BLU were instrumental in developing marine mammal noise exposure regulatory guidelines. After having a calf in 2007, she became a working mom by beaching and transporting to the test pool each day, then returning to her calf at the conclusion of the session. BLU would go on to participate in a number of studies involving directional hearing, tactile sensitivity, and biosonar beam patterns. This talk will review BLU's career with the Navy and highlight her research contributions that have resulted in over 35 peer-reviewed journal articles.

2:25

4pAB5. Learning from Freja the harbor porpoise to appreciate movements as an important behavioral readout to understand echolocation-based target discrimination. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Michael Ladegaard (Biology, Aarhus Univ., Aarhus, Denmark), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), Peter T. Madsen, Kristian Beedholm (Biology, Aarhus Univ., Aarhus, Denmark), and Peter L. Tyack (Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, Fife, United Kingdom)

Toothed whales routinely discriminate and select prey via echolocation. Over a decade, the harbor porpoise Freja in Fjord and Bælt, Kerteminde, Denmark participated in a series of target discrimination experiments involving metal objects. Different from experiments in which the subjects were constrained to be stationary, Freja's free and voluntary movements offered insights into the tightly coupled nature of acoustic sampling and movements in echolocation-based target discrimination. Compared with observations from wild animals with unknown prey, the controlled experimental task and stimuli made it possible to interpret her behavior based on predictable target echo features. Here, we test the hypothesis that an echolocating animal plans its movement based on prior echo returns to gather better target discrimination information. We trained Freja to approach and select a sphere against a spheroid in a two-alternative forced-choice discrimination task, and show that her performance and movement trajectory varied depending on the aspect ratio and presentation angle of the spheroid. Consistent with our hypothesis, we further show that Freja's movement patterns were likely driven by echo information received earlier during the approach that is correlated with the discrimination difficulty. These results highlight movements as an important behavioral readout of the dynamic, closed-loop sensorimotor feedback in echolocation. [Work supported by ONR.]

2:45

4pAB6. Celebrating the Navy's sea canaries: The lives and bioacoustic research accomplishments of white whale (*Delphinapterus leucas*) collaborators. Carolyn E. Schlundt (Peraton / US Navy Marine Mammal Program, 4045 Hancock St., Ste. 210, San Diego, CA 92110, cmelka@peraton.com)

The Navy Marine Mammal Program has engaged many different species over the years to accomplish missions of finding and retrieving objects in the marine environment, and asset protection, but perhaps none has been so engaging as the white whale, or beluga (*Delphinapterus leucas*). At a time when the Cold War focus moved to the frigid waters of the Artic, the Navy needed echolocating, deep diving, cold-water experts. Enter, the belugas. Unique because of their all-white color, lack of a dorsal fin, and extensive repertoire of bird-like vocalizations (hence the moniker "sea canary"), the group with their naturally curious, gregarious demeanor and malleable melons, also quickly began to grace the pages of scientific journals and book chapters with their acoustic capabilities for a period of approximately 30 years. Most notably among them were a female named MUK (after the Inuit word muk tuk for whaleskin and blubber) and a male named NOC (after the tiny biting summer flies known as no-see-ums). Together, these two whales expanded our knowledge of marine mammal bioacoustics and physiology with their collaboration on projects related to detection, hearing, echolocation, nasal pressure, sound production, human-speech mimicry, net-aided foraging, diving physiology, stress hormones, reproduction, growth, and development.

3:05–3:20 Break

3:20

4pAB7. A legacy of excellence: SAY's contributions to bioacoustics research at the Navy Marine Mammal Program. Katie A. Christman (Psych., Univ. of California San Diego, 2240 Shelter Island Dr., San Diego, CA 92106, katie.christman@nmmf.org)

The dolphin SAY is considered a jack-of-all-trades in marine mammal science. This comes as no surprise as her mother was one of the five dolphins deployed to Cam Ranh Bay, Vietnam as part of the Navy's first deployable marine mammal system. SAY was born in 1979 and began participating in research from birth; she was the first dolphin to have her caloric intake and weight tracked as a calf into adulthood, and while carrying her own calf in 1991. In her early years, she participated in cutting-edge studies using high-speed video endoscopy to better understand dolphin click production. She later participated in echolocation vigilance tasks that spearheaded our understanding of dolphin cognitive abilities focusing on duration and the effects of anthropogenic noise. During this same time, she diligently contributed to projects that revolutionized our understanding of dolphin biosonar and its relation to perception and auditory system neurophysiology. SAY will be remembered by her charismatic personality and her 38 years of service to science and the United States Navy. Her daughter continues to carry out her legacy working for the Navy's fleet mine-hunting system.

3:40

4pAB8. A mammal for all seasons: Contributions from the bottlenose dolphin TRO to bioacoustics research at the US Navy Marine Mammal Program. Jason Mulsow (Biologic and Bioacoustic Res., National Marine Mammal Foundation, 2240 Shelter Island Dr. Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org)

Among the notoriously small sample sizes of marine mammals in cooperative behavioral research, there is an even smaller subset of subjects that can be considered "experts." In essence, these are subjects that have "learned how to learn" in the contexts of novel experimental paradigms. One such subject is the bottlenose dolphin TRO at the US Navy Marine Mammal Program. The first scientific publication including TRO heralded his potential; aided by a phantom echo generator, he was likely the first marine mammal in history to echolocate while out of water. Subsequent behavioral experiments with TRO have examined numerous aspects of biosonar, including transmit beam characteristics and the use of click "packets" during long-range target detection. The patient nature of TRO has made him an ideal subject for electrophysiological studies, where he has diligently listened to exotic acoustic stimuli designed to examine auditory processing in the dolphin brainstem. The most significant scientific contributions from TRO are arguably from studies that have used auditory evoked potential methods during biosonar tasks. These data provide the temporal resolution necessary to examine hearing at the level of click production and echo reception and are a result of TRO's dependable participation in studies combining multiple experimental techniques

4:00

4pAB9. "Nacho"—A willing and adventurous dolphin with diverse research accomplishments. Dorian Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, dorian.houser@nmmf.org)

The bottlenose dolphin, WEN (also known as "Nacho"), worked in the swimmer interdiction system of the United States Navy Marine Mammal Program (MMP) before being transferred to MMP research in the mid-2000s. During his time away from systems work, WEN participated in diverse research projects that benefited our understanding of dolphin hearing, bioacoustics, and auditory physiology. His contributions include several ground-breaking studies: (1) he demonstrated that dolphins could remain vigilant at an acoustic target detection task for days without rest or performance decrement, (2) participated in the first study matching dolphin structural (MRI) and functional imaging (SPECT, PET) with an emphasis on the anatomy of sound-reception and transmission, (3) performed open-ocean work to investigate the target strength of dolphins at depth and the potential for intravascular bubble formation during diving, (4) was instrumental in the development of auditory evoked potential (AEP) methods for use in MMP dolphins, and (5) performed the first direct comparison of AEP and behavioral thresholds to the same acoustic stimulus in a marine mammal. Not only have WEN's contributions significantly advanced marine mammal science, but with his return to systems work, he continues to contribute to the national security of the United States.

Contributed Papers

4:20

4pAB10. Ronan and the legacy of Schusterman's California sea lions. Ryan A. Jones (UC Santa Cruz, Long Marine Lab, 115, Santa Cruz, CA 95060, ryajones@ucsc.edu), Peter F. Cook (New College Florida, Sarasota, FL), and Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

California sea lion Ronan was named in honor of Dr. Ron Schusterman, a pioneer in psychoacoustic studies of marine mammals. His early work with sea lions Sam, Rocky, Rio and others established rigorous methods for sensory assessment including those based on signal detection theory and the psychological principles underlying auditory learning and discrimination. Ronan is the most recent sea lion to join the research program in Santa Cruz, CA, adding to a long legacy of sensory and cognitive research. For more than a decade, Ronan has contributed to studies that increase understanding of marine mammal bioacoustics. These include marine and terrestrial audiograms that are often cited as representatives for the species, studies of ultrasonic hearing ability, and listening tasks performed in complex masking scenarios. Ronan is best known for her work on rhythmic entrainment, demonstrating auditory-motor synchronization and challenging theories as to the neural origin of this behavior. Ronan's current psychoacoustics work is focused on understanding hearing at very low frequencies (<100 Hz) and the effects of parameters such as noise bandwidth and level on auditory masking.

4:35

4pAB11. The huaka'i of Hawaiian monk seal *Kekoa*: Conservation through sound science. Kirby Parnell (Hawaii Inst. of Marine Biology, Univ. of Hawaii Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, keparnel@hawaii.edu), Brandi Ruscher, Jillian M. Sills, and Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

The Hawaiian monk seal (*Neomonachus schauinslandi*) is an endangered marine mammal and the subject of significant conservation concern. Limited bioacoustic information was available for this species until recently. The adult male Hawaiian monk seal *Kekoa* (KE18) was removed from the wild after repeated problematic interactions with conspecifics; he was then transferred temporarily to UC Santa Cruz, where he participated in studies to increase understanding of monk seal auditory biology. Compared to other seals, *Kekoa's* behavioral hearing data suggest that monk seals have less sensitive hearing and a reduced functional frequency range of hearing in air and under water. A year-round characterization of his spontaneous underwater vocalizations revealed at least six low-frequency call types with a simultaneous peak in calling behavior and testosterone levels during the breeding season. *Kekoa's* huaka'i, or journey, has provided the first description of underwater communication for this protected species and contributed much-needed perspective about amphibious hearing abilities. *Kekoa's* work has also inspired ongoing research with captive and wild individuals to confirm species-level traits in sound reception and production. These efforts have applications to studies of free-ranging monk seals through passive acoustic monitoring, development of automated call detectors, and the use of multi-sensor biologging devices.

Session 4pBAa

Biomedical Acoustics, Engineering Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Advances in Elastography II

Thomas Royston, Cochair
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Matthew W. Urban, Cochair
Department of Radiology, Mayo Clinic, Department of Radiology, Mayo Clinic, Rochester, MN 55905

Invited Papers

1:30

4pBAa1. Magnetic resonance elastography of the brain: Recent advances. Curtis L. Johnson (Univ. of Delaware, 540 S College Ave., Newark, DE 19713, clj@udel.edu)

Magnetic resonance elastography (MRE) is an imaging technique for generating maps of quantitative tissue mechanical properties. Mechanical properties of the brain offer sensitive new biomarkers for diagnosing neurological conditions and individual differences in brain health that support cognitive functioning. However, the mechanical complexity of the brain—e.g., heterogeneity in space and scale, frequency- and direction-dependent behavior, skull encasement—require advanced MRE techniques to reliably and accurately map brain tissue properties. We will discuss recent developments in MRE imaging and inversion methods and experiments aimed at more completely estimating brain properties. This includes approaches for determining mechanical anisotropy in white matter tracts comprising aligned axonal fiber bundles from rich acoustic wavefields. And new experiments capturing time-dependent variation in properties and the relationships to cerebral blood flow. Such advances increase the capabilities of MRE as an imaging tool and concomitantly enrich our understanding of the physiological bases of *in vivo* brain mechanical properties.

1:50

4pBAa2. Brain MR elastography: Stiffness and beyond. Ziying Yin (Mayo Clinic, 200 First St. SW, Rochester, MN 55905, yin.ziying@mayo.edu)

Brain MR elastography (MRE) is becoming a powerful tool in the investigation of neurophysiological and neuropathological states. In brain MRE, quantitative measures of brain mechanical properties can be assessed noninvasively by analyzing the induced brain motion in response to extrinsic or intrinsic excitation, providing unique clues to the mechanical rigidity, viscosity, friction, and connectivity of brain tissues. Changes in brain elasticity and viscosity are associated with inflammation, demyelination, and neurodegeneration during the onset and progression of many brain diseases. Beyond shear stiffness and modulus, MRE contains rich information that can be derived to enhance the characterization of disease-related structural changes, especially in intracranial tumors and traumatic brain injury, which will be the focus here. MRE-based slip interface imaging (SII) utilizes shear strain mapping to qualitatively assess the degree of adhesiveness of intracranial tumors. Together with tumor consistency, MRE offers a unique opportunity for improving preoperative planning before tumor resection as well as predicting patient outcomes. Analysis of head dynamics induced by non-impact dynamic loading from MRE measures the mechanical dampening and isolation capability of the skull-brain interface and changes in those features in sub-concussive repetitive head impacts, which could provide imaging biomarkers to facilitate risk management for future brain injury.

2:10

4pBAa3. Simultaneous acquisition of vibration and diffusion by multifrequency diffusion tensor imaging-magnetic resonance elastography (mDTI-MRE). Dieter Klatt (Biomedical Eng., Univ. of Illinois at Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, dklatt@uic.edu), Shujun Lin, Richard Magin (Biomedical Eng., Univ. of Illinois at Chicago, Chicago, IL), Aaron T. Anderson, and Bradley P. Sutton (Beckman Inst. for Adv. Sci. and Technol., Urbana, IL)

Magnetic Resonance Elastography (MRE) and Diffusion Tensor Imaging (DTI) are MRI techniques that provide complementary diagnostic information and utilize motion-sensitive magnetic field gradients. In MRE, tissue vibrations are encoded into the MRI signal phase, while in DTI the diffusive motion of water molecules is assessed by means of the MRI signal magnitude. Careful consideration of the motion-encoding gradient timing conditions and the mechanical frequencies allows the simultaneous acquisition of vibration and diffusion data without observable mutual interferences. We propose a multifrequency Diffusion Tensor Imaging-Magnetic Resonance Elastography (mDTI-MRE) protocol that measures the diffusive behavior along 24 directions and the full 3D displacement field at two mechanical frequencies. Using this method, mDTI-MRE provides information about the mechanical wave field and about a preferred direction, which may facilitate the accurate mechanical characterization of anisotropic biological tissues, such as brain white matter and skeletal muscle.

2:30

4pBAa4. From dynamic to passive elastography, what technique used? Jean-Luc Gennisson (Université Paris Saclay - CNRS, BioMaps, 4 Pl. du général Leclerc, Orsay 91401, France, jean-luc.gennisson@universite-paris-saclay.FR)

Elastography is a method that allows to quantify mechanical properties of biological tissues. In medical imaging, this last technique has been developed mainly in ultrasound (US) and in magnetic resonance imaging (MRI). Both imaging techniques have advantages and drawbacks depending on the organs and pathologies investigated. In this presentation we focused on two organs, where each technique has its advantage to investigate mechanical properties: brain and muscle. In brain, the preferred technique is MRI since US propagation is very complicated through the skull bone. Taking advantage of 3D acquisition, we investigated brain tumors with a new elastography approach: passive elastography. From the natural vibration, such as cardiac beating or respiration, we locally recover the natural shear wavelength that propagates giving access a stiffness mapping of the brain. Results are compared with US acquire during neurosurgery, showing a good accordance between both techniques. In muscles, US are used to recover not only stiffness but also others mechanical parameters such as viscoelastic anisotropy or shear non-linearity. Special US sequences were developed to quantify these two parameters in different physiological conditions that change in real-time. This presentation shows that depending on the investigated organ, physicists and physicians need to adapt the imaging strategy to better characterize tissues.

2:50

4pBAa5. Quantification of liver fat and fibrosis using elastography plus composite theory. Kevin J. Parker (Elec. & Comput. Eng., Univ. of Rochester, Comput. Studies Bldg. 724, Box 270231, Rochester, NY 14627-0231, kevin.parker@rochester.edu), Sedigheh Poul (Mech. Eng., Univ. of Rochester, Rochester, NY), and Juvenal Ormachea (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY)

There are a growing set of ultrasound and viscoelastic measures that can be correlated with fat or fibrosis in the liver. We find that fat and fibrosis jointly influence important properties of the liver and can be considered as confounding cofactors within most simple measures. For example, shear wave attenuation is sensitive to the accumulation of viscous fat, but is also influenced by the degree of fibrosis, so attenuation by itself is insufficient for accurate estimation of liver fat. However, using some robust elastography techniques to assess both the shear wave phase velocity and the shear wave attenuation in a region of the liver, these measured values are found to be sufficient information for solving for both the unknown fat percent volume and the liver stiffness related to fibrosis score. A classical theory of composite viscoelastic materials is used to solve for the unknowns. Examples from human clinical studies are shown to correspond to biopsy proven grades of steatosis and fibrosis.

3:10

4pBAa6. Ultrasound shear wave attenuation in nonalcoholic fatty liver disease: performance of the revisited frequency-shift method with pre-clinical and human datasets. Ladan Yazdani, Iman Rafati (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montreal, QC, Canada), Damien Olivié, Jeanne-Marie Giard (Hepatology, Univ. of Montreal Hospital, Montreal, QC, Canada), Giada Sebastiani (Hepatology, McGill Univ., Montreal, QC, Canada), Bich N. Nguyen (Pathol., Univ. of Montreal Hospital, Montreal, QC, Canada), An Tang (Radiology, Univ. of Montreal Hospital, Montreal, QC, Canada), and Guy Cloutier (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada, guy.cloutier@umontreal.ca)

Assessment of ultrasound shear wave attenuation (SWA) can increase the accuracy of liver steatosis grading. Recent developments allowed to propose the revisited frequency shift method for SWA imaging. This method was tested on force-fed ducks for *foie gras* production ($n = 6$), and a feasibility study on patients with non-alcoholic steatohepatitis (NASH $n = 27 + 13$ healthy volunteers) was conducted. Liver biopsy and magnetic resonance imaging proton density fat fraction (MRI-PDFF) were available as reference standards. For the human study, shear wave dispersion (SWD) was also computed. A subset of participants had repeated measurements (<1-month) to assess repeatability. The mean SWA (coefficient of variation within the attenuation map) for healthy duck livers were 0.77 (0.66), 1.18 (0.22), and 1.52 Np/m/Hz (0.14), and for fatty duck livers they were higher at 3.13 (0.55), 3.16 (0.24), and 4.84 Np/m/Hz (0.23). Receiver operating characteristic (ROC) curves using biopsy as reference were compared for steatosis grades >0 (>S0), >S1, and >S2. SWA was as good as MRI-PDFF, and both were better than SWD for steatosis grading ($p < 0.05$). The repeatability of SWA was very good (mean intraclass correlation coefficient of 0.97). In summary, SWA seems promising to become an ultrasound reference standard for steatosis grading in NASH.

3:30–3:45 Break

Contributed Papers

3:45

4pBAa7. Limitations of the Caputo time-fractional wave equation when applied to shear waves in pig liver. Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, Michigan State University, East Lansing, MI 48824, mcgough@egr.msu.edu) and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

When applied to compressional waves, the Caputo time-fractional wave equation describes power law attenuation in soft tissue. However, when applied to shear waves, the Caputo wave equation produces some unexpected behavior. To demonstrate examples of the time-domain waveforms generated by this fractional calculus model, the Caputo wave equation is numerically evaluated for multiple orders of the time-fractional derivative between 0 and 1, for an extended range of values for the relaxation time, and for a fixed value of the shear wave speed extracted from *ex vivo* pig liver. The computed waveforms are then compared to the measured shear wave particle velocities obtained from *ex vivo* pig liver. These comparisons reveal that the full-width at half maximum (FWHM) of the positive component of the main shear wave particle velocity waveform measured in pig liver is at least an order of magnitude greater than the FWHM obtained from the Caputo wave equation for all parameter combinations evaluated. These preliminary results indicate that the Caputo wave equation produces time-domain waveforms that are inconsistent with measured shear wave data in pig liver and that further efforts are required to establish more effective fractional calculus models for shear waves in soft tissue.

4:00

4pBAa8. Parametric evaluation of Kelvin-Voigt dispersion curve fitting.
Luiz Vasconcelos (Dept. of Radiology, Mayo Clinic, 202 4th St. SW, Rochester, MN 55902, vasconcelos.luiz@mayo.edu), Piotr Kijanka (Dept. of Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Ultrasound shear wave elastography (SWE) is a useful technique for non-invasive tissue assessment. For effective quantitative analysis the employment of physical models is necessary for rheological parameter estimation, such as the Kelvin-Voigt (KV) model. From SWE data, the frequency response can be calculated, and the shear wave velocity dispersion can be fit, yielding the shear elasticity and shear viscosity, μ_1 and μ_2, respectively. In this study, a parametric evaluation is performed using staggered-grid finite-difference simulations of materials with $\mu_1 = 1$–25 kPa (increment: 1 kPa) and $\mu_2 = 0$–10 Pa s (increment: 0.5 Pa s), with an acoustic radiation force push of f-number equal to 2. The novel Stockwell transform combined with a slant frequency-wavenumber analysis (GST-SFK) was compared with the two-dimensional Fourier transform for dispersion curve estimation over a variety of frequency ranges. Regardless of dispersion curve estimation technique, the accuracy of estimating μ_1 is confounded for μ_2 values above 5 Pa s. It was found that KV fitting benefits from wider frequency ranges (up to 1 kHz), with lower μ_1 and μ_2 estimation errors. The GST-SFK was the best dispersion curve calculation technique, yet the KV fitting process was found to be unreliable for parameter estimation in highly viscous media.

4:15

4pBAa9. Nonlinear least-squares estimation of shear modulus and shear viscosity in viscoelastic media.
Nicholas A. Bannon (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, bannonni@msu.edu), Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Time-domain estimates of the shear wave speed are obtained by performing cross-correlations or by evaluating the time-to-peak for two laterally separated shear wave particle velocity waveforms, where the distance divided by the propagation time yields an approximate value for the shear wave speed. Challenges associated with these time-domain estimation approaches include increasing errors as the shear viscosity increases and the absence of an estimated value for the shear viscosity parameter which is responsible for the rapid attenuation of shear waves in soft tissue. To obtain a time-domain estimate of the shear viscosity that also provides an estimate for the shear wave speed, a nonlinear least squares estimation approach is applied to three-dimensional (3D) shear wave particle velocities computed for a shear elasticity of 1.5 kPa, shear viscosities of 1, 2, 3, and 4 Pa.s, and a realistic simulated 3D acoustic radiation force excitation. The results show cross-correlations tend to over-estimate the value of the shear wave speed, the nonlinear least squares approach tends to under-estimate the value of the shear wave speed, and the nonlinear least square approach produces more accurate estimates as the shear viscosity increases. Two-dimensional maps of each estimated parameter are also shown.

4:30

4pBAa10. Homogeneity versus inhomogeneity in muscle elastography.
Lara Nammari (Biomedical Eng., Univ. of Illinois at Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, lnamma2@uic.edu), Dieter Klatt, and Thomas Royston (Biomedical Eng., Univ. of Illinois at Chicago, Chicago, IL)

The hierarchal microstructure and arrangement of skeletal muscle fibers results in both anisotropy and inhomogeneity. Most elastography reconstruction methods assume homogeneity and many also assume isotropy. In this computational (finite element) and experimental (MR) elastography study we investigate the impact of assuming homogeneity as muscle architecture dimensions and property differences are varied. This analysis also takes into consideration the inherent anisotropy (transverse isotropy) created by homogenization of the fiber structure, and the added complication of nonhomogeneous (pre-stress) boundary conditions. Specifically, the computational harmonic analysis uses a parametric sweep across 200 Hz to 2000 Hz applied to 4-, 40- and 400-fiber models with the aligned fiber cross-sectional area fixed (reduced fiber diameter as fiber count increases) and comprising 50% of the circular cross-sectional area in the fixed diameter cylindrical phantom. Fiber elastic moduli two to five times that of connective tissue are considered. This phantom also undergoes tensile axial pre-loading simultaneous to elastography studies. Predicted results are compared to a homogeneous transverse isotropic model that is reached in the limit as the fiber count increases to infinity while still summing to 50% of the cross-sectional area. Selected cases, such as the 4-fiber model are also evaluated experimentally using MR elastography.

4:45

4pBAa11. Uniaxial prestress and waveguide effects on estimates of the complex shear modulus using magnetic resonance elastography in a transverse isotropic muscle phantom and excised muscle.
Melika Salehabadi (Biomedical Eng., Univ. of Illinois at Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, msaleh25@uic.edu), Joseph Crutison, Lara Nammari, Dieter Klatt, and Thomas Royston (Biomedical Eng., Univ. of Illinois at Chicago, Chicago, IL)

Tensile prestress is inherent to the functional role of some biological tissues currently being studied using elastography, such as skeletal and cardiac muscle, arterial walls, and the cornea. Therefore, the impact of prestress coupled with waveguide effects due to small dimensions in one or more directions needs to be better understood. An experimental configuration is designed, fabricated and experimentally tested using magnetic resonance elastography (MRE). Cylindrically-shaped isotropic and transversely isotropic phantoms, as well as the excised cat soleus muscle are statically stretched along their main axis to specific prestrain levels while simultaneously conducting MRE measurements that enable synchronous motion-encoding in all three dimensions (polarization directions). In the case of the excised cat soleus, diffusion tensor imaging is also performed to enable a co-registered mapping of fiber structure. Guided by analytical models and numerical finite element simulations, experimental measurements are post-processed to obtain an estimate of the complex (viscoelastic) shear modulus as a function of prestress level and frequency of vibratory motion.

5:00

4pBAa12. Elastography using torsional wave motion in transverse isotropic material.
Aime Luna (Biomedical Eng., Univ. of Illinois at Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, aluna25@uic.edu), Melika Salehabadi, Dieter Klatt, and Thomas Royston (Biomedical Eng., Univ. of Illinois at Chicago, Chicago, IL)

An experimental configuration for driving and measuring torsional wave motion is designed, fabricated and experimentally tested using magnetic resonance elastography (MRE) in both a ultra-high field preclinical MRI system and in a cryogen-free low-field tabletop MRI system. Cylindrically-shaped isotropic and transversely isotropic phantoms, as well as the excised cat soleus muscle are statically stretched along their main axis to specific prestrain levels while simultaneously conducting MRE measurements with torsional waves. In the case of the excised cat soleus, diffusion tensor imaging is also performed to enable a co-registered mapping of fiber structure. The potential advantage of torsional wave excitation to drive shear wave motion over a large volume with uniform amplitude, and hence SNR, is tested by comparison to excitation of shear waves using other methods.

5:15

4pBAa13. Ultrasound elastography evaluation of age-related muscle atrophy: An *ex vivo* study.
Bushira Musa (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, bushira.musa@udc.edu), Justin An, Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, the feasibility of ultrasound elastography to evaluate age-related muscle atrophy is investigated. Ultrasound elastography is used to assess muscle stiffness alterations of bovine shoulder samples during aging treatment. Of particular interest is to investigate the decrease in muscle stiffness due to two potential reasons: (1) decrease in collagen fibers; and (2) mechanical breakdown of the muscle fascia (the soft tissue component of connective tissue surrounding muscle). Two methods of aging treatments are employed: (i) chemical treatment, and (ii) percussion treatment. Areas of the bovine shoulder samples studied, were the triceps brachii and infraspinatus. The chemically treated bovine samples demonstrated significant alterations in muscle stiffness under ultrasound in comparison to percussive treatment. The significance of these results will be discussed.

Session 4pBAb

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Making and Using Cavitation Images for Therapeutic Ultrasound II

Meaghan O'Reilly, Cochair
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Michael Gray, Cochair
Institute of Biomedical Engineering, University of Oxford, Institute of Biomedical Engineering, University of Oxford, Oxford, OX3 7DQ, United Kingdom

Kevin J. Haworth, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

Invited Papers

1:30

4pBAb1. Passive acoustic mapping within the cranial vault during microbubble-mediated ultrasound brain therapy. Ryan M. Jones (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Dallan McMahon, Lulu Deng, Meaghan O'Reilly, and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

Over the past 15 years, the use of passive sensor arrays combined with established beamforming algorithms to image acoustic activity during ultrasound therapies, so-called passive acoustic mapping (PAM), has been increasingly investigated for cavitation-based treatment monitoring and control purposes. Our group is interested in applications of microbubble-mediated ultrasound therapy in the brain, during which the skull bone presents unique challenges for both treatment delivery and acoustic emissions monitoring. We have demonstrated that skull-specific transcranial aberration correction methods can be applied during receive beamforming to augment PAM image quality through the skull bone, borrowing techniques developed originally for transmit beam focusing. Using custom clinical-prototype transmit/receive phased arrays, we have performed 3D microbubble imaging *in vivo* through *ex vivo* human skullcaps, and have exploited the resulting spatiotemporal cavitation information for real-time exposure level calibration and offline bioeffect distribution prediction. Ultrafast processing of acoustic emissions data can uncover cavitation dynamics hidden by conventional whole-burst temporal averaging, as well as inform temporal under-sampling strategies when millisecond-long tone bursts are applied. This talk will provide a historical overview of PAM for ultrasound therapy monitoring throughout the body, followed by a summary of recent progress made in mapping cavitation activity within the cranial vault during brain applications.

1:50

4pBAb2. Real-time transcranial mapping in non-human primates and human subjects during opening of the blood-brain barrier. Elisa Konofagou (Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, ek2191@columbia.edu)

Opening of the blood-brain barrier (BBB) with focused ultrasound and microbubbles has been shown repeatedly in several preclinical disease models and clinical studies. Transcranial 2D mapping of the cavitation mechanism responsible for BBB opening is however challenging in primate brains due to the skull thickness. A real-time cavitation mapping system will be presented that can transcranially map the cavitation occurrence and dose in real time throughout the BBB opening procedure in Alzheimer's disease (AD) adult and diffuse intrinsic pontine glioma (DIPG) pediatric patients. Passive acoustic mapping (PAM) with coherence factor (CF) correction is used to passively map the microbubble activity within the brain. Compared to passive cavitation detection (PCD), multi-element CF-PAM allows us to determine the exact location of the BBB opening without the need of contrast-enhanced MRI. An open-architecture ultrasound imager (Vantage, Verasonics, Redmond, WA) with a P4-2 (ATL, Philips) is used to provide the best tradeoff between transcranial propagation, spatial resolution and depth penetration. The system is first optimized in targeting the putamen during BBB opening in non-human primates followed by feasibility in the prefrontal cortex in AD and the pons in DIPG patients, achieving feedback rates of 2 Hz during the clinical procedures.

Contributed Papers

2:10

4pBAb3. Transcranial cavitation mapping of blood–brain barrier opening regions in Alzheimer's disease patients using a neuronavigation-guided focused ultrasound system. Sua Bae (Dept. of Biomedical Eng., Columbia Univ., 630 W 168th St., New York, NY 10032, sb4495@columbia.edu), Antonios Pouliopoulos, Robin Ji, Keyu Liu, Sergio Jiménez-Gambín, Omid Yousefian (Dept. of Biomedical Eng., Columbia Univ., New York, NY), Danae Kokossis (Dept. of Radiation Oncology, Columbia Univ., New York, NY), Lawrence Honig (Dept. of Neurology, Columbia Univ., New York, NY), and Elisa Konofagou (Dept. of Biomedical Eng., Columbia Univ., New York, NY)

We present real-time cavitation monitoring and mapping in a clinical trial for blood-brain barrier (BBB) opening in Alzheimer's disease (AD) patients using a neuronavigation-guided focused ultrasound (FUS) system. Six AD patients ($N = 6$, age $= 68.5 \pm 9.5$) were sonicated at the right prefrontal lobe with a single-element FUS transducer ($PNP = 0.2$ MPa, $fc = 0.25$ MHz, pulse length $= 10$ ms, $PRF = 2$ Hz, duration $= 2$ min) with the microbubble administration (Definity). We performed passive cavitation detection (PCD) using a single-element hydrophone ($N = 4$), and passive acoustic mapping (PAM) with a 64-element phased array transducer ($N = 2$). Five patients showed successful BBB opening (volume $= 654 \pm 394\ mm^3$) on day0, and closure on day3, confirmed by contrast-enhanced T1-weighted MRI, while one patient served as a negative BBB opening case due to the insufficient microbubble administration. Ultra-harmonic cavitation dose (CD) correlated with the opening volume ($R^2 = 0.66$, $N = 4$), but harmonic and broadband CDs showed a relatively poor correlation ($R^2 = 0.06$ and 0.33). Cavitation maps ($N = 2$) showed a spatial distribution of acoustic energy that roughly coincided with the respective BBB opening location. We thus demonstrated the successful BBB opening and cavitation monitoring with the neuronavigation-guided system that allowed for a cost-effective procedure compared to the MR-guided approaches. The PCD and PAM showed promising results for potentially predicting the BBB opening volume and location found in MRI.

2:25

4pBAb4. Theoretical and experimental investigation of the vaporisation of superheated perfluorocarbon droplets. Luca Bau, Qiang Wu (Univ. of Oxford, Oxford, United Kingdom), Nicholas Ovenden (Mathematics, Univ. College London, London, United Kingdom), and Eleanor P. Stride (Univ. of Oxford, Inst. of Biomedical Eng., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

Acoustically activatable "nano"droplets consisting of superheated liquid perfluorocarbons have been investigated for both imaging and therapeutic applications. They offer longer circulation half-lives, higher surface area to volume ratio and the ability to perfuse the microvasculature more easily than gas-filled microbubbles. Optimising droplet formulations to avoid spontaneous vaporisation whilst keeping the peak negative pressure required for acoustic activation within the clinically safe range has proved challenging. The aim of this study was to derive a theoretical model to predict the probablity of droplet vaporisation under different conditions, the resulting bubble dynamics and radiated pressure; and compare the results with observations made by optical microscopy including high speed video. It was found that spontaneous vaporisation of perfluoropropane droplets occurred readily in serum at 37 °C in the absence of ultrasound excitation. Higher molecular weight perflurocarbons required acoustic activation and experimentally measured rates were higher than those predicted by modelling of homogeneous nucelation alone. The results suggest that droplet aggregation and heterogeneous nucleation play important roles in droplet vaporisation and should be accounted for in selecting appropriate ultrasound exposure conditions. The prevalence of spontaneous vaporisation may also have important safety implications for clinical applications of low boiling point perfluorocarbon droplets.

2:40–3:10 Break

Invited Papers

3:10

4pBAb5. Real-time passive cavitation mapping with high spatial-temporal resolution. Junjie Yao (Duke Univ., 100 Sci. Dr., Hudson Hall Annex 261, Durham, NC 27708, junjie.yao@duke.edu)

Shock wave lithotripsy (SWL) and laser lithotripsy (LL) have been widely used for clinical treatment of kidney stones. Cavitation plays an important role in stone fragmentation in both SWL and LL, yet it may also contribute to renal tissue injury. It is therefore crucial to determine the spatiotemporal distributions of cavitation activities to maximize stone fragmentation while minimizing tissue injury. Passive cavitation mapping (PCM) has most practical applications in deep biological tissues and is most promising for clinical translation. We have developed a set of technologies for 2D/3D PCM that can be seamlessly integrated with ultrasound imaging and photoacoustic imaging. Our 2D/3D PCM has achieved a spatial resolution of hundreds of micrometers and a temporal resolution of several microseconds. We also developed a transient angular spectrum approach for PCM reconstruction, which is ten times faster than the traditional delay-and-sum method. Using the 2D/3D PCM system, we imaged shockwave- and laser-induced single cavitation bubbles in both free field and constricted space, as well as on large animal models. Collectively, our results have demonstrated the high reliability and spatial-temporal accuracy of the 2D/3D PCM approach, which paves the way for future *in vivo* applications and human studies during SWL and LL.

3:30

4pBAb6. Acoustic cavitation localization during histotripsy using transmit-receive capable arrays. 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), Mahmoud Komaiha, Scott Haskell, Ning Lu, Greyson Stocker, and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

The nucleation and collapse of cavitation generated during histotripsy results in point-like emissions of shockwaves. Owing to the short duration acoustic pulses utilized during histotripsy the acoustic background from pulse reflections off intervening tissues is minimal and temporally isolated. This allows acoustic cavitation emission (ACE) shockwaves to be differentiated from the background in measured signals using simplified approaches such as peak detection and time gating, even in signals acquired using narrow bandwidth receivers such as the transmitting elements of the histotripsy array. The array elements allow acquisitions from across the array's entire aperture, which in turn allows cavitation to be localized in 3D. Here we describe multiple methods for localizing cavitation and evaluating induced tissue damage based on ACE signals measured using transmit-receive capable histotripsy arrays. The speed and accuracy of the methods were evaluated in *ex vivo* and *in vitro* experiments. Cavitation could be localized in 3D at rates >100Hz. Mean localization errors, with respect to measurements from optical images, were <2 mm (approximately the diameter of the generated bubbles). Cavitation lifespans could be assessed concurrently with the bubbles' 3D locations and correlated well with histological observations of induced damage, allowing 3D assessments of damage.

Contributed Papers

3:50

4pBAb7. Histotripsy of healthy and tendinopathic *ex vivo* bovine tendons. Molly Smallcomb (Graduate Program in Acoust., Penn State Univ., University Park, PA) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu)

Histotripsy has successfully fractionated most soft tissues; however, highly collagenous tissues like tendon have been resistant to histotripsy fractionation. Previously, we showed that some histotripsy parameters could create mild mechanical microdamage in healthy *ex vivo* rat tendons. Our objective here is to evaluate whether complete histotripsy fractionation is possible in tendons. Eight bovine tendons were injected with collagenase to induce tendinopathy; an additional four tendons were unaltered. Tendons were exposed to single- or dual-frequency histotripsy at 1.07-, 1.5-, and/or 3.68-MHz with 10-ms pulses delivered at 1 Hz for 60 s. Treatments were monitored with passive cavitation detection (PCD) and samples were evaluated for damage. Results show that exposure of tendinopathic tendons to 1.5- and 3.68-MHz dual-frequency histotripsy produces a distinct, fractionated hole; 1.5- and 3.68-MHz single-frequency histotripsy produces partial fractionation without a distinct hole. No fractionation was observed in healthy tendons, or in tendinopathic tendons exposed to 1.07 MHz single- or dual-frequency histotripsy. Histologically, all tendons showed evidence of thermal necrosis independent of whether a hole was observed. PCD shows increases in amplitude and sustainment of cavitation in exposures that successfully fractionated tendon. These results suggest histotripsy fractionation is possible in tendinopathic tendons. [Work supported by NIH R21EB027886 and R01EB032860.]

4:05

4pBAb8. Monitoring cavitation dynamics evolution in tissue mimicking hydrogels for repeated exposures via acoustic cavitation emissions. Scott Haskell (Biomedical Eng., Univ. of Michigan, 1435 Wisteria Dr., Ann Arbor, MI 48104, haskells@umich.edu), Jonathan R. Sukovich, and Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Acoustic cavitation emission (ACE)/shockwave signals generated by cavitation during histotripsy have been demonstrated to undergo characteristic changes correlated with the level of damage generated in select target media following repeated cavitation exposures, and may allow assessment of therapy-induced tissue damage during histotripsy. Here we investigate the impacts of nucleation medium on the evolutions of cavitation dynamics and ACE signals during repeated cavitation exposures. Repeated cavitation is generated in agarose, gelatin, and polyacrylamide hydrogels using a 700-kHz histotripsy array. High-speed imaging, broadband hydrophones, and histotripsy array elements are used to measure ACE signals and bubble dynamics. Bubble lifespans and collapse shockwave amplitudes were observed to be equivalent in optical and acoustic measurements. However, the evolutions of these properties, as well as bubble maximum radii, varied significantly across materials. Maximum radii initially increased and then decreased in agarose, but remained constant in gelatin and polyacrylamide. Lifespans increased monotonically in agarose and gelatin, but decreased in polyacrylamide. Collapse shockwave amplitudes evolved differently in all materials. The observed evolutions of the maximum radii, lifespans, and collapse shockwave amplitudes between media suggest that these features are dependent on the structure and stiffness of the nucleation medium and must be further investigated in tissues.

4:20

4pBAb9. Contribution of bubble activity to the efficacy of histotripsy and catheter-directed recombinant tissue plasminogen activator for treatment of porcine thrombi *in vitro*. Shumeng Yang (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45229, yang2s5@mail.uc.edu), Chadi Zemzemi, Christy K. Holland, and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Histotripsy has been shown to be a promising adjuvant treatment for deep vein thrombosis (DVT), using the mechanical action of bubble clouds to enhance thrombolytic activity and reduce the risk of complications. The objective of this study was to compare the effects of histotripsy with and without recombinant tissue-plasminogen activator, rt-PA in a physiologic, *in vitro* model mimicking porcine DVT. Highly retracted porcine whole blood clots were treated for 1 h with catheter-directed infusions of either saline or rt-PA [0.04 mg/ml] at 25 ml/h, either with or without histotripsy exposure. Five-cycle, 1.5 MHz histotripsy pulses with a peak negative pressure of 32 MPa and pulse repetition frequency of 40 Hz were applied along the catheter at 0.3 mm/s and repeated 1 mm on either side. This application pattern was repeated 6 times, totaling approximately 120 to 144 K pulses per clot. B-mode and passive cavitation images were acquired every 10 pulses. The combination of rt-PA and histotripsy was more efficacious for recanalization than saline with histotripsy, saline, or rt-PA alone. Histotripsy exposure, but not rt-PA, improved mass loss ($p < 0.01$). These *in vitro* data support our hypothesis that the mechanical action of histotripsy-induced bubble clouds enhances catheter-directed thrombolytic.

Session 4pCA

Computational Acoustics, Structural Acoustics and Vibration, Musical Acoustics, Signal Processing in Acoustics, Underwater Acoustics, Engineering Acoustics, Biomedical Acoustics, and Physical Acoustics: Showcases of HPC Across Acoustics II

Kuangcheng Wu, Cochair
Naval Surface Warfare Center - Carderock Division, 9500 MacArthur Blvd, West Bethesda, MD 20817

Ralph T. Muehleisen, Cochair
Energy Systems, Argonne National Laboratory, 9700 S. Cass Ave, Bldg 362, Lemont, IL 60439-4801

Invited Papers

1:30

4pCA1. Numerical modeling of knocking combustion in spark ignition engines. Muhsin Ameen (Argonne National Lab., 9700 S Cass Ave., Lemont, IL 60439, mameen@anl.gov) and Pinaki Pal (Argonne National Lab., Lemont, IL)

At present, around 85% of the 1.1 billion passenger cars in the world are powered by gasoline spark ignition (SI) engines. Engine downsizing coupled with boosted operation and transition to sustainable fuels is considered an attractive strategy to enhance power density and reduce fuel consumption of SI engines. However, these operating conditions result in severe thermodynamic conditions, thereby promoting the likelihood of abnormal combustion phenomena such as knock. The severity of knock can vary significantly, and the efficiency of engines at high loads is limited in practice by heavy knocking phenomena. Since, a thorough analysis of such recurrent but non-cyclic phenomena via experiments alone becomes highly cumbersome, in the present work, a multi-cycle large-eddy simulation study was performed to quantitatively predict cyclic variability in the combustion process and cyclic knock intensity variability in a direct injection spark-ignition engine. For both the mild knock and heavy knock conditions, the numerical results were validated against experimental measurements. Based on the simulation results, a correlation analysis was performed considering combustion phasing, peak cylinder pressure and maximum amplitude of pressure oscillation. Furthermore, a detailed three-dimensional spatial analysis illustrated the evolution of auto-ignition kernel development and propagation of pressure waves during knocking combustion.

1:50

4pCA2. Modeling the performance of a family of phononic pseudo-crystal interposers. S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov), Chandler Smith, Rick A. Kellogg, and Ihab F. El-Kady (Sandia National Labs., Albuquerque, NM)

Recent technical efforts at Sandia National Laboratories have uncovered a need for high-frequency isolation in solid media across a range of ultrasonic frequencies; these needs include the requirement to suppress crosstalk between piezo-mechanical data and power channels situated on the same substrate. Broadband isolation larger than is possible with comparable phononic crystal designs is provided using a recently-developed phononic pseudo-crystal isolator concept. Insertion loss results from 2-D finite element simulations in COMSOL and 3-D simulations in Sierra-SD, Sandia's HPC compatible structural dynamics simulation platform, are used to characterize the performance of characteristic examples of this family of phononic isolators. Sierra-SD uses massively parallel computing and high-order polynomial finite elements (P-elements) to mitigate the computational cost required to simulate coupled electrical-mechanical 3-D ultrasonic wave propagation. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

2:10

4pCA3. Evaluating numerical techniques with HPC systems in predicting vibratory responses of large finite element models. Alyssa Bennett (Naval Surface Warfare Ctr. - Carderock Div., 9500 MacArthur Blvd, Bethesda, MD 20817, alyssa.m.bennett7.civ@us.navy.mil), Hector Flores, Kuangcheng Wu (Naval Surface Warfare Ctr. - Carderock Div., West Bethesda, MD), Steve Wong (AFRL/RCM, Wright-Patterson AFB, OH), and Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD)

Finite element (FE) analyses have been widely used to support design optimizations in various disciplines to improve performance and possibly save material, cost, and schedule. As computational resources (i.e., CPU, memory, storage, and accelerator) are getting more powerful and cost effective, the complexity and model sizes of numerical models are greatly increasing to capture geometry details and features of the design. Correspondingly, the computational time for complex and large FE models are significantly growing. Traditional brute-force approach is not quickly enough to support design iterations. In this presentation, various numerical techniques are investigated and exercised in the DoD High Performance Computing systems to identify their advantages and limitations in conducting large FE analyses. Those techniques include utilization of GPGPU, direct versus iterative solvers, approximation of frequency sweep, etc. As a benchmark case, frequency response functions of a flat plate with viscoelastic material is exercised at the DoD HPC systems to examine the numerical techniques' impacts.

2:25–2:40 Break

2:40

4pCA4. Low-frequency acoustic rotorcraft classification. Gordon M. Ochi (U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil) and Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Rotorcraft emit unique acoustic signatures tied primarily to their blade pass frequencies. In this study, we examine the ability of artificial neural network machine learning methods to successfully classify several distinct types of rotorcraft based on their low-frequency acoustic signatures. Time series, power spectral density, and Welch power spectral density input types are examined, corresponding to convolutional neural networks for the first input type, and multi-layer perceptrons for the latter two input types. A K-fold cross-validation is used to initially assess the generalization performance of the various models, along with confusion matrices for a deeper dive into the model performance. Ablation studies are discussed for insights into model behavior. These insights are used to showcase further enhanced neural network models with increased robustness and accuracy.

2:55

4pCA5. Parallelized optimization for substructured finite element structural acoustic problems. 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 parallelized optimization techniques for structural acoustics problems. In particular, the underlying problems studied here utilize finite element substructuring in order to solve the objective function, though the substructuring aspect is not the primary focus of this work. Instead, several different optimization strategies were employed on a selection of problems to better understand the tradeoffs between the different algorithms. Throughout this effort, the Python library PyGMO is used due to its capabilities to wrap around many different optimization algorithms and distribute optimization tasks over multiple CPUs. The candidate algorithms studied in this work include both heuristic global optimizers and gradient based local optimizers such as the following: Extended Ant Colony (GACO), Conjugate Gradient (CG), Sequential Quadratic Programing (SLSQP), and Method of Moving Asymptotes (MMA). Each method is evaluated individually as well as in combination with some or all of the others to evaluate if a mixed algorithm approach is advantageous. A range of computational resources are employed to understand the scalability and other aspects of the parallelized calculations.

3:10

4pCA6. Parallel implementations of the fast nearfield method for continuous-wave and transient pressure calculations. Jacob S. Honer (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI, honerja1@msu.edu) and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

FOCUS, the "Fast Object-oriented C++ Simulator" (https://www.egr.msu.edu/~fultras-web/), enables rapid calculations of continuous-wave and transient pressure fields generated by single transducers and phased arrays. FOCUS achieves small errors in relatively short computation times through memory-efficient calculations with the fast nearfield method, which converges exponentially. The fast nearfield method is also an embarrassingly parallel algorithm that supports further reductions in the computation time through hardware acceleration. Previous efforts to accelerate calculations with the fast nearfield method successfully implemented multithreading within multiple central processing unit (CPU) cores using OpenMP, and this capability is presently included with FOCUS. Prior success with OpenMP motivates ongoing efforts to further extend the functionality of FOCUS by interfacing the fast nearfield method to an NVIDIA graphics processing unit (GPU). To facilitate FOCUS calculations on a GPU, calculations with the fast nearfield method are implemented in CUDA for time-harmonic and transient pressure computations with each supported transducer shape. This combination achieves a significant reduction in the computation time compared to single-threaded execution. When evaluated on an NVIDIA 1670 GPU, the computation time is also approximately halved relative to the multithreaded OpenMP implementation. These results encourage further efforts to parallelize existing and future FOCUS routines.

Session 4pEA

Engineering Acoustics, Signal Processing in Acoustics, Underwater Acoustics, Architectural Acoustics, Biomedical Acoustics, Noise, and Physical Acoustics: Acoustic Measurement in Extreme Environments

Robert D. White, Chair
Mechanical Engineering, Tufts University, 200 College Ave., Medford, MA 02155

Invited Papers

1:30

4pEA1. Measuring under-ice sounds in extremely remote and cold environments. Andrew Barnard (Acoust., Penn State, 201C Appl. Sci. Bldg., University Park, PA 16802, barnard@psu.edu), Miles Penhale (Keweenaw Res. Ctr., Michigan Technolog. Univ., Houghton, MI), and Steven Whitaker (Naval Undersea Warfare Ctr., Houghton, MI)

Nearshore, shallow-water, ice-covered environments are unique and interesting places to study acoustics. However, these places also tend to be extremely remote, very cold, and dangerous places to work. This team has worked on nearshore ice-covered lakes and seas in the Houghton, MI, area and in the Utqiagvik, AK, region of the Arctic. Similarities and differences of these areas will be discussed as well as safety considerations for on-ice testing. Finally, data acquisition, sensor selection, mounting, and housings will be discussed, with data repercussions shown.

1:50

4pEA2. Capturing the sounds of the stratosphere using solar hot air balloons. Daniel Bowman (Sandia National Labs., Albuquerque, NM, dbowma@sandia.gov), Sarah Albert (Sandia National Labs., Albuquerque, NM), Siddharth Krishnamoorthy (Jet Propulsion Lab. California Inst. of Technol., Pasadena, CA), Emalee Hough, Zach Yap (Oklahoma State Univ., Stillwater, OK), Elizabeth Silber (Sandia National Labs., Albuquerque, NM), and Jonathan Lees (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Low frequency sounds can travel vast distances across the planet, carrying information about the events that generated them as well as the medium through which they travel. These sounds are usually recorded on surface-based sensors. Recently, however, sensors have been lofted on high altitude balloons, where they have revealed a rich soundscape quite different than that of the Earth's surface. Here we describe acoustic observations of the lower stratosphere conducted using inexpensive microbarometers lofted via passive solar hot air balloons. We discuss background noise as well as individual events, some of whose origins remain enigmatic. We delve into how to carry out an acoustic sensing mission using our instrumentation and flight system. Finally, we outline future directions for this technology, including deployment on extraterrestrial bodies such as Venus. [SNL is managed and operated by NTESS under DOE NNSA under Contract No. DE-NA000352.]

2:10

4pEA3. Decrease in volcano jet noise peak frequency as crater expands. Kathleen F. McKee (NASA Goddard Space Flight Ctr., 8800 Greenbelt Rd., Greenbelt, MD 20771, kfmckee@alaska.edu), Eveanjelene Snee (Cardiff Univ., Taupo, New Zealand), Sean Maher (U.S. Geological Survey, Vancouver, WA), Cassandra Smith (U.S. Geological Survey, Anchorage, AK), Kevin Reath (Cornell Univ., Ithaca, NY), Diana Roman (Carnegie Inst. for Sci., Washington, DC), Robin Matoza (Univ. of California Santa Barbara, Santa Barbara, CA), Simon Carn (Michigan Technolog. Univ., Houghton, MI), Larry Mastin (U.S. Geological Survey, Vancouver, WA), Kyle Anderson, David Damby (U.S. Geological Survey, Moffett Field, CA), Anna Perttu (Massey Univ., Palmerston North, New Zealand), Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrecht, the Netherlands), Rodrigo de Negri Leiva (Univ. of California Santa Barbara, Santa Barbara, CA), Artem Degterev, Alexander Rybin, Marina Chibisova (Sakhalin Volcanic Eruptions Response Team, Inst. of Marine Geology and Geophys., Yuzhno-Sakhalinsk, Russian Federation), Ima Itikarai, Kila Mulina, and Steve Saunders (Rabaul Volcano Observatory, Geological Survey of Papua New Guinea, Rabaul, Papua New Guinea)

Volcanic jet noise is the sound, often $<$20 Hz and termed infrasound, generated by momentum-driven fluid flow through a volcanic vent. Assuming the self-similarity of jet flows and audible jet noise extends to infrasonic volcanic jet noise, the Strouhal number, St, connects frequency changes to those in jet length (expanded jet diameter) and/or velocity scale (jet velocity). We examine the infrasound signals from the June 2019 eruptions of Raikoke, Kuril Islands and Ulawun, Papua New Guinea volcanoes with changes in crater geometry. We use data from the International Monitoring System infrasound network and pre- and post-eruption satellite data (PlanetScope imaging). We observe a decrease in infrasound peak frequency during the transition to a more intense phase, which remains through the end of both eruptions. With the PlanetScope data, we measured a crater area increase of $\sim$50 000 m^2 at Raikoke and $\sim$31 000 m^2 at Ulawun. We use crater diameter as a proxy for expanded jet diameter. Our analysis suggests that the increases in crater diameter alone cannot account for the decreases in peak frequency for a constant St. This suggests the jet velocity also increased, which fits satellite data, and or the fluid properties (e.g., particle loading, nozzle characteristics, etc.) changed. This is reasonable as the eruptions intensified, which likely involved an increase in jet velocity and erosion of the crater walls. This is the first study to corroborate the decrease in infrasound peak frequency with documented increase in crater area.

2:30

4pEA4. Measurement of electroacoustic transducer sensitivity in extreme hydrostatic pressure environments. David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., UMass Dartmouth, Fall River, MA 02723, dbAcousticsDB@gmail.com) and Eric K. Aikins (Elec. and Comput., Univ. of Massachusetts, New Bedford, MA)

The measurement and acoustic calibration of electroacoustic transducers for underwater sound under extreme pressures (to 10,000 psi / 7km depth) and variety range of temperatures is a great challenge. Specialized facilities consisting of large pressure vessels with pressure and temperature control do exist at Navy faciliites (www.navsea.navy.mil + USRD) but operate to moderate pressures, have limited availability, are expensive and limited access. An alternative or supplement is to measure the change in the electro-mechanical impedance of the of a transducer and use equivalent electrical cirucuit analysis to determine the resulting change in all electrical, electro-mechanical, and mechanical properties. Having the pressure and temperature depedence of all the consitutent parameters allows an accurate pediction of the transduer performance sensitivity and frequency response (such as TVR Transmit Pressure per Volt, and receive voltage per pressure sensitivity. We measured transducer performance of several devices in the laboratory pressure vessels up to 10 000 psi and report on findings.

Contributed Papers

2:50

4pEA5. AudioMoth adventures: How does a low-cost recording device perform in extreme environments? Nathan Wolek (Creative Arts, Stetson Univ., 421 N Woodland Blvd, Unit 8252, DeLand, FL 32723, nwolek@stetson.edu)

When it was originally released in 2018, the AudioMoth acoustic recording device delivered an attractive new platform for passive acoustic monitoring. With a price point under 100 US dollars, a size smaller than a deck of cards, and open-source software, the AudioMoth is an exciting option for getting started with bioacoustics monitoring. However, these same features often invite skepticism from those who have never used it. How good is the recording quality from its tiny MEMS microphone? How does it hold up to extreme outdoor conditions? The addition in 2022 of the HydroMoth for underwater recording, only invited more questions about its ability to perform in extreme environments underwater. This presentation will offer personal reports of the author's own experiences using the AudioMoth and HydroMoth. It will also include original field recordings captured in extreme environments like strong winds, heavy rains, and salt water lagoons. Special attention will be given to the two hurricanes that hit Florida in 2022, which were both captured by the author during a HydroMoth deployment. The audience will leave with a better idea of what to expect from the AudioMoth platform should they consider using it for their own projects.

3:05

4pEA6. Ultrasonic damage detection in lithium-ion cells with localized thermal abuse histories. Tyler McGee (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, tyler.m.mcgee@gmail.com), Barrett Neath (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Washington DC, DC), Samuel B. Matthews, Ofodike A. Ezekoye, and Michael R. Haberman (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Electric vehicles require nearly 1000 individual lithium-ion batteries to provide appropriate power and capacity. It has recently been shown that ultrasonic inspection can detect localized heating in a LIB cell with a combination of input frequencies and propagation paths [*J. Acoust. Soc. Am.* **152**, A283 (2022)]. However, monitoring the thermal conditions of every cell in a battery pack is highly challenging to implement. This work explores the use of ultrasonic inspection to diagnose LIB cells with damage histories due to local, thermal abuse. LIBs were interrogated with ultrasonic waves while subjected to electrical and thermal loading, specifically, standard charge-discharge cycling followed by moderate localized thermal abuse and another phase of charge-discharge cycling. Ultrasonic signals from each portion of the test and data for the cycling before and after heating are directly compared for indicators of past abuse. Experimental data are compared to a transfer-matrix model to simulate the time-of-flight (TOF) through an individual cell using temperature-specific material properties for individual components to simulate the effect of heat on TOF. Experimental results indicate that deviations in time-domain features of the received signals can be used to detect previous thermal abuse via ultrasonic testing during charge-discharge cycling after thermal abuse.

Session 4pID

Interdisciplinary and Student Council: Guidance From the Experts: Applying for Grants and Fellowships

Pratik Ambekar, Cochair
Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Brijonnay Madrigal, Cochair
Marine Mammal Research Program, University of Hawai'i at Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744

E. K. Ellington Scott, Cochair
Rensselaer Polytechnic Institute, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Zane T. Rusk, Cochair
The Pennsylvania State University, 104 Engineering Unit A, University Park, PA 16802

A panel of successful fellowship winners, selection committee members, and fellowship agency members will answer questions regarding grants and fellowships, application advice, and funding opportunities. The panelists will briefly introduce themselves, followed by a question and answer session with the audience.

THURSDAY AFTERNOON, 11 MAY 2023 DENVER/HOUSTON/KANSAS CITY, 1:00 P.M. TO 3:25 P.M.

Session 4pMUa

Musical Acoustics: Acoustics of Percussion Instruments

Andrew C. Morrison, Cochair
Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431

Colin Malloy, Cochair
Music, University of Victoria, 3330 Richmond Rd., Victoria, V8P 4P1, Canada

Chair's Introduction—1:00

Invited Papers

1:05

4pMUa1. Percussion instruments and their relationship to well-known structural radiators. Micah Shepherd (Brigham Young Univ., N249 ESC, Provo, UT 84602, mrs74@byu.edu)

Several structural radiators have been well studied in the past century and are commonly used to illustrate principles of structural-borne acoustic radiation. These structural radiators include the membrane, thin beam and thin plate. The principles of vibration and acoustic radiation of these radiators will be overviewed in terms of their normal modes. The standard assumptions which allow these structures to be useful in illustrating general principles of structural acoustics will also be outlined. The relationship of common percussion instruments to these structural radiators will then be presented with emphasis on which of the previously discussed assumptions hold for each instrument and which assumptions do not. Modern methods for studying the vibration and acoustic radiation of these instruments will then be outlined.

1:25

4pMUa2. Steelpan building and tuning: General basics and personal observations. Noah M. Sanderson (551 Judd Rd., Easton, CT 06612, noah.m.sanderson@gmail.com)

In this presentation, I will share the steps I use to build and tune steelpans, as well as some insights and observations from my experience as a builder/tuner. Building includes welding, sinking (stretching the steel into a concave surface), note shaping, and steel smoothing. After that, the steelpan is burned and then tuned. Tuning is done using hand hammering and occasional heating. It involves tuning multiple modes of vibration independently in a single note panel, adjusting the timbre, and getting the desired interactions between notes. Oftentimes, the tuning experience and final product is somewhat unpredictable. Small differences in the steel shape can result in large changes to the sound, and operations that will work on one note might not for another. Furthermore, there are many problems that arise from detrimental note-note interactions and note-skirt interactions. All this results in a seemingly mysterious nature to steelpans, requiring a degree of trial and error to get the desired qualities. My hope is that a dialogue between tuners and researchers can strengthen our understanding of the instrument from both viewpoints.

1:45

4pMUa3. Characterizing the vibrational behavior of steelpans with a reduced number of notes. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

The skilled artisans who construct lead (tenor) steelpans generally attempt to tune the first three resonances of each note harmonically. When successful, the tuning leads to neighboring notes sharing the same frequency for the first two harmonics of each note on the outer ring of the steelpan. The sympathetic excitation of the neighboring notes is believed to contribute significantly to the unique timbre of the steelpan's characteristic sound. A series of custom steelpans with fewer notes was commissioned to understand better the nature of the coupling between the struck note and sympathetically vibrating notes. Electronic speckle pattern interferometry was used to compare the coupling between notes on standard tenor steelpan with the coupling between notes on the customized steelpan having fewer notes.

2:05

4pMUa4. Audio features of steelpans with few notes. Colin Malloy (Music Dept., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, malloyc@uvic.ca)

Tenor steelpans are unusual, even among percussion instruments, in that all notes share a common surface. This physical coupling between the notes causes complex interactions between notes when activated. In an effort to better understand how these interactions work, we commissioned the building of three custom steelpans with one, two, and four notes instead of the typical 29. The one note pan has a single note in its standard position and the rest of the sunken bowl is empty. The two note steelpan adds a second note one octave higher. The four note steelpan further adds two notes a musical fifth higher. With the reduction in notes, the interactions are minimized and we can see how they affect the timbre of a full steelpan. In this paper, the results of audio feature extraction performed on these custom instruments is presented and compared with the audio features from a full range steelpan. Through this we gain insight into how octaves and fifths affect a note's timbral characteristics and lead to the attributes of a full steelpan.

Contributed Papers

2:25

4pMUa5. Modeling the sound radiation of gamelan gongs using closed-form rigid spherical models. Samuel D. Bellows (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, samuel.bellows11@gmail.com), Dallin T. Harwood, Micah Shepherd, Kent L. Gee, and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The structural modes of gamelan gongs have clear connections with the gongs' far-field radiated patterns. However, the instruments' unique geometry and modal characteristics limit the applicability of simple theoretical closed-form models, such as a radially vibrating cap on a sphere, for understanding their radiation. This work develops and applies two different models, a vibrating cap on a spherical shell with a circular aperture and a vibrating cap with imposed mode shapes, to better understand the gongs' directional characteristics. The models agree with acoustical measurements, predicting dipole and cardioid-like patterns and lobes formed from constructive and destructive interference.

2:40

4pMUa6. A comparison of the timbral characteristics of steel and wooden drum sets. Colin Malloy (Music Dept., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, malloyc@uvic.ca)

Although it is quite common to construct snare drum shells from various metals, it is still rare for full drum sets to have shells made from metal. Several manufacturers have produced metal drum sets intermittently since the 1970s. However, they still remain a niche version of the instrument that has received less attention than their wooden counterparts. This paper compares the timbral characteristics of a drum set made from steel against a maple drum set. Audio samples of the two drum sets are meticulously recorded in a recording studio with high grade equipment. The configurations of the two drum sets are made to match as closely as possible. Then the recorded audio samples are analyzed using computational methods—looking at both time-domain and spectral features for both instruments. In this way we observe how the different construction material and methods result in contrasting timbral characteristics.

2:55–3:25
Panel Discussion

THURSDAY AFTERNOON, 11 MAY 2023 LOS ANGELES/MIAMI/SCOTTSDALE, 4:20 P.M. TO 5:20 P.M.

Session 4pMUb

Musical Acoustics: Acoustics of Percussion Instruments Steelpan Ensemble Performance

Andrew C. Morrison, Cochair
Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431

Colin Malloy, Cochair
Music, University of Victoria, 3330 Richmond Rd., Victoria, V8P 4P1, Canada

THURSDAY AFTERNOON, 11 MAY 2023 CHICAGO C, 1:00 P.M. TO 5:30 P.M.

Session 4pNS

Noise, Engineering Acoustics, Computational Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Validation of Environmental Noise Modeling

Joseph Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, STE 209, Woodbridge, NJ 07095

Eric Reuter, Cochair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

Invited Papers

1:00

4pNS1. Environmental noise modeling: A case study near the busiest airport in the world. Jessica S. Clements (Special Technologies, Newcomb & Boyd, LLP, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com) and John Garretson (Special Technologies, Newcomb & Boyd, LLP, Atlanta, GA)

This case study examines modeling and measurements taken at a project site in Atlanta, Georgia. The site is an expansion of a movie studio to encompass 18 new large sound stages, 8 fixed stages, and support spaces. The site is located just inside the Atlanta I-285 perimeter and directly under a major flight path from Hartsfield Jackson Atlanta International Airport. The studio includes 8 existing sound stages located 3 miles from the expansion. Both long term and short term measurements were take and used to confirm the environmental noise modeling. Modeling was used to predict the impacts on the site and inform the building shell design. At late stages of the project, the Department of Transportation purchased land adjacent to the site and planned to regrade and move a major exit ramp closer to the site. This case study will review the multiple challenges of the project and the environmental noise modeling performed. The project is on-going and not in construction.

1:20

4pNS2. Loudspeaker modeling using CadnaA in an urban environment. Valerie Smith (Salter, 130 Sutter St., San Francisco, CA 94104, valerie.smith@cmsalter.com), Jason Duty, and Jeremy Decker (Salter, San Francisco, CA)

CadnaA is traditionally used to model environmental noise sources, such as traffic, trains, and industrial noise sources. In this presentation, we will talk about past experiences using CadnaA to model outdoor noise levels from loudspeakers at music venues in an urban environment. This noise modeling was validated by comparing to “real world” measured data.

1:40

4pNS3. Utility and transportation noise modeling. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, Ste. 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

A few case studies regarding public utility and transportation noise are presented. These address concerns regarding the prediction and attenuation of noise from electrical transformers, gas regulators, and interstate highways. Our approach to computer modeling, model adjustments to match measured conditions, and before/after measurements will be presented.

2:00

4pNS4. Validation of sound propagation from outdoor music venues—A comparison of measurement and models. David S. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

The post pandemic United States has seen a surge in new outdoor entertainment venues springing up with various levels of planning in regard to sound impact on communities. The variability of sound systems, artists, audio engineers and operators, low frequency content ratios, and weather conditions beg the need to predict potential impact of venues on communities and understand what sound management is possible. Two venues will be shown with controlled sound system tests measured compared with propagation computer models. In addition, iterative design efforts with architectural noise control to reduce impact on the nearby community will be shown, in particular an attempt to reduce low frequencies and the limits of modeling standards in regard to diffraction.

2:20

4pNS5. Roadway noise field measurements and modeling large sites in CadnaA. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Steph Ahrens (DLR Group, Omaha, NE)

Modeling roadway automobile traffic noise in CadnaA prediction software offers the opportunity to extrapolate larger site information from a small sample of field measurements. This presentation will discuss two case studies where multiple short term field measurement locations were used to calibrate large site noise analyses. Using publicly available Average Annual Daily Traffic (AADT) counts from local municipalities as the basis for site calculations was attempted but found to not correlate well with field measurements. A review and discussion of the differences will be presented.

2:40

4pNS6. Comparison of soft ground and miscellaneous attenuation in pastureland of the American West. Robert D. Miller (Threshold Acoust., LLC, 141 W. Jackson Blvd, Ste. 2080, Chicago, IL 60604, rmiller@thresholdacoustics.com)

Environmental noise measurements were performed in American West pastureland including incremental measurements over soft ground and miscellaneous attenuation through thick willow brush. Measurements will be compared with standard prediction methods such as ISO 9613. Additional distant highway measurements complimented by an ambisonic microphone will also be discussed.

3:00–3:20 Break

3:20

4pNS7. Verification of an acoustic model of outdoor sound propagation from a natural resource compressor station over complex topography. James E. Phillips (Intertek, 4703 Tidewater Ave., Ste. E, Oakland, CA 94601, james.phillips@intertek.com)

Outdoor sound propagation from a natural resource compressor station with multiple, large, reciprocating compressors enclosed within a structure was modeled using DGMR iNoise. Data from sound level measurements taken near the station were used to estimate the sound power of the operating compressor station equipment and used as input to the model. The model was then used to project the sound pressure levels at multiple measurement locations over complex topography. Good agreement was achieved between the projected and measured sound pressure levels as far as $^{1}/_{2}$-mile from the station, particularly after accounting for meteorological influences upon sound propagation in the field. Observations and lesson learned while measuring and modeling the sound propagation will be discussed.

3:40

4pNS8. Sound barrier performance: Lab verus reality/new product development = lessons learned. Craig Cook (Sales, AIL Sound Walls, 102 W Hill St., Decatur, GA 30030, ccook@ailsoundwalls.com)

In this session, we’ll explore the real world performance of sound barriers versus their lab testing along with lessons learned while developing a new upgraded reflective acoustic panel.

4:00

4pNS9. Validation of environmental noise models from a manufacturer’s persepctive. David J. Smith (Scantek, Inc, 6430 Dobbin Rd., Ste. C, Columbia, MD 21045, d.j.smith@scantekinc.com) and Antonio Notario (DataKustik, Madrid, Spain)

Validation is a critical part of the modelling process. Without this stage, it may be impossible for users to tell how close their model is to the real world. Validating an environmental noise model can be tricky, but there are some important ideas to keep in mind that can help you. DataKustik, the developers of the environmental noise prediction software CadnaA, and Scantek, Inc., the sole distributor of DataKustik products in North America, would like to offer some dos and don’ts of validating an environmental noise model from a manufacturer’s point of view.

4:20

4pNS10. Environmental noise modeling, quality control, verification, and validation concepts. Hans J. Forschner (Navcon Eng. Network, 701 W. Las Palmas Dr., CA 92835, forschner@navcon.com)

Over the past 30 + year's noise modeling has developed in all engineering fields. ISO 9001 has developed detail simulation validation criteria's for variouse engineering fields. Engineering software application for environmental noise modeling are just one of many fields that offer advanced simulation options. Unfortunately, the validation of environmental noise simulation is still under development. With the increase of environmental noise projects the model validation between test and simulation is scrutinized by the public or in court. The presentation discusses the definition of Model, Mathematical Model, Computational Model and Conceptual Models and the definition of Verification, Validation and Updating. What are the tools for quality assurance (QA) for the noise modeling software, software verification and project quality control (QC)? What is the state of noise modeling standards in North America? What are the uncertainties and variabilities that the environmental noise engineer can control? Are we sure that we are modeling a worst-case scenario? What can consultants do to improve the quality of noise simulations.

4:40

4pNS11. Multi model environmental noise projects—What are we doing? Hans J. Forschner (Navcon Eng. Network, 701 W. Las Palmas Dr., CA 92835, forschner@navcon.com)

For large-scale community projects the environmental noise engineers is charge to develop a noise model that can consist of various types of noise sources. The project may require the consideration of road, light commuter rail/commercial rail traffic, construction (stationary, moving) and operation commercial manufacturing. In North America, the consulting community uses various standards that are dramatically different in the detail of the noise emission characterization, frequency range and the noise propagation algorithms. The presentation discusses difference between TNM, FTA/FRA and ISO9613 and raise awareness of the underlying void of consistent standards in North America. The EU Noise Directive gave the European Acoustic Societies a push to developed a consistent set of standards. The variety of standards and methodologies was detrimental to modeling of END projects.

Contributed Papers

5:00

4pNS12. Managing community noise impacts from blasts. Gethin Manuel (School of Sci., Eng. & Environment, Univ. of Salford, Newton Bldg., The Crescent, Salford, Greater Manchester M5 4WT, United Kingdom, g.w.manuel@edu.salford.ac.uk) and David Waddington (School of Sci., Eng. & Environment, Univ. of Salford, Salford, Greater Manchester, United Kingdom)

The aim of this work is to examine noise impacts at long range from a variety of blast and explosion testing. This work forms part of an ongoing collaborative research project on blast noise supported by DNV and carried out at their Spadeadam Testing & Research site in the UK. The site carries out full-scale major hazards research to support safety concerns associated with industry decarbonization sectors and protect people and property from accidental and intentional threats. The diversity and novelty of blast testing involve a variety of source characteristics that need to be understood to make predictions of long-range environmental noise at off-site residential and recreational locations. Range-dependent ground impedance and complex topography surround the site, and local meteorological features are difficult to predict in advance of testing, leading to difficulties in making long-range sound predictions via computational methods. Presented here is a data-driven heuristic method developed upon a database of off-site noise measurements and meteorological data. It is concluded that meteorological forecasts can be used to make useful predictions of noise from a variety of blast testing at long-range receivers. It is proposed that further measurements at additional receptor points are made to improve the performance of the model.

5:15

4pNS13. Assessment of sound propagation for the urban air mobility. Patrice R. Malbéqui (Daaa, Onera, 29 Ave. Div. Leclerc, Châtillon 92320, France, patrice.malbequi@onera.fr)

The noise generated by vertical takeoff and landing vehicles (VTOL) operating in urban environnement is a determinant factor for the community acceptance. The ray-tracing method used for predicting road and raylway noise in urban areas can also be applied to the Urban Air Mobility context. In densely populated urban area, the multipath of sound wave occurs due to numerous reflections on the buildings walls. Such a configuration generates both constructive and destructive interferences, given rise to a complex shape of the spectra, that can be difficult to explain. To assess ray codes dedicated to road noise for this kind of configurations, simple test cases such as a corner and a canyon based on image sources are proposed. The prediction is performed in octave bands in the frequency range 63 Hz to 8000 Hz, including acoustical source at high altitudes to account for the VTOL flight. A good agreement is found between the reference solution derived from the image-source and the ray tracing technique. In particular close to the corner a significant increase of about 9 dB can be found. When both the source and the microphone are in the canyon, an inscrease up to 14 dB can be found.

Session 4pPA

Physical Acoustics: Acoustic Propagation Effects

Joel B. Lonzaga, Cochair
Structural Acoustics Branch, National Aeronautics and Space Administration, 2 N. Dryden St. (MS 463), Hampton, VA 23681

William Doebler, Cochair
NASA Langley Research Center, NASA Langley Research Center, MS 463, Hampton, VA 23681

Contributed Papers

1:30

4pPA1. Turbulence effects on shaped booms: Central composite design of modeled atmospheric turbulence parameters for sonic boom propagation. William Doebler (NASA Langley Res. Ctr., NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov), Peter A. Parker (NASA Langley Res. Ctr., Hampton, VA), Robert Downs (Environ. Measurement and Modeling Div., Volpe National Transportation Systems Ctr., Cambridge, MA), and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

Propagation of sonic booms through turbulence reduces mean sonic boom perception metric levels and also causes considerable variability. NASA's PCBoom suite of sonic boom acoustic propagation modules includes an approximate method for accounting for the effects of turbulence on traditional N-wave sonic booms. The current implementation is ineffective for shaped sonic booms or low-booms, and it also has limited values for turbulence and ambient input parameters. NASA's future X-59 low-boom community noise surveys require an accurate estimate of the effects of turbulence in regions across the USA, so the module must be improved. This work presents the methods of selecting which ambient and turbulence parameters should be included in an improved PCBoom turbulence module. Turbulence and ambient data were collected from two atmospheric model databases, the Climate Forecast System Version 2 and European Centre for Medium-Range Weather Forecast Reanalysis Version 5 (ERA5), hourly from 7 AM to 7 PM local time for 10 years at 19 locations across the USA. A fully-factorial propagation analysis using these parameters would be exceedingly computationally expensive. Instead, a central composite design was chosen resulting in 45 combinations of ambient and turbulence parameters. These 45 cases effectively sample the space balancing computational burden.

1:45

4pPA2. Turbulence effects on shaped booms: Propagation simulations using KZKFourier. Robert Downs (Environ. Measurement and Modeling Div., Volpe National Transportation Systems Ctr., 55 Broadway, V-324, Cambridge, MA 02135, robert.downs@dot.gov), Michael Barzach (Environ. Measurement and Modeling Div., Volpe National Transportation Systems Ctr., Cambridge, MA), William Doebler, Peter A. Parker, Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA), and Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, Hampton, VA)

Upcoming X-59 aircraft flight tests as part of NASA's Quesst Mission are expected to occur in a range of atmospheric conditions. Sensitivity of ground waveform acoustic metrics to turbulent perturbations through which booms propagate is a complicating factor in determining noise levels. A series of simulations through turbulence was executed using the KZKFourier model of Stout *et al*. to develop a database of results with which to expand functionality of NASA tools for estimating turbulence effects on shaped-boom ground waveforms. For 45 cases covering a seven-factor design space, multiple realizations of turbulence were generated to characterize statistical turbulence effects on levels of six acoustic metrics. Each individual simulation produced over one thousand waveforms across a virtual microphone array. Different measures were used to evaluate refinement of results with increasing number of simulations. Data suggest that mean effects and variability were most strongly influenced by propagation distance as well as velocity fluctuation intensity, and that, although local increases in acoustic metrics were common, the overall average result in all cases was a reduction in metric levels. Sensitivity of acoustic metrics varied, with mean reductions in Perceived Level occurring in the range of up to 2 dB across the 45 cases.

2:00

4pPA3. Turbulence effects on shaped booms: Finite impulse response filter development. Robert Downs (Environ. Measurement and Modeling Div., Volpe National Transportation Systems Ctr., 55 Broadway, V-324, Cambridge, MA 02135, robert.downs@dot.gov), Michael Barzach (Environ. Measurement and Modeling Div., Volpe National Transportation Systems Ctr., Cambridge, MA), Jacob Klos (NASA Langley Res. Ctr., Hampton, VA), Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, Hampton, VA), William Doebler, Peter A. Parker, and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

Numerical simulations of propagation through turbulent atmospheres can quantify effects on ground waveforms, but such simulations are computationally expensive. To enable quick turnaround analyses as required by NASA's Quesst Mission, updating the N-wave filtering approach developed by researchers at The Pennsylvania State University to include shaped booms is proposed as an alternative method for estimating turbulence effects on acoustic metrics more quickly. Beginning with a nearfield pressure cylinder modeled after the on-design X-59 configuration, a database of propagation results at 45 turbulence conditions was compiled using nonlinear turbulence propagation modeling code (KZKFourier) and used as input to a process for generating finite impulse response (FIR) filters. Ground waveforms distorted by turbulence were selected to represent mean and mean-standard deviation levels for six metrics, and corresponding FIR filters were generated through a matrix deconvolution process. In order to evaluate how well the FIR filters perform, additional KZKFourier verification cases were devised with different input conditions, and results used as a benchmark. Convolution of shaped boom waveforms modeled using nonturbulent propagation simulations with the new FIR filters showed better agreement on average with KZKFourier statistical results than the N-wave-based FIR filters.

2:15

4pPA4. Influence of ground blocking on the acoustic phase variance in a turbulent atmosphere. Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), D. Keith Wilson, and Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

A number of previous studies have analyzed the impacts of anisotropy and inhomogeneity on sound propagation in a turbulent atmosphere. However, these studies did not address the potentially important factor of ground blocking, namely the inhibition of buoyancy-produced velocity fluctuations by the ground surface. The current presentation accounts for this effect following the reference D. K. Wilson, "A three-dimensional correlation/spectral model for turbulent velocities in a convective boundary layer," *Boundary-Layer Meteorol.* **85**, 35–52 (1997). The temperature and shear-produced velocity fluctuations are not significantly blocked by the ground and are modeled as in the study by V. E. Ostashev and D. K. Wilson [*Acoustics in Moving Inhomogeneous Media*, 2nd ed. (2015)]. The resulting turbulence model is used to calculate the variance of the phase fluctuations for vertical and slanted sound propagation in the atmosphere. The calculations are done without the Markov approximation which might not be applicable for the largest turbulence eddies in the atmospheric boundary layer such as those due to buoyancy-produced velocity fluctuations. Theoretical formulations for the phase variance are compared with the data in a recent experiment involving sound propagation from a ground-based source to microphones installed at different heights of a 135-m tower.

2:30

4pPA5. Nonlinear acoustic propagation in low-density media. Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, 2 N. Dryden St. (MS 463), Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

Nonlinear acoustic propagation in the atmosphere is usually modeled using an augmented Burgers equation accounting for atmospheric absorption and weak nonlinearity. However, the weak-nonlinearity assumption may not apply to acoustic signals propagating from the lower atmosphere that are subsequently refracted downward from the upper atmosphere (e.g., stratosphere and thermosphere) due to the decreasing air density with increasing altitude [Lonzaga, *et al*., *Geophys. J. Int*. **200**(3), 1347–1361]. Consequently, this paper discusses the effects of a strong nonlinearity that lead to an amplitude-dependent increase in signal propagation speed. This propagation speed is obtained using a perturbation expansion where the leading-order term is simply the small-amplitude sound speed while the first-order term is the existing expression that gives rise to waveform steepening and stretching. Furthermore, the second-order term is proportional to the spectral density of the acoustic signal and is inversely proportional to the local density of the medium. Consequently, for an impulsive signal such as a sonic boom or an infrasound, the second-order effect causes a dispersion of the signal similar to the observed dispersion of acoustic signals from supersonic Concorde as well as from large explosions. This paper also discusses the extension of these results in general low-density propagation media.

2:45

4pPA6. Acoustic scattering from rough elastic cylinders partially insonified by directional sonars: Effects of rms roughness and correlation length. Miad Al Mursaline (Mech. Engineering/ Appl. Ocean Phys. & Eng., Massachusetts Inst. of Technology/Woods Hole Oceanographic Inst., 77 Mass Ave., Cambridge, MA 02139, miad@mit.edu), Timothy K. Stanton, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

A modal-series-based model for acoustic scattering from smooth and rough elastic cylinders insonified by directional sonars is derived. Sonar directivity, Fresnel zone effects, spherical spreading of the incident field, and axially propagating guided waves excited by this spreading are included while end-effects are assumed to be negligible. The performance of the scattering model is evaluated against laboratory monostatic and bistatic measurements of scattering from both smooth and rough elastic cylinders. Three aspects of scattering — overall scattering levels, resonance frequencies, and resonance shapes — are analyzed both theoretically and experimentally. The roughness on the cylinder has a Gaussian distribution and is characterized by a random variation of the cylinder radius along its length. Effects of various statistical properties of this one-dimensional roughness, such as different rms roughness for a given correlation length and different correlation lengths for fixed rms roughness, are investigated.

3:00–3:15 Break

3:15

4pPA7. Atmospheric acoustic propagation modeling using heterogeneous wind profiles along the acoustic path. Andrea Vecchiotti (Mech. Eng., Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, DC 20064, vecchiotti@cua.edu), Teresa J. Ryan, Jeff Foeller (Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents atmospheric sound propagation predictions using a parabolic equation solver that accounts for heterogeneous wind profile distribution along the acoustic path. Transmission loss predictions using both homogeneous and heterogeneous wind speeds information are compared with data. A three-dimensional scanning Doppler LIDAR wind profiler captures real-time wind speed gradients at many locations along the acoustic propagation path providing the heterogeneous wind speed profiles. The wind measurements are concurrent with a pitch catch transmission loss measurements. A long-range acoustic device on an anchored pontoon sends known chirp sequences to a seven-channel receiver array at the water's edge at ranges up to approximately one kilometer. Additional synchronized meteorological observations include temperature, humidity, and wind measured with anemometers. Key differences in model results are highlighted and an assessment of the value of the computational cost is presented.

3:30

4pPA8. Parabolic equation simulation of diffraction effects in a sound beam propagating through a flowing fluid. Daudel Tchatat Ngaha (Elec. Eng., Western Norway Univ. of Appl. Sci., Inndalsveien 28, Bergen 5063, Norway, dantc@hvl.no) and Kjell E. Frøysa (Elec. Eng., Western Norway Univ. of Appl. Sci., Bergen, Norway)

Ultrasonic transit time difference flow meters are today industrially accepted for custody transfer measurements of oil and for natural gas. Such meters are currently planned to be used also subsea and in remote operations, where the calibration possibilities are few. In such applications the speed of sound measured by these meters will be a powerful input for estimation of density and calorific value of the flowing oil or gas. The ultrasonic transit time measurements in such meters are carried out in a flowing oil or gas, over a range typically between 6 and 40 in. For precise transit time measurements over such ranges, diffraction corrections may be of high importance. Diffraction effects for an acoustic beam generated by a uniform piston source and propagating through a flowing fluid are therefore studied numerically. The flow direction will be perpendicular to the propagation direction of the acoustic beam. The investigation is based on a narrow-angle three-dimensional parabolic equation. Effects both on amplitude and on phase will be presented.

3:45

4pPA9. Wind profiles measurements and predictions for atmospheric acoustic propagation modeling. Andrea Vecchiotti, Diego Turo, Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), Jeff Foeller (Eng., East Carolina Univ., Greenville, NC), and Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu)

This work presents a comprehensive experimental system to measure meteorological conditions with both anemometers and a three-dimensional scanning Doppler LIDAR wind profiler. The LIDAR system captures real-time wind speed gradients at many locations along a particular heading. This work describes a method for extracting a cartesian grid of wind speeds gradients from a range height indicator (RHI) scan pattern. The LIDAR derived wind profiles are compared to logarithmic wind profiles based on single point traditional anemometer measurements. Differences in wind speed profiles observed using the two methods are presented and the implications on long range acoustic propagation modeling are discussed.

4:00

4pPA10. Modeling sea state induced atmospheric sound transmission loss. Andrea Vecchiotti (Mechaincal Eng., Catholic Univ. of America, 620 Michigan Ave., N.E., Washington, DC 20064, vecchiotti@cua.edu), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents a numerical study on atmospheric sound propagation over rough rigid surfaces. The intent is to simulate acoustic propagation over water. Methods to estimate sea state induced atmospheric sound transmission loss and relative uncertainties are evaluated. In previous studies, a flat surface with an equivalent impedance was used to account for the effect of surface roughness on sound transmission loss. Equivalent impedances were estimated based on time-domain numerical simulations of atmospheric sound propagation above pseudorandom sea surfaces coherent with a Pierson-Moskowitz spectra. Estimation of equivalent impedances using time- and frequency- domain approaches are compared for cases up to sea state 4. Acoustic excess attenuation due to propagation over a rough surface was predicted by the authors, in a previous work, by correcting the excess attenuation of a propagation over a flat, perfectly reflecting surface. The correction factor was frequency and sea state dependent and included an additional term to account for the uncertainties characteristic of different sea states. Excess attenuation predictions estimated using the equivalent impedance approach are compared with those obtained from the authors' previous study. Implications of the use of these methods for source detectability determinations are discussed.

4:15

4pPA11. Using machine learning to identify and assess cloud coverage. Hannah Blackburn (Eng., East Carolina Univ., 1000 East Fifth St., Greenville, NC 27858, Hlblkbrn@gmail.com), Andrea Vecchiotti, Joseph Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

This machine learning project is part of ongoing longitudinal long distance atmospheric acoustic propagation research being conducted at the East Carolina University Outer Banks campus in Wanchese, NC. The overall project seeks to connect changes in the atmosphere by taking concurrent acoustic and meteorological readings and relating them to differences in sound propagation. Wide angle images of the sky are used to correlate cloud cover with concurrent near-surface temperature gradient measurements. The camera is mounted on a mast that houses the temperature logger array. Images are imported, segmented, and labeled in MATLAB and used as both a training and test image set. The results of this project enable more accurate characterization of cloud cover. This information supplements knowledge of heat flux between ground and the atmosphere which in turn supports improved modeling of long distance sound propagation.

4:30

4pPA12. Air temperature profiling over different littoral surfaces. Matthew Stengrim (Eng., East Carolina Univ., 1000 East Fifth St., Greenville, NC 27858, stengrimm19@students.ecu.edu), Nicole Obando, Hannah Blackburn (Eng., East Carolina Univ., Greenville, NC), Andrea Vecchiotti, Diego Turo, Joseph Vignola (Mechaincal Eng., Catholic Univ. of America, Washington, DC), Jeff Foeller, and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

There are many factors that must be considered in the study of atmospheric sound propagation over long distances; these include surface characteristics as well as wind speed and air temperature profiles. This work presents measurements of air temperature profiles over various surfaces. Building a catalogue of this type can enable more realistic case assumptions to be made in an atmospheric acoustic transmission loss model. Accurate air temperature profiles are critical because of the potentially significant impact on the transmission loss predictions. Air temperature profile measurements have been collected by solar radiation shielded temperature loggers mounted 1 m apart on a 7 m mast. Long duration measurements were taken to capture the variability in day and nighttime temperature differentials present over surfaces such as: gravel, vegetated shoreline, marsh grass, lawn grass, water, and asphalt. In particular, measurements performed off shore were done by placing the sensor array on the deck of a pontoon boat. The initial results revealed that this configuration captures a very low elevation warming due to re-radiation from the deck of a boat instead of the true over water temperature. This study highlights the influence of wind speed on the development of the near surface temperature inversions.

Session 4pPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session II

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All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd-numbered abstracts will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered abstracts will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

4pPP1. Perception of global properties, objects, and settings in natural auditory scenes. Margaret A. McMullin (Psych., Univ. of Nevada, Las Vegas, 4505 S Maryland Parkway, Las Vegas, NV 89154, mcmulm1@unlv.nevada.edu), Nathan C. Higgins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Brian Gygi (East Bay Inst. for Res. and Education, Martinez, CA), Rohit Kumar, Mounya Elhilali (Elec. and Comput. Eng., Johns Hopkins, Baltimore, MD), and Joel S. Snyder (Psych., Univ. of Nevada, Las Vegas, Las Vegas, NV)

Theories of auditory scene analysis suggest our perception of scenes relies on identifying and segregating objects within it. However, a more global process may occur while analyzing scenes, which has been evidenced in the visual domain. In our first experiment, we studied perception of eight global properties (e.g., openness), using a collection of 200 high-quality auditory scenes. Participants showed high agreement on their ratings of global properties. The global properties were explained by a two-factor model. Acoustic features of scenes were explained by a seven-factor model, and linearly predicted the global ratings by different amounts (R-squared = 0.33–0.87), although we also observed nonlinear relationships between acoustical and global variables. A multi-layer neural network trained to recognize auditory objects in everyday soundscapes from YouTube shows high-level embeddings of our 200 scenes are correlated with some global variables at earlier stages of processing than others. In a second experiment, we evaluated participants' accuracy in identifying the setting of and objects within scenes across three durations (1, 2, and 4 s). Overall, participants performed better on the object identification task, but needed longer duration stimuli to do so. These results suggest object identification may require more processing time and/or attention switching than setting identification.

4pPP2. Relating spatial release from masking to lateralization using interaural differences. Brittany T. Williams (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, brittany.williams@boystown.org), Margaret Miller (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 acoustic cues such as interaural differences in time (ITD) and level (ILD) contribute to the benefit of reduced interference (masking) that occurs when competing sounds (maskers) differ in spatial location from the target talker. This benefit (spatial release from masking or SRM) is well studied but the contributions of ITD and ILD to SRM during speech-on-speech masking remain poorly understood. One hypothesis regarding mixed results in the literature is that SRM primarily reflects differences in overall perceived spatial location rather than independent contributions of ITD and ILD. We evaluated this hypothesis by relating SRM directly to the perceived spatial locations of speech stimuli that were processed in five conditions: ITD only, ILD only, consistent (ITD + ILD), and opposing (ITD – ILD or ILD – ITD). We measured lateralization for speech stimuli in each condition. Lateralization was consistently strongest for consistent ITD + ILD cues, but weaker and more variable across other conditions. Finally, we evaluate a model relating individual differences in SRM to ITD/ILD-based lateralization of target and masker sounds. [Work supported by NIH R01-DC016643]

4pPP3. Effects of face masks on novel word learning in preschool children. Tina M. Grieco-Calub (Rush Univ. Medical Ctr., 4711 Golf Rd., Ste. 1100, Skokie, IL, tina_griecocalub@rush.edu), Katherine R. Gordon, Kaylah Lalonde (Boys Town National Res. Hospital, Omaha, NE), Diana M. Cortez (Rush Univ. Medical Ctr., Skokie, IL), Stephanie L. Lowry, and Grace A. Dwyer (Boys Town National Res. Hospital, Omaha, NE)

Face masks worn by talkers compromise the speech signal children use to learn language. There is debate, however, about which type of face mask will be more supportive of language learning in young children. Although clear masks provide more visual access to the talker's face than surgical masks, they distort the speech signal more than surgical masks. Our goal is to better understand the effects of different face masks on word learning in preschool-age and kindergarten-age children. To achieve this goal, we are comparing children's ability to learn novel words in conditions that vary on the spectral fidelity of speech and visual access to the talker's face to simulate the effects of different mask types. Children's performance is quantified by the number of words learned and the phonological precision of words learned. To date, twelve children have completed the protocol, and the study is ongoing. We will compare children's word-learning across conditions and evaluate the influence of individual factors such as visual speech reading skills, verbal working memory, and vocabulary knowledge on children's performance in the various conditions. These findings will have implications for supporting word learning in young children in educational and childcare settings when masks are used.

4pPP4. Horizontal-plane localization of a target in the presence of a distractor: Effect of onset asynchrony. 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), and Mallika Chadaga (Psych., Univ. of Minnesota, Minneapolis, MN)

Listeners localized a target in isolation and in the presence of a distractor. The target consisted of 10 10-ms Gaussian pulses (energy between 3–5 kHz); the distractor was either a 100- or 500-ms narrowband (1–2 kHz) noise burst. The listener, with head unrestrained, was seated in the center of 36 loudspeakers spaced every 10 degrees on the horizontal plane and obscured by screens. On each trial, the target was played through a randomly selected loudspeaker. Listeners used a laser pointer to indicate target

location. Pointer position was detected by infrared cameras and the response angle calculated. In separate conditions, distractor position was either fixed or randomized across trials, with target-to-distractor onset synchrony an additional parameter. The left-right component of responses was essentially unaffected by the presence of the distractor in any condition. With notable inter-subject variability, the front-back response component also showed little effect of the distractor. This absence of consistent effect of the distractor contrasts with previously reported work. The target and distractor in the present research, however, were generally more discriminable than in past studies, which likely mitigated at least some of the deleterious effect of the distractor on target localization ability.

4pPP5. Benefits of long-term music training for segregation of competing speech by tonal language speakers. Yang-wenyi Li, Xiaoting Cheng, Chenru Ding (Dept. of Otology and Skull Base Surgery, Eye Ear Nose and Throat Hospital, Fudan Univ., Shanghai, China), John J. Galvin (House Inst. Foundation, 1127 Wilshire Blvd., Ste. 1620, Los Angeles, CA 90017, jgalvin@hifla.org), Bing Chen (Dept. of Otology and Skull Base Surgery, Eye Ear Nose and Throat Hospital, Fudan Univ., Shanghai, China), and Qian-Jie Fu (Dept. of Head and Neck Surgery, David Geffen School of Medicine, UCLA, Los Angeles, CA)

Extended experience with meaningful pitch information has been shown to benefit music perception as well as speech perception where pitch cues are important, such as segregation of competing speech and tonal language perception. Interestingly, pitch perception has been shown to be similar between non-musicians who speak a tonal language and musicians who speak a non-tonal language, both of which outperform non-musicians who speak a non-tonal language. However, it is unknown whether extensive music training can further benefit pitch perception in tonal language speakers. In this study, melodic contour identification, spectro-temporal pattern perception, and masked speech recognition was measured in 16 adult normal-hearing musicians and 16 non-musicians; all were Chinese native speakers of Mandarin. Melodic contour identification, spectro-temporal pattern perception, and masked speech recognition all were significantly better for musicians than for non-musicians. Compared to non-musicians, musicians better utilized talker sex cues to segregate competing speech; utilization of talker sex cues by musicians was associated with the onset and extent of music training. Across all participants, spectro-temporal pattern perception was associated with better masked speech understanding. The data suggest that early and extensive music training may further benefit tonal language speakers' perception and utilization of pitch cues.

4pPP6. Exploring the parameter space of the modified rhyme test to improve efficiency. Gregory M. Ellis (Audiol. and Speech Lang. Pathol., Walter Reed National Medical Military Ctr., 2240 Campus Dr., Evanston, IL 60208, gellis@alakaina.com) and Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed National Medical Military Ctr., Bethesda, MD)

The modified rhyme test (MRT) is a words-in-noise test that is used as a part of screening and testing procedures within the United States military. Each trial has a carrier phrase followed by a target word ("Please select the word CAT"). Listeners choose the target from a set of six options. Competing words differ only in the starting or ending consonant sound (e.g., "BAT," "VAT," "SAT," "RAT," and "THAT"). The present study had two goals. The first goal was to improve the efficiency of the MRT by eliminating the carrier phrase and providing only the test word. This should quicken data collection. The second goal was to determine the effect of reducing the number of alternatives. Reducing the number of alternatives should reduce response time, but may affect data quality. Aside from the primary goals, other parameters were adjusted: signal-to-noise ratio, presentation level, noise type, and filtering condition. Over 3000 service members participated in the study from multiple sites. The MRT was administered on calibrated tablet and headphones. Findings will be discussed. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy of the Department of Defense or the U.S. Government.]

4pPP7. Single-trial decoding of stimulus pitch from EEG-FFR. Letitia Ho (Psych., Univ. of Chicago, 5848 S University Ave., Chicago, IL 60637, letitiayhho@uchicago.edu), John Veillette, and Howard Nusbaum (Psych., Univ. of Chicago, Chicago, IL)

The frequency-following response (FFR) is a phase-locked evoked response recorded at the scalp that directly mirrors the frequency content of acoustic stimuli. While once believed to be primarily of subcortical origin, MEG and EEG research has more recently shown that the FFR originates from multiple subcortical and cortical neural sources (Coffey *et al.*, 2016; Hartmann and Weisz, 2019; Tichko and Skoe, 2017). The present study tests whether the frequency of the stimulus can be reliably decoded from frequency-specific power of single-trial high-density EEG, despite the extremely small amplitude of the FFR relative to the background EEG. Since the amplitude of the FFR contributions from the multiple constituent cortical and subcortical sources can be modulated independently, it is reasonable to posit that the scalp distribution of the FFR may be malleable to psychological processes such as selective attention. Past studies on the attentional modulation of the FFR have been inconclusive (e.g., Forte *et al.*, 2017; Hoormann *et al.*, 2004). We therefore also examine the accuracy of the single- trial FFR-based decoding as a function of different attention manipulations.

4pPP8. Modelling the hydrodynamic behaviour of the cochlea: is the stiffness of the Reissner's membrane important for frequency selectivity? Arthur Vermeulen (Netherlands Defence Acad., Het Nieuwe Diep 8, Den Helder 1781 AC, the Netherlands, af.vermeulen@gmail.com) and Henk Knoll (Netherlands Defence Acad., Den Helder, the Netherlands)

The cochlea contains two membranes, the Reissner's membrane and the Basilar membrane. In most models of the cochlea only the Basilar membrane is considered. The displacement of the Basilar membrane is crucial for stimulating the sensory cells within the cochlea. The omission of the Reissner's membrane is often justified because it is believed to be floppy, and it makes the calculation of the travelling wave on the Basilar membrane analytically possible. In the present work we have looked into the influence of the stiffness of the Reissner's membrane on the frequency selectivity of the Basilar membrane with a finite-volume model of the cochlea. Simulations are performed with a logarithmic distribution of the Basilar membrane stiffness. Each segment of the Basilar membrane oscillates with an increasing amplitude for a different frequency of the stimulus (the local resonance frequency) and shows the expected frequency selectivity. Our simulations indicate that the local resonance frequency does not change with the stiffness of the Reissner's membrane as long as the sum of the stiffness of the Basilar membrane and the Reissner's membrane remains the same. Consequently, the division of stiffness over both membranes has no impact on the frequency selectivity of the Basilar membrane.

4pPP9. The effect of explicit and implicit voice training on speech-on-speech intelligibility and listening effort. Ada Bicer (Dept. of Otorhinolaryngology/ Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, the Netherlands), Deniz Baskent (Dept. of Otorhinolaryngology/ Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Hanzeplein 1, PO BOX 30.001, Groningen 97100RB, the Netherlands, d.baskent@rug.nl), Carolyn McGettigan (Speech, Hearing and Phonetic Sci., Div. of Psych. and Lang. Sci., Univ. College London, London, United Kingdom), and Thomas Koelewijn (Dept. of Otorhinolaryngology/ Head and Neck Surgery, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, the Netherlands)

Listening to familiar voices might improve intelligibility of target speech in multiple-talker situations. In addition to personally familiar voices, voice training—through implicit exposure or explicit learning of a previously unheard voice—can improve speech intelligibility. However, there is no consensus on which method is more effective. We investigated the effect of explicit and implicit voice training on speech-on-speech perception and listening effort (pupillometry), among normal hearing listeners. There was no significant difference in speech intelligibility performance between trained

and untrained voices, with either voice training method. However, an interaction between voice training method and target-to-masker ratio (TMR) on accuracy scores showed that the benefit of increased TMR on accuracy scores was significantly larger when voice training was implicit compared to explicit. Pupillometry results showed that explicit voice training significantly reduced listening effort for most disadvantageous TMR (−6 dB), but not for more advantageous TMRs (0 and + 6 dB), and not for implicitly trained voices. Our results may imply that explicit voice training is more effective than implicit voice training in reducing listening effort. Still, implicit voice training, perhaps via procedural learning, might be more beneficial on the performance level when TMR becomes more advantageous.

4pPP10. Voice cue perception and voice gender categorisation via a humanoid robot interface. Laura Rachman (Otorhinolaryngology, Univ. Medical Ctr. Groningen, Univ. of Groningen, Hanzeplein 1, Groningen 9713 GZ, the Netherlands, l.rachman@rug.nl), Luke Meyer, Gloria Araiza-Illan (Otorhinolaryngology, Univ. Medical Ctr. Groningen, Univ. of Groningen, Groningen, the Netherlands), Etienne Gaudrain (Otorhinolaryngology, Univ. Medical Ctr. Groningen, Univ. of Groningen, Lyon, France), and Deniz Baskent (Univ. of Groningen, Groningen, the Netherlands)

Two voice cues, fundamental frequency (F0) and vocal-tract length (VTL), are important for characterising voice gender. Due to their often repetitive nature, psychophysical tests evaluating perception of these vocal cues can at times cause individuals to lose engagement during the test. To help with this, we propose the use of an interactive humanoid NAO robot as an alternative to the conventionally used laptop interface. As a first step, we compare the performance of two psychophysical tests between the robot and laptop interfaces. Experiment I measured F0 and VTL cue perception through an adaptive test in just noticeable differences (JNDs). Experiment II measured voice gender categorisation using F0 and VTL manipulated stimuli. The robot implementation made use of the tactile sensors on the hands and head of the robot for response logging. Performance accuracy between the computer and robot interfaces was functionally similar, confirming data reliability. Test duration comparison showed that both experiments were longer on the NAO robot than the laptop. Despite potential design limitations of the robot interface, both interfaces showed that the F0 and VTL JNDs are very small in normal-hearing listeners, and that normal-hearing listeners make effective use of F0 and VTL cues for voice gender categorisation.

4pPP11. Emotion perception in children with cochlear implants and with hearing aids. Deniz Başkent (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen, Groningen, the Netherlands, d.baskent@umcg.nl), Gizem Babaoğlu, Başak Yazgan, Pınar Ertürk (Audiol. Dept., Hacettepe Univ., Ankara, Turkey), Etienne Gaudrain (Lyon Neurosci. Res. Ctr., CNRS UMR5291, Inserm U1028, UCBL, UJM, Lyon, France), Leanne Nagels (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Groningen, the Netherlands), Tjeerd Jong (Dept. of Otorhinolaryngol., Erasmus Univ. Medical Ctr., Rotterdam, the Netherlands), Stefan Launer (Phonak, Sonova AG, Staefa, Switzerland), Gurjit Singh (Phonak Canada, Mississauga, ON, Canada), R. Peter Derleth (Phonak, Sonova AG, Stäfa, Zürich, Switzerland), Francien Coster (Dept. of Otorhinolaryngol., Univ. Medical Ctr. Groningen, Groningen, the Netherlands), Monita Chatterjee (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Debi Vickers (Cambridge Hearing Group, Clinical Neurosciences Dept., Univ. of Cambridge, Cambridge, United Kingdom), Petra Hendriks (Ctr. for Lang. and Cognition Groningen, Univ. of Groningen, Groningen, the Netherlands), Esra Yücel, Gonca Sennaroğlu (Audiol. Dept., Hacettepe Univ., Ankara, Turkey), Marc Schroeff, Jantien Vroegop (Dept. of Otorhinolaryngol., Erasmus Univ. Medical Ctr., Rotterdam, the Netherlands), Bert Maat, Rolien Free (Dept. of Otorhinolaryngol., Univ. Medical Ctr. Groningen, Groningen, the Netherlands), Ruben Benard (Pento Audiol. Ctr., Zwolle, the Netherlands), Evelien Dirks (Nederlandse Stichting voor het Dove en Slechthorende Kind (NSDSK), Amsterdam, the Netherlands), and Laura Rachman (Pento Audiol. Ctr., Groningen, the Netherlands)

Emotion recognition is an important part of human communication and this ability contributes to children's social development. Children with hearing loss may have difficulties perceiving relevant acoustic cues conveying emotions, but the combined effects of neuroplasticity, physiological effects from hearing loss, and the compensatory features of hearing aids and cochlear implants are not fully understood. As a result, it is also unclear if difficulties in vocal emotion perception are mostly due to a reduced access to relevant acoustic cues, an acute effect. Additionally, the ability to recognize and label emotions may develop differently in children with hearing loss, an accumulated effect over a longer period of time that may be less directly related to hearing difficulties. We will present findings from recent studies on vocal emotion recognition from semantically meaningless sentences in children (6–18 years) with cochlear implants or with hearing aids, and preliminary results from ongoing work on both vocal and facial emotion recognition in hearing aided children (6–18 years). These data show a large variability in performance, indicating some children with hearing loss may have great difficulties. We will discuss potentially relevant contributing factors in the development of emotion perception in children with hearing loss.

4pPP12. Language dependent cue-weighting of tone cues in speech and non-speech stimuli. Zhenting Liu (Linguist., The Chinese Univ. of Hong Kong, Hong Kong NT, Hong Kong 999077, Hong Kong, ztliu@link.cuhk.edu.hk) and Regine Lai (Linguist., The Chinese Univ. of Hong Kong, Hong Kong, Hong Kong)

Previous research on tone perception has identified several important pitch related cues including average pitch height (AH), contour, onset and offset, and the weighting of these cues has shown to be language dependent. However, since multiple pitch cues are covarying with each other, few studies have directly compared relative importance of these cues. It is also not clear whether there is a same ranking of cues in speech and non-speech stimuli. The current study aims to tease apart the relative role of each cue using AX discrimination. Four pairs of tone contrasts with minimal pitch differences were created. Tone contrasts within contour condition are two level tones with 7 Hz differences. Tone contrasts within AH, onset and offset conditions have one rising tone and one falling tone sharing the same AH, onset and offset respectively. If one cue is important, then when this cue is kept constant, variation in other cues should be hard to perceive. 48 Mandarin speakers and 48 Cantonese speakers were recruited. Results showed highest importance of AH for both Mandarin and Cantonese listeners and higher importance of contour (offset) than onset for Mandarin (Cantonese) listeners in speech stimuli. This ranking was not held for nonspeech stimuli.

4pPP13. The role of voice fundamental frequency in the perception of anger in clear speech. 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), Sierra N. Bennion, Tara E. Smalley, and Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

In previous work, listeners have rated clearly-spoken neutral speech materials as sounding angry significantly more often than identical materials spoken conversationally. This effect varies widely among talkers, and a recent study compared acoustic characteristics of clear and conversational speech produced by talkers whose clear speech sounds angry to those of talkers whose clear speech does not sound angry. Of several well-known clear speech acoustic changes, only raised voice fundamental frequency (f_o) differed for talkers who do and do not sound angry when speaking clearly. To test whether raised f_o causes clear speech to sound angry, the present study used six talkers whose clear speech was rated as sounding angry and who also made sizeable f_o shifts when speaking clearly. For each talker, nine pairs of identical sentences (one from each speaking style) were identified. Praat was then used to lower the pitch of the clear sentences to match the pitch of the conversational sentences and to raise the pitch of the conversational sentences to match the pitch of the clear sentences. Listeners with normal hearing heard these sentences and rated the emotion they heard in each item; shifting f_o had only minimal effects on perceived anger.

4pPP14. A systematic review of measurements of real-world interior car noise for the "Cadenza" machine-learning project. Jennifer L. Firth (Univ. of Nottingham, University of Nottingham, Nottingham NG7 2UH, United Kingdom, jennifer.firth@nottingham.ac.uk), Trevor J. Cox (Univ. of Salford, Salford, Greater Manchester, United Kingdom), Alinka Greasley (Leeds Univ., Leeds, United Kingdom), Jon P. Barker (Univ. of Sheffield, Sheffield, United Kingdom), William M. Whitmer (Hearing Sci. - Scottish Section, Glasgow, United Kingdom), Bruno Fazenda (Univ. of Salford, Salford, United Kingdom), Scott Bannister (Leeds Univ., Leeds, United Kingdom), Simone Graetzer, Rebecca Vos (Univ. of Salford, Salford, United Kingdom), Gerardo Roa (Univ. of Sheffield, Sheffield, United Kingdom), and Michael A. Akeroyd (School of Medicine, Univ. of Nottingham, Nottingham, United Kingdom)

Interior car noise refers to the general noise generated by the engine transmission, the interaction between road and types, and weather conditions such as turbulent wind. For drivers or passengers with hearing loss, these can create especially challenging listening situations. The Cadenza Project is organising a series of machine learning challenges to advance signal processing of music for listeners with a hearing loss, and a key scenario in its first challenge is listening in a car to music in the presence of noise. To create enough machine-learnable training materials we need to simulate typical car noises rather than just use one particular recording. We are systematically reviewing the literature on real-world recordings to determine the range of parameters for these simulations. We searched Web of Science with the terms "(car noise, car noise interior, interior noise) AND (speed OR FFT OR spectr*)." A total of 126 studies have been found so far and 12 papers retained on the basis that a frequency spectrum for interior car noise was provided that was suitable for numerical analysis. Results will be presented.

4pPP15. Influence of public space type on soundscape perception in the residential context. Yichun Lu (Dept. of Architecture, National Univ. of Singapore, 4 Architecture Dr., Singapore 117566, Singapore, e0554244@u.nus.edu) and Siu-Kit Lau (Dept. of Architecture, National Univ. of Singapore, Hong Kong, Hong Kong)

Many studies have validated that soundscape quality will contribute to the satisfaction of public spaces. However, the requirements for a sound environment differ in different public spaces. Few studies focus on the difference among effects of soundscape on satisfaction in various types of public spaces in residential areas. Besides, the public space type may also moderate the effect of audio-visual environment on soundscape quality, which there may be a lack of existing studies. The present research aims to investigate the relationship among space type, audio-visual environment, and soundscape to improve satisfaction in public spaces. An on-site questionnaire survey was conducted in public spaces in residential areas, and 50 subjects participated in this survey. The correlation analysis showed that the audio-visual indicators affect soundscape perception in passive and active space differently. In particular, traffic noise may have a less negative effect on pleasantness in active space than passive space. Also, interesting visual elements may enhance eventfulness in passive zone while they may not affect eventfulness in active zone. The result of the Analytic Hierarchy Process (AHP) indicated that pleasantness is most important to people's satisfaction in both passive and active spaces, while eventfulness is more critical in active than passive spaces.

4pPP16. Study of nonlinearity in a tapered, viscous cochlear model. Vipin Agarwal (Mech. Eng., Univ. of Michigan, 3632 G.G. Brown Bldg., 2350 Hayward St., Ann Arbor, MI 48109, vipin@umich.edu) and Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI)

The mammalian cochlea is responsible for transforming incoming acoustic energy into neural signals. Efficient modeling of the cochlear response is extremely challenging, because of the length scales, which vary from the sub-micron to centimeters, and time scales, which vary from microseconds to seconds, that must be resolved. In the current work, we predict the stationary nonlinear response of a base-to-apex cochlear model to a harmonic input, considering the taper of the cochlear scalae. We seek to understand the influence of bulk fluid viscosity and geometric fluid-duct tapering on the nonlinear response of the system. Coupled equations are derived from a kinematically constrained Langrangian dynamics formulation. To solve these equations, we have used an iterative algorithm, the alternating frequency-time method. The algorithm swaps between frequency and time domains using the Fourier and inverse Fourier transforms, and is based on a fixed-point iteration. We analyzed the solutions for nonlinear models where the frequencies and stimulus levels are varied from 6 kHz to 20 kHz, and 10 dB SPL to 90 dB SPL, respectively. We show that our model predicts previously unexplained responses in the cochlea, including so-called hypercompression (when increasing sound levels produce a reduced cochlear response).

4pPP17. On the influence of the middle ear on the sound pressure level generated in open and occluded earcanals under a bone-conducted stimulation. Kévin Carillo (Institut de Recherche Robert-Sauvé en Santé et Sécurité du Travail (IRSST), 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, kevin.carillo.1@ens.etsmtl.ca), Franck Sgard (Institut de Recherche Robert-Sauvé en Santé et Sécurité du Travail (IRSST), Montreal, QC, Canada), Guillaume Ramadier, and Olivier Doutres (Mech. Eng., École de Technologie Supérieure (ETS), Montreal, QC, Canada)

The occlusion effect is an acoustic discomfort commonly encountered by earplugs' users, which corresponds to the increased auditory perception of the bone-conducted sound mainly at low frequencies. This perception is accompanied by the augmentation of the sound pressure level in the earcanal when occluded by the earplug compared to the open case. While the source of the SPL generated in the earcanal in both open and occluded cases is mainly attributed to the vibration of the earcanal wall, the sound radiation of the tympanic membrane that stems from the middle ear vibration is commonly considered as not significant. To investigate this phenomenon, a finite element model that incorporates the middle ear components into a realistic human outer ear is developed. First, the finite element model is evaluated by comparing the acoustic impedance computed at the eardrum surface to literature data obtained on human subjects. Second, the finite element model is used to quantify the contribution of the sound radiation that originates from the middle ear on the sound pressure level generated in both open and occluded earcanals under a bone-conducted stimulation.

4pPP18. Evaluating compliance on a gamified auditory training task in Veterans. Jessica G. Mendiola (Commun. Sci. & Disord., Western Washington Univ., 2402 Francis Rd., Mount Vernon, WA 98273, mendioj@wwu.edu), E. S. Lelo de Larrea-Mancera (Psych. & Brain Game Ctr., Univ. of California Riverside, Cdmx, Cdmx, Mexico), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Aaron Seitz (Psych. & Brain Game Ctr., Univ. of California Riverside, Riverside, CA), and Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., Bellingham, WA)

Service members and Veterans are regularly exposed to environmental factors that have been shown to contribute to hearing deficits. These listening deficits may present as auditory processing complaints such as difficulty listening in background noise in the absence of a peripheral hearing loss. The public app, *Listen: An Auditory Training Experience*, developed by University of California Riverside Brain Game Center, aims to improve overall speech comprehension by targeting skills related to spectro-temporal modulations, spatialized sound cues, and auditory memory tasks. Ten Veterans participated in a two-week gamified auditory training program. Compliance on the auditory training tasks was assessed by monitoring the number of trials accomplished each day, accuracy above chance, reaction time, and improvements in performance over time. Improvements were also monitored for untrained auditory tasks, such as speech-in-noise performance and a battery of central auditory processing assessments.

4pPP19. Long-latency component of stimulus frequency otoacoustic emissions: Relationship to behavioral measures. Uzma S. Akhtar (Dept. of Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina, Chicago, IL 60612, uzma_akhtar@rush.edu), Arturo Moleti (Dept. of Phys., Univ. of Roma Tor Vergata, Rome, Italy), Renata Sisto (Dept. of Occupational and Environ. Medicine, Epidemiology and Hygiene, Italian Workers Compensation Authority, Rome, Italy), and Sumitrajit Dhar (Roxelyn & Richard Pepper Dept. of Commun. Sci. & Disord., Northwestern Univ., Evanston, IL)

Uncertainties regarding the generation mechanism and source of human stimulus frequency otoacoustic emissions (SFOAEs), as well as the possible generation of artifacts, have previously restricted the use of SFOAEs in clinical settings for assessing cochlear function. Over the years, advanced time-frequency analysis of SFOAEs, using models of SFOAE delays as a function of frequency, have been implemented in an effort to isolate the long-latency (LL) component of SFOAEs—improving the signal-to-noise ratio and the artifact rejection as noteworthy byproducts. In order to test the hypothesis that behavioral measures and SFOAE LL component both reflect the activity of limited cochlear regions, the relationship between SFOAE LL component and behavioral measures was investigated. SFOAEs, behavioral thresholds and psychophysical tuning curves (PTCs) were measured for probe frequencies centered around 0.75, 1, 2, 4, 8, 10, 12.5 and 14 kHz. Stimulus levels for PTCs (10 dB SL) and SFOAEs (10, 20 and 30 dB SL) were referenced to behavioral thresholds at each frequency. Using the same *in situ* calibration technique using Thevenin-equivalent source parameters, behavioral thresholds and SFOAE-estimated thresholds from input-output functions of the SFOAE LL component were strongly correlated. Furthermore, there was a strong correlation between SFOAE-based and psychophysical tuning estimates.

4pPP20. Localization of novel reduced-annoyance audio interface sounds. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., 1201 Western Rd., EC 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca) and Michael Schutz (Dept. of Psych., Neurosci. & Behaviour, McMaster Univ., Hamilton, ON, Canada)

Foley *et al*. [*JASA* **151**, 3189–3196 (2022)] have demonstrated that annoyance of auditory interface sounds can be reduced both by shortening the duration of upper harmonics and by applying percussive rather than flat amplitude envelopes. However, sounds in that study had a maximum frequency of 2400 Hz, which would likely affect localization based on spectral cues. In a setting with multiple interface devices (e.g., a multi-patient hospital ward), localization of interface sounds is a concern. We characterized normally hearing listeners' ability to localize a variety of candidate reduced-annoyance interface sounds (flat or percussive envelopes; durations between 360 and 1600 ms; all but one with uppermost harmonic limited to 2400 Hz) in quiet or at +4-dB and −11-dB SNR in spatially diffuse multi-talker babble. Listeners stood at the center of a 360-degree loudspeaker array in a darkened anechoic chamber and used a head-pointing response to report the perceived location of each target. Decreasing SNR increased response variability and the rate of front/rear confusions. For all sounds with restricted bandwidth, front/rear confusions were frequent in the absence of head movements, but when head movements were initiated before target offset, confusions were substantially reduced. The results highlight the need to consider localizability when designing improved auditory interface sounds.

4pPP21. How do perceptual benefits from musical experience compare with those from speech experience? Anya E. Shorey (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, anya.shorey@louisville.edu), Caleb J. King (Univ. of Louisville, Louisville, KY), and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Acoustic consistency facilitates perceptual processing while acoustic variability challenges it. This is evident in perceptual interference tasks, in which listeners categorize a target word/pitch when produced by either a single talker/musical instrument or multiple talkers/instruments. Responses are faster and/or more accurate for single-talker (e.g., Stilp & Theodore, 2020 AP&P) and single-instrument conditions (e.g., Shorey *et al*., 2022 ASA). While patterns of results are parallel across domains, listening experience is not. Participants in speech tasks are experts at their native language, but in the music task the contribution of musical experience is unclear because Shorey *et al*. tested solely nonmusicians. This raises two questions: (1) does more musical training increasingly protect against timbral variability when making pitch judgments, and (2) how does musical expertise compare with speech expertise? To evaluate these questions, we tested native English speaking nonmusicians (<2 years of formal musical training), intermediate musicians (2–9 years), and experienced musicians (10+ years) in speech interference and music interference tasks. We predict that musical training offers resilience to variability, such that interference in the music task will decrease with increasing musical experience. We also predict correlated interference for advanced musicians (i.e., experts) across speech and music tasks. Results will be discussed.

4pPP22. Quantifying impacts of hearing protection devices on sound localization in azimuth and elevation: Refinement of acoustic predictors. Andrew D. Brown (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, andrewdb@uw.edu), Nathaniel Greene (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), David J. Audet, Aoi A. Hunsaker (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Carol A. Sammeth (Physiol. & Biophys., Univ. of Colorado School of Medicine, Aurora, CO), Mallory A. Butler (Otolaryngol., Univ. of Colorado School of Medicine, Aurora, CO), Alexandria Podolski, Jennifer Jerding (Appl. Res. Assoc., Inc., Littleton, CO), David A. Anderson (Elec. Eng., Univ. of Minnesota Duluth, Duluth, MN), Gregory T. Rule, and Theodore F. Argo (Appl. Res. Assoc., Inc., Littleton, CO)

Modern hearing protection devices (HPDs) mitigate the risk of noise-induced hearing loss when used as intended, but negative auditory perceptual side-effects continue to limit usability in critical settings. Dozens of studies have shown that HPDs lead to significant errors in sound source localization. Several studies, including previous studies by our group, have linked such errors to distortions of the spectra of transmitted signals. Here we provide an update on a multi-site effort to develop acoustic predictors and summary metrics to quantify HPD impacts on sound source localization. At two independent study sites, human listeners localized brief broadband signals with open ears and during use of passive and active earplug and earmuff-style devices. Sources spanning 360° in azimuth and −30° to +60° in elevation enabled measurement of two-dimensional localization error across a broad range of source locations. An initial set of parallel acoustic measurements was obtained using a standardized acoustic test fixture. Present efforts are focused on evaluating prospective advantages of (1) incorporating binaural acoustic data in the prediction of errors and (2) accounting for individual HPD-impacted acoustic variability. The long-term goal of this work is to support application-specific selection of existing HPDs and inform the design of new HPDs.

4pPP23. Masking effects of amplitude modulation on frequency-modulated tones. Kelly L. Whiteford (Psych., Univ. of Psych., N218 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55414, whit1945@umn.edu), Neha Rajappa (Neurosci., Univ. of Minnesota, Minneapolis, MN), PuiYii Goh (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Andrew J. Oxenham (Psych., Univ. of Psych., Minneapolis, MN)

Human sensitivity to frequency modulation (FM) is best for low carrier frequencies (<∼4–5 kHz) and slow modulation rates (<5–10 Hz). This high sensitivity is thought to be afforded by neural phase locking to temporal fine structure (TFS). At faster rates and higher carriers, TFS cues may no longer be available, with sensitivity to FM relying instead on conversion to amplitude modulation (AM) via cochlear filtering. One way to test this hypothesis is to measure the masking produced by additional AM, with the prediction that AM masking will be more pronounced when FM is encoded via AM than via TFS cues. This study tested AM masking of FM and of "simulated FM," which involves out-of-phase AM dyads. FM detection was measured for carrier frequencies of 1 and 6 kHz and modulation rates of 2 and 20 Hz with and without AM masking. Preliminary results suggest that FM and simulated FM sensitivity is more affected by AM at fast than slow rates at both low and high carrier frequencies. Overall, the results do not

provide support for the idea that AM interference can be used to distinguish between TFS- and envelope-based codes for FM. [Work supported by NIH grant R21 DC019409.]

4pPP24. Portable automated rapid testing: Validation of automated testing on listeners from India. Prashanth Prabhu, Vardha Pattundan, Sonal Priya, Dibyendu Das (Dept. of Audiol., All India Inst. of Speech and Hearing, Mysuru, India), Chhayakanta Patro (Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD), Nirmal Kumar Srinivasan (Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Portable Automated Rapid Testing (PART) is an iPad application (https://braingamecenter.ucr.edu/games/p-a-r-t/) capable of measuring various psychoacoustic thresholds (speech on speech masking, temporal, spectral, and spectrotemporal modulation detection, temporal gap detection, frequency modulation tasks, and pure tone detection in noise) in an automated and rapid format using commercially available headphones. The app has been validated previously in native speakers of English and an adapted version of the app has been validated in young normal hearing listeners from Mexico. Here, we present psychoacoustic threshold data for a large cohort of younger college going listeners from India whose native language is not English but are proficient in English. The thresholds obtained from this group will be compared against previously published thresholds based on native speakers of English. It is hypothesized that there will not be any significant differences in thresholds between listeners from India and the thresholds published in the literature. These results will give us the required evidence to start using this freely available app to measure reliable thresholds of various central auditory processing measures in non-native speakers of English who are proficient in the language as well.

4pPP25. Vibratory responses to outer hair cell-generated distortion products along the width of the mouse organ of Corti. 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)

Mammalian hearing sensitivity depends on the amplification of cochlear vibrations by outer hair cells (OHCs). OHCs are thought to generate amplifying forces through electromotility, i.e., voltage-driven changes in cell length, though how exactly these forces influence the motion of the surrounding organ of Corti structures remains incompletely understood. Here, in the mouse cochlear apex, optical coherence tomography was used to characterize electromotility-induced vibrations along the axis of the OHCs and along the widths of both the basilar membrane (BM) and reticular lamina (RL). Two-tone stimuli were used to elicit OHC-driven vibrations at a variety of intermodulation distortion-product (DP) frequencies. For a wide range of DP frequencies, out-of-phase motions were observed between the bottoms and tops of the OHCs, as well as between the tops of the OHCs and the lateral supporting cells. However, along the BM, phase variations were only observed at frequencies above the characteristic frequency (CF), with in-phase motions being observed at lower frequencies. The data indicate that OHCs can rapidly deform the top of the organ of Corti, while the BM response is more complex, likely being dominated by traveling-wave energy at and below the CF. [Work supported by NIH/NIDCD R21 DC019209 and the Hearing Health Foundation.]

4pPP26. Group conversation assessment in realistic acoustic scenes. Stefan Klockgether (R&D, Sonova AG, Laubisrütistrasse 28, Stäfa, Zürich 8712, Switzerland, stefan.klockgether@sonova.com), Manuel Cattaneo, Robin Weiss (ICAI Interdisciplinary Ctr. for Artificial Intelligence, Eastern Switzerland Univ. of Appl. Sci., Rapperswil, Switzerland), and R. Peter Derleth (R&D, Sonova AG, Stäfa, Zürich, Switzerland)

Successful communication in background noise is a continuous interaction between conversation partners. They continuously exchange conscious, unconscious, verbal, and non-verbal information regarding how well the other person is understood. Human communication is not limited to one-on-one interactions but can also happen in group conversation situations. The complexity of taking an active part in a group conversation is much higher though, since the continuous interaction happens with more than one conversation partner at the same time. To investigate human communication behavior, real conversations between groups of six people were observed in the Sonova Real Life Lab. All participants were allowed to move around freely within the lab space. The head position and orientation of all six participants was continuously monitored with an optical motion capturing system. The voices of all participants were recorded with wireless headset microphones, which allowed to assess individual vocal effort. During the conversation situation, background noise was played from all four directions simultaneously and systematically varied in overall level. The individual position data and vocal effort were used to estimate perceived communication difficulty and signal to noise ratios. The head orientation data was analyzed to estimate which individuals the participants were talking and listening to.

4pPP27. Rate dependence in outer hair cell mediated active processes: Determining Prestin's Speed Limit. Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI) and Wen Cai (Mechancial Eng., Univ. of Michigan, 2350 Hayward St., 3646 G. G. Brown Bldg., Ann Arbor, MI 48109, caiwen@umich.edu)

The electromotility of the outer hair cell (OHC) contributes to the sensitivity of the mammalian cochlea by amplifying traveling waves through electrical-to-mechanical energy conversion realized at the molecular level by an electromotile protein called prestin. Rate-dependent effects, including viscous damping, transmembrane electrical impedance, and state-dependent conformal transitions, hold the potential to attenuate OHC-mediated active processes at high frequencies thereby rendering prestin ineffective in high frequency cycle-by-cycle amplification. Determining the upper frequency limit of prestin remains a central challenge in cochlear biophysics. In this study, we will build a simplified OHC model to explore the influence of the rate dependence on active force generation and power deposition. Through numerical investigations, the proposed model can be used to model charge and electromotility data from *in vitro* experiments, but subtle variations in the rate parameters significantly change the predictions of electromechanical force as well as the power deposition. Based on these theoretical considerations, we propose an experimental approach to consistently determine the rate dependence and other OHC response parameters by characterizing the electrical and mechanical behavior about a resting position. This approach holds the potential to conclusively determine the upper frequency limit of prestin under physiological resting conditions.

4pPP28. A molecular psychophysical approach to identify cues for tone-in-noise detection in simultaneous masking. Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84112, skyler.jennings@hsc.utah.edu), Douglas M. Schwarz, Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY), and Kenneth S. Henry (Otolaryngol., Univ. of Rochester, Rochester, NY)

Experiments involving the detection of a tone presented in noise form the basis of many models of psychophysics that are used to estimate frequency resolution and cochlear compression for listeners with normal and impaired hearing. Despite the wide-spread application of tone-in-noise experiments, the cues used by listeners to detect the tone are a matter of debate. Here we adopt a molecular psychophysics approach – decision variable correlation (DVC) – to evaluate the extent to which trial-by-trial responses are consistent with candidate decision variables, including stimulus energy or the temporal envelope. We measured detection thresholds in normal-hearing adults for a 1000-Hz tone presented in one-third octave noise centered on the tone frequency. The tone and noise were gated simultaneously and thresholds were measured for durations of 10, 20, 50, and 100 ms. We limited the reliability of energy-based cues by incorporating a roving-level paradigm. Similarly, the use of short tones (e.g., 10 ms) limited the reliability of envelope-based cues. Our results reveal that listeners adopted a strategy that emphasized energy-based cues for short-duration tones and envelope-based cues for long-duration tones. This finding suggests that models of human psychophysics may benefit from incorporating a decision device that adjusts cue weights based on stimulus duration.

4pPP29. Is the rapid decline in interaural time difference sensitivity above 700 Hz explained by downward spread of excitation into the frequency dominant 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), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Daniel J. Tollin (Physiol., Univ. of Colorado School of Medicine, Aurora, CO)

Low-frequency temporal-fine-structure interaural time differences (ITDs) are so important that they almost completely dominate localization of broadband sounds. This "Dominant Region" occurs at ~700 Hz and also coincides with best ITD discrimination sensitivity. Above this frequency range, ITD sensitivity worsens, until ~1400 Hz where it rapidly becomes impossible. Explaining how listeners' ITD sensitivity transitions from best sensitivity to impossible to detect within ~1 octave challenges models of ITD processing. One possible explanation is that listeners continue to use binaural information from the Dominant Region even for higher-frequency pure tones. Within this explanation, the steep decline in ITD sensitivity reflects the high-frequency limb of peripheral frequency tuning for neurons within the Dominant Region (i.e., downward spread of excitation). To test this hypothesis, we measured the upper frequency limit of ITD processing as a function of stimulus level. Normal-hearing listeners were tested on a left-right ITD discrimination task. Stimuli were pure tones with 500–1500 Hz frequencies and 10–50 dB sensation levels. It was hypothesized that the upper frequency limit of ITD processing would decrease in frequency with decreasing level, which would indicate that the downward spread of excitation into the Dominant Region at least partially explains the rapid decline in ITD sensitivity.

4pPP30. A physiological study of forward masking: Recovery of discharge rate to probe tones in inferior colliculus. Swapna Agarwalla (Biomedical Eng. Dept., Univ. of Rochester, Rochester, NY 14642, swapwiz16@gmail.com) and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

In forward masking the perception of a target sound (probe) is degraded due to the presence of a preceding sound (masker). Two factors contributing to the probe response are (i) the temporal separation (delay) between the masker and probe and (ii) the interstimulus interval (ISI). Human listeners recover from forward masking to quiet probe thresholds for delays of 150–300 ms [Jesteadt *et al.*, *JASA* **71**, 950 (1982)]. Similarly, physiological studies have reported neural correlates of behavioral forward masking at the level of the inferior colliculus (IC), with 300-ms recovery of neural thresholds to the probe [Nelson *et al.*, *J. Neurosci.* **29**, 2553 (2019)]. Here, we estimated recovery of the discharge rate of IC neurons after forward maskers. We also quantified the impact of previous masker trials on IC rates by varying ISI. We recorded from the IC of awake rabbits using gaussian noise forward maskers and pure-tone probes. Interestingly, we observed significant effects of prior masking trials on discharge rates in response to the probe, even for ISIs larger than those typically used in psychophysical experiments. Our results show that ISIs of at least 1.5 s were required for full recovery of discharge rates after 70 dB SPL gaussian noise maskers. [NIH-NIDCD-010813.]

4pPP31. Speech recognition in the presence of speech maskers in children. Jithin R. Balan (Speech Lang. and Hearing Sci., Univ. of Texas at Austin, 300 W. Dean Keeton (A0900), Apt. 708, Austin, TX 78712-1069, jithin.balan@austin.utexas.edu), Qian-Jie Fu (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, CA), John J. Galvin (House Inst. Foundation, Los Angeles, CA), and Srikanta K. Mishra (Speech Lang. and Hearing Sci., Univ. of Texas at Austin, Austin, TX)

Listening to a target talker in the presence of other talkers follows a protracted developmental period. Such listening skills are critical for speech-language development and have clinical value. The purpose of the present study was to examine speech-in-speech recognition ability using digits in children. The major advantages of using digits are that it overcomes the biggest challenge of test administration, and digits are among the few first words children learn. Data were collected from 32 normal-hearing children (4–12 years) using full bandwidth speech materials. The test had four conditions like the Listening in Spatial Noise test. (1) Low cue: The target (male talker) and two male maskers were presented from the front. (2) Talker advantage: The target and two female maskers were presented from the front. (3) Spatial advantage: The target was presented from the front, and two male maskers were presented, one each from ±90°. (4) The target was presented from the front, and two female maskers were presented, one from ±90°. Results demonstrate developmental effects, good test-retest reliability, and feasibility of the computerized version of the test. Furthermore, results will be discussed in the context of various conditions and potential clinical applications.

4pPP32. Behavioral and neurophysiological signatures of the modulation filterbank in an animal model. Kenneth S. Henry (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave., Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu)

Fluctuations in the temporal envelope of sound, called amplitude modulation (AM), are an important information-bearing feature of many complex communication signals including speech. Human listeners show diminished sensitivity to AM signals in the presence of competing modulations of similar frequency, known as modulation masking. Modulation masking is not explainable by classic power-spectrum models, but instead suggests a "modulation filterbank" processing strategy that separates concurrent sounds (e.g., AM signals from noise) that have different modulation frequencies. The modulation filterbank is an exciting theoretical development because in addition to explaining modulation masking, the model predicts differences in speech perception across noise environments with different envelope statistics. However, the physiological underpinnings of the modulation filterbank remain uncertain due to limited nonhuman animal models. New behavioral and neurophysiological studies of the modulation filterbank are presented in budgerigars, a parakeet species with human-like behavioral sensitivity to many simple and complex sounds. Behavioral modulation-masking results show compelling evidence of the modulation filterbank in this animal model. Neural recordings from the inferior colliculus (midbrain level) further suggest that rate-based AM sensitivity can explain behavioral modulation-masking results. This new animal model of the modulation filterbank can be used to explore hearing mechanisms in real-world noisy listening environments.

4pPP33. The effects of preceding sound on three measures related to gain reduction. Elizabeth A. Strickland (SLHS, Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, estrick@purdue.edu), Madison McNeill, and Heesun Park (SLHS, Purdue Univ., West Lafayette, IN)

The medial olivocochlear reflex (MOCR) decreases the gain of the cochlear active process in response to sound. We have used psychoacoustic techniques to show behavioral effects of gain reduction, which could be consistent with the MOCR. We have used forward masking paradigms understood to measure frequency selectivity and the input/output function at the level of the cochlea using stimuli (masker and signal) that should be too short to evoke the MOCR. A precursor sound is then presented before these stimuli to evoke the MOCR. In previous studies, we have shown that a precursor: (1) increases signal threshold more after an off-frequency masker than the on-frequency one for forward maskers matched in effectiveness to increase signal threshold to 5 dB SL, (2) may increase on-frequency masker threshold required to mask a signal of 15-20 dB SL, and (3) may decrease suppression in forward masking. All of these effects would be consistent with a reduction in gain by the precursor, possibly by the MOCR. In the current study, these effects are all measured within the same listeners, using a more effective precursor than has been used in previous studies. The relationship between the measures will be discussed.

4pPP34. Measuring the efficiency of listening effort. Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall Rm 115, Minneapolis, MN 55455, mwinn@umn.edu)

Listening effort is a commonly reported difficulty among those who have hearing loss. This project aims to operationalize the concept of effort precision, as the ability to engage or disengage mental resources at strategic times could be an important signature of a person's capacity to guard against wasted effort. Stimuli were designed to simulate the situation of seeking

clarification of a misperceived word or attending to specific information in a sentence. Some stimuli included verbatim repetitions (which would ideally elicit reduced effort) and others featured sentences with a masked word followed by a clear repetition (which would ideally evoke effort aimed only at the newly unmasked word). Pupillometry was used as an index of moment-to-moment changes in listening effort. Data showed pupil dilations linked in time with the unmasked word, enabling precise measurements of the efficiency of effort. Listeners with normal hearing displayed the ability to plan and exert effort at specific times while also reducing effort in situations where it was unnecessary. Conversely, listeners who wear cochlear implants generally did not display these efficient effort characteristics. These results highlight the need to expand the concept of listening effort beyond a "more" or "less" framework, toward a framework of efficiency.

4pPP35. Spectral degradations in the TIMIT, QuickSIN, NU-6, and other popular bandlimited speech materials. Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, monson@illinois.edu) and Emily Buss (Univ. of North Carolina, Chapel Hill, Chapel Hill, NC)

Many popular speech corpora were recorded decades ago using transducers, recording techniques, or post-processing techniques that resulted in spectral degradations. The use of such materials very likely has consequences for measures of speech perception, but this is not often considered in studies that use these materials. We analyzed several of these popular corpora to ascertain spectral degradations that may be present. Spectral degradations we frequently observed included limited extended high-frequency (EHF) (>8 kHz) content, and, for a few corpora, substantial low-frequency roll-off <500 Hz. Two corpora analyzed here, the TIMIT/PRESTO and NU-6, exhibited both high- and low-frequency degradations. Such spectral degradations will likely have an effect on behavioral measures because they do not match the high-fidelity speech signals listeners encounter in their everyday lives. This should be considered when selecting stimuli for speech perception experiments. [Work supported by NIH Grant R01-DC019745]

4pPP36. Envelope-following responses to single and double amplitude modulation: No correlate of modulation masking. Magdalena Wojtczak (Psych., Univ. of Minnesota, N640 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, wojtc001@umn.edu), PuiYii Goh, and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Magneto- and electroencephalographic (M/EEG) measures of neural responses to speech and other natural sounds provide a noninvasive window into the neural tracking of temporal dynamics of natural sounds. However, the interpretation of these measures and their relation to perception remain unclear. In this study, amplitude modulation (AM) of a low-pass noise carrier (cutoff 4 kHz) was used to measure EEG envelope following responses (EFRs) for two AM rates presented simultaneously or in isolation. The spectral separations between the two AM rates were selected to produce varying degrees of perceptual modulation masking. The AM rates used spanned the range from 8 to 203 Hz, reflecting cortical and subcortical EFR generators. For double-rate AM, the EFRs had two distinct spectral peaks, corresponding to the component AM rates. In all conditions, the peak EFR amplitudes for two simultaneous AM rates did not differ significantly from those presented singly, even when the two rates produced significant perceptual masking. The results show that EEG measures of neural responses synchronized to temporal envelope fluctuations fail to reflect the important perceptual phenomenon of modulation masking. [Work supported by NIH R01 DC012262 (Oxenham) and R01 DC015987 (Wojtczak).]

4pPP37. Children's use of lexical cues for speech recognition in interleaved noise. Carla Youngdahl (Speech Lang. Pathol., Saint Mary's College, 45 Madeleva Hall, Notre Dame, IN 46556, cyoungdahl@saintmarys.edu)

This study investigates the role of lexical cues for speech recognition in noise across adults and children (5-10 years old). Target stimuli consists of phonetically balanced CVC and CVCCVC words and nonwords. Speech and speech-shaped noise (SSN) were separately divided into 30 contiguous 1-equivalent rectangular bands. Fifteen bands of speech were mixed with SSN bands to create four listening conditions: interleaved speech and noise, overlapped speech and noise, broadband noise and speech, and speech in quiet. To roughly equate performance between groups, speech and noise were presented at 0 dB SNR for adults and + 5 dB SNR for children. Anticipated results are that adults will achieve masking release when noise is interleaved with speech due to their ability to isolate frequency regions of clear speech. Children in this age range, however, may not have acquired this adult-like listening strategy. Analysis will be completed for recognition performance for words and nonwords at the whole word and consonant level. This analysis will help determine the extent that adults and children rely on lexical cues across noise conditions of varying spectral regions. Overall, these results may contribute to our understanding of how listening strategies for speech in noise develops in children.

4pPP38. Spatial attentional behavior analysis based on cognitive style during speech-in-noise task. Akira Takeuchi (Golisano College of Computing and Information Sci., Rochester Inst. of Technol., 1 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)

We designed a spatial selective attention task to investigate a listener's speech understanding ability in various masking noise conditions. The masking noise stream is music and non-intelligent speech combined sound which was played from the front-right and back directions. The target speech stream which has an auditory cue "Ready," following color and number information, for example, 'Ready Blue Five,' was played from the front-left direction. A listener's attentional ability was evaluated through the accuracy of the correct number and color selection in each condition of various SNRs and the spatial location of the masker. We formed two subject groups showing distinct cognitive styles, independent-analytic and interdependent-holistic, from a pre-experimental analysis. The analysis of variance (ANOVA) evinced a significant interaction between the group and SNR. Moreover, the independent-analytic group appeared to maintain their attention regardless of spatial location while another group scored poorly for the rear-positioned masker. Specifically, when the task got difficult with lower SNR conditions, this discrepancy between the two groups became significant. From these discussions, this study supports that individual differences associated with subconscious cognitive constructs influence speech-in-noise understanding ability.

4pPP39. Gamified spatial release from masking and relationships with self-reported auditory difficulties. Tess K. Koerner (NCRAR, VA Portland Medical Ctr., 3710 SW US Veterans Hosp. Rd., Portland, OR 97239, Tess.Koerner@va.gov), Karen Garcia, Lauren Charney, Vanja Pešić, Nicole Dean (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), Laura Coco (School of Speech, Lang., and Hearing Sci., San Diego State Univ., San Diego, CA), William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Aaron Seitz (Dept. of Psych., Univ. of California Riverside, Riverside, CA), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Many individuals with and without hearing loss, including those with a history of traumatic brain injury (TBI), often report difficulties understanding speech in noise that are not well predicted by common behavioral measures. Recent work in the gamification of behavioral auditory tasks has led to the development of measures that aim to better mimic the demands of real-world auditory environments through the use of video game features that engage and motivate the user. This study examined the effects of gamification on a spatial release from masking (SRM) task in 37 adults who ranged in age, hearing sensitivity, and history of TBI. Participants completed a gamified and non-gamified version of the SRM task on an iPad using the Portable Automatic Rapid Testing (PART) application. Regression models were used to assess whether gamification had an effect on SRM and whether participant age, hearing sensitivity, or history of TBI differentially impacted results. This study was also designed to determine whether performance on

the gamified SRM task was predictive of self-reported auditory difficulties that were measured using three hearing health questionnaires. Results of this work will have important implications for the development of clinically useful gamified tasks that better replicate real-world auditory environments. [R01DC018166, VA CDA1 RX003187]

4pPP40. Visual phonemic knowledge and audiovisual speech-in-noise perception in school-age children. Kaylah Lalonde (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68104, kaylah.lalonde@boystown.org) and Grace A. Dwyer (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Our mental representations of speech sounds include information about the visible articulatory gestures that accompany different speech sounds. We call this visual phonemic knowledge. This study examined development of school-age children's visual phonemic knowledge and their ability to use visual phonemic knowledge to supplement audiovisual speech processing. Sixty-two children (5–16 years) and 18 adults (19–35 years) completed auditory-only, visual-only, and audiovisual tests of consonant-vowel syllable repetition. Auditory-only and audiovisual conditions were presented in steady-state, speech-spectrum noise at individually set SNRs. Consonant confusions were analyzed to define visemes (clusters of phonemes that are visually confusable with one another but visually distinct from other phonemes) evident in adults' responses to visual-only consonants and to compute the proportion of errors in each participant and modality that were within adults' visemes. Children were less accurate than adults at visual-only consonant identification. However, children as young as 5 years of age demonstrated some visual phonemic knowledge. Comparison of error patterns across conditions indicated that children used visual phonemic knowledge during audiovisual speech-in-noise recognition. Details regarding the order of acquisition of viseme will be discussed.

4pPP41. Testing phenomenological auditory-nerve model predictions for selective inner- and outer-hair-cell dysfunction. Madhurima Patra (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, mpatra@purdue.edu), Andrew Sivaprakasam (Purdue Univ., West Lafayette, IN), David Axe (Mathworks, West Lafayette, IN), and Michael G. Heinz (Purdue Univ., West Lafayette, IN)

Sensorineural hearing loss (SNHL) effects on neural coding and perception have been largely associated with outer-hair-cell (OHC) dysfunction (e.g., reduced cochlear gain, reduced compression, broadened tuning). However, both inner-hair-cell (IHC) and OHC dysfunction occur in common hearing-loss etiologies, e.g., noise-induced and age-related (metabolic). Although IHC effects are largely ignored based on the insensitivity of audiograms to IHC loss up to 80%, the same selective IHC damage affects suprathreshold hearing (e.g., tone-in-noise detection, EFRs to periodic stimuli). Phenomenological auditory-nerve (AN) models are useful for studying neural-coding effects of SNHL, but their parametric control of OHC and IHC function (e.g., cohc and cihc parameters, Bruce *et al.*, 2018) are designed based on data from animals with noise-induced hearing loss (OHC/IHC dysfunction combined). Although this approach has proven successful for OHC effects, testing of IHC-based effects has been largely indirect. Here, we compare model predictions to our previously collected AN-fiber responses to amplitude-modulated stimuli in animals with either carboplatin-induced selective IHC dysfunction or gentamicin-induced selective OHC damage to provide more direct testing of IHC and OHC effects. Model parameters were estimated from histology, ABR thresholds, AN-fiber rates, and spike-train statistical metrics (e.g., vector strength, discriminability, mutual information).

4pPP42. A new psychophysical paradigm for measuring distortion products. Babak Ahadian (Cognit. Sci., Univ. of California Irvine, Irvine, CA 92697, bahadian@uci.edu) and Bruce Berg (Cognit. Sci., Univ. of California Irvine, Irvine, CA)

A unique contribution of this work is the demonstration that distortion product (i.e., Cubic difference tone (CDT)) effects can be systematically investigated with a discrimination task. Much of our knowledge about the perceptual effects of distortion products has been obtained with probe-tone, cancellation experiments, which are cumbersome and require highly trained, knowledgeable listeners. In this 2IFC discrimination study, the standard interval consists of two primaries, f_1 and f_2, having the same amplitude and phase. For the two experimental conditions, the signal interval consists of a change in the phase of f_2 to either $\pi/2$ or π. A third tone, f_3, was added to both intervals at the frequency of cubic distortion tone (CDT). The amplitude of f_3 is titrated with an adaptive threshold procedure. In order to acquire data relevant to the interaction between f_3 and the CDT, the phase of f_3 was varied across conditions. As predicted, the phase difference of π between the two primaries lowered the threshold by as much as 10 dB compared to a phase difference of $\pi/2$. This gain in sensitivity improves the precision of the methodology in estimating the phase and the amplitude of CDT in psychophysical tasks.

Session 4pSA

Structural Acoustics and Vibration, Education in Acoustics, Engineering Acoustics, and Physical Acoustics: Tutorial on Structural Acoustics and Vibration

Anthony L. Bonomo, Cochair
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Stephanie Konarski, Cochair
Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Road, Laurel, MD 20723

Chair's Introduction—1:30

Invited Paper

1:35

4pSA1. Tutorial on vibration and sound. Stephen Hambric (Hambric Acoust., LLC, 118 Westwood Rd., Asheville, NC 28804, hambricacoustics@gmail.com)

Over the years I have written many tutorials on vibroacoustics. This tutorial summarizes much of the material in those articles, starting with structural flexural vibrations and how they radiate sound. I explain structural modes of vibration, how they sum into structural mobilities, and how to use infinite plate theory to estimate the means of those mobilities. I also show that as structural damping increases, the mobilities of finite structures converge to that of an infinite structure. Next, you will see how structural flexural waves radiate sound into adjacent acoustic spaces. The radiated sound power may be normalized to a nondimensional radiation efficiency. Next we will examine fluid loading effects on structures, including loading from exterior and internal acoustic regions. I will show how exterior heavy fluid loading (from water) significantly mass loads and radiation damps structural vibrations. I will also show how internal air cavities can stiffen low order structural modes, and in some cases cause coupled oscillator behavior when structural and cavity mode resonance frequencies coincide. Finally, I will explore the fundamentals of sound transmission loss through structural barriers. My complete tutorial series on vibroacoustics is available at hambricacoustics.com.

Session 4pSC

Speech Communication: Speech Perception II - Bilingualism and Second Language Acquisition (Poster Session)

Shawn N. Cummings, Chair
University of Connecticut, Storrs, CT 06269

All posters will be on display from 2:00 p.m. to 5:00 p.m. Authors of odd-numbered abstracts will be at their posters from 2:00 p.m. to 3:30 p.m. and authors of even-numbered abstracts will be at their posters from 3:30 p.m. to 5:00 p.m.

Contributed Papers

4pSC1. A comparison of perceptual training efficacy and trajectories in young and old adults. Poulsen Birgitte (English, Aarhus Univ., J. C. Skous Vej 3, Aarhus DK-8000, Denmark, poulsen@cc.au.dk), Ocke-Schwen Bohn (English, Aarhus Univ., Aarhus, Denmark), and Christoph Draxler (Phonet. and Speech Processing, Ludwig-Maximilians-Universität München, Munich, Germany)

Current speech learning models propose that the ability for the reorganization of phonetic systems remains intact over the entire life span. This claim is well supported for adults up to the age of ca. 40 years, but empirical evidence on phonetic flexibility in mature learners is very scarce. We report on the first of a series of studies which examine seniors' (age 60 plus) phonetic learning ability. Two age groups of native Danish participants (young: age 18–35, and old: age 60–75) took part in ten sessions which trained the perception of the English initial /s-z/ contrast, which is absent from Danish (with only /s/.) The training groups, as well as age-matched control groups, also took part in pre- and post-tests examining their perception and production of English initial and final /s/ and /z/. The results address questions concerning age differences in phonetic learning and in learning trajectories, as well as questions concerning allophone-specific versus phoneme-general learning, and the nature of the production-perception link in phonetic learning. [Research supported by a grant from the Independent Research Aco-Fund Denmark, grant DFF2 0132-00008B.]

4pSC2. Crosslinguistic differences in cue primacy affect bilinguals' discrimination performance. Yuhyeon Seo (Linguist., Purdue Univ., 2501 Soldiers Home Rd., Apt. 5E, West Lafayette, IN 47906-1752, seo86@purdue.edu) and Olga Dmitrieva (Linguist., Purdue Univ., West Lafayette, IN)

Onset f0 – pitch at the onset of the vowel following the consonant – is an important cue to laryngeal stop distinctions in Korean, especially the lenis-aspirated contrast, while in English onset f0 plays only a secondary role in distinguishing voiced and voiceless stops. The current study investigates to what extent Korean heritage speakers ($n = 29$) who are English dominant can perceptually discriminate Korean lenis and aspirated stops differing acoustically only in terms of onset f0, but not in terms of VOT. Heritage speakers' performance is compared to that of Korean-immersed native speakers as a control group ($n = 29$). We hypothesized that heritage speakers would experience more difficulty than Korean-immersed speakers in using onset f0 as the sole cue to laryngeal contrast, due to the influence from English, where VOT dominates onset f0 as a cue to voicing. The results of a mixed-effects logistic regression model confirmed that heritage speakers were less accurate in the lenis-aspirated discrimination task than Korean-immersed speakers, providing support to the hypothesis that bilinguals' perception of their first language speech can be subject to crosslinguistic influence from their second language.

4pSC3. The effect of prosodic structure on cue integration in pitch perception. May Pik Yu Chan (Dept. of Linguist., Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu) and Jianjing Kuang (Linguist., Univ. of Pennsylvania, Philadelphia, PA)

Listeners integrate voice quality cues when perceiving pitch, and listeners' pitch perception strategy is modulated by their musicality. Prior work has shown that speakers of English, Mandarin and Cantonese performed similarly on stimuli based on English prosody (pitch contour being driven by stress), but it remains unknown whether the prosodic structure of the stimuli affects pitch perception. 47 L1-English and 95 L1-Cantonese speakers participated in a pitch classification experiment. The stimuli included two F0 peaks modelled after Cantonese prosody (pitch movement being mostly modulated by tones), they were resynthesized to contrasting in two spectral slopes creating four permutations. The first peak had a consistent F0, while the second peak varied in F0 along an 11-step continuum. Listeners judged whether the second 'ma4-ma1-ma4' was higher in F0 than the first word. Listeners also completed a musicality test. Results show that all participants integrated spectral cues in pitch perception similarly with breathier-tenser stimuli favoring hearing a higher pitch and vice versa, and individuals with high musicality integrate spectral cues less. However, the effect of musicality was significantly smaller for Cantonese than for English speakers, differing from stimuli modelled after English prosody. Overall results suggest an effect of prosody on speakers' pitch perception.

4pSC4. Young Pashtun L1 speakers' laryngeal contrast and voicing variation in Korean word-initial stops production. Soowon Yeom (Korean Lang. and Lit., Yonsei Univ., School, Seodaemun-gu State, Seoul 03722, South Korea, bulanyeom@gmail.com)

While age proved to have a critical influence on learning L2 phonology, the reason behind it is yet controversial. (brain maturation, LOR, quality l2 input, crib or sequential bilingualism, etc.) This paper examines 6 young people in the same household using Pashtun as their L1. As the elder children arrived in Korea at ages 10, 8, 6, and 3 respectively, the younger two were born in Korea now aged 10 and 8. All family members have equal LOR(10 years except for the youngest), equal L1, and equal L1 and L2 use extent in the home. Examining the differences between VOT and f0 production in these young people will clarify some of the explanations behind the age effects. Moreover, the data reveals the unique L1 transfer phenomena. While Pashtun is a true voicing language, Korean is an aspirating language where all 3- way contrast stops have positive VOT values. This can lead the young people to produce word-initial Korean stops in intersonorant positions with a marked voicing-ratio and to produce allophones that neither exists in their L1 nor in L2 such as voiced aspirated stop sounds (bʱ, d̪ʱ, gʱ). This paper will also examine whether these young people's VOT and f0 productions would align with the current tonogenesis like sound change in

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Korean. This paper used picture naming tasks with easy words instead of reading non-words to not test their grapheme-phoneme correspondences. Each person produces 9 phonemes of word-initial Korean stops (3-way laryngeal contrasts in 3 POA) both in the position in isolation and carrier sentence.

4pSC5. Influences of talker variability, reading ability, and language ability on word learning. Sandy Abu El Adas (Basque Ctr. on Cognition, Brain and Lang., Easo Kalea 71-2D, San-Sebastián, Gipuzkoa, Spain, sandy.abuadas@gmail.com), Ivy Yen, and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Studies show that talker variability is beneficial for learning second language contrasts, though less is known about variability in processing speech in a speaker's native language. Talker variability benefits have also been found for word learning studies with children, with better performance for children in the high variability condition than those in the low variability condition. Furthermore, it has been shown that individuals with a range of reading and language abilities perform differently on tasks that tap into phonological processes. Therefore, the current study examines how individual differences in reading and language abilities influence word learning and the extent to which talker variability improves performance. One hundred and forty listeners were randomly assigned to three conditions: (1) single talker, (2) multiple talker (randomized), and (3) multiple talker (blocked). Our results showed that individuals with poorer reading or language skills performed worse than individuals with higher reading and language skills. Surprisingly, variability was not beneficial and instead, listeners in the single talker condition performed better than those in the multiple talker conditions. These results suggest that talker variability benefits in word learning may be dependent on other aspects such as the amount of difference between talkers or the task.

4pSC6. Prosodic variation in intelligible L2-accented speech: Using acoustic measures to predict listening effort. Ruth Altmiller (Psychol. and Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., Campus Box 1125, St. Louis, MO 63130, ruth.altmiller@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Despite the relative ease with which listeners understand spoken language in their daily lives, speech perception is a complex task that becomes challenging in certain conditions, such as listening to speech in an unfamiliar second language (L2) accent. In the current study, we use pupillometry data from McLaughlin and Van Engen (2020) to ask whether prosodic variation predicts listening effort for L2-accented speech. In that study, L1 speakers listened to L1- and L2-accented English sentences. The pupillary response was larger for the L2-accented speech, even though it was fully intelligible. We hypothesize that prosodic differences in L2- versus L1-accented speech may account for such differences in listening effort for fully intelligible speech. To test this, we apply four acoustic measures to each of the stimuli used in the experiment. The first two measures, wiggliness and spaciousness (Wehrle *et al*., 2018), quantify the number and degree of F0 contour changes across each trial. The remaining two measures are relative word duration (Baker *et al*., 2009) and vowel pitch (Li & Shuai, 2011). We will use these trial level measures as fixed effects in Growth Curve Models to test the extent to which prosodic deviation predicts pupillary responses.

4pSC7. The effects of social manipulation of talkers on heritage bilingual speech perception. Maria F. Gavino (Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, mariagavino@u.northwestern.edu) and Matthew Goldrick (Linguist., Northwestern Univ., Evanston, IL)

The social information listeners are given about a talker affects their perception. For example, when speech is embedded in noise, L1 English listeners are more likely to accurately perceive a talker when they are told the talker is a native versus non-native speaker, even though it is the same acoustic signal across the two conditions. Among other factors, this effect has been attributed to listeners (inappropriately) relying on prior experiences rather than the acoustic information, and/or listeners devoting less attention to speech when listening to stigmatized accents. The current study examines whether this social information similarly impacts English-dominant Spanish heritage bilinguals' perceptual processes. 112 Mexican-American Spanish heritage bilinguals transcribed speech in noise in English-only and Spanish-only blocks. In each block, there were two talkers of different genders associated with native or non-native social information. Results show that listeners perform significantly better in their dominant language, however, did not reliably perform better when told a talker was a native speaker of the language they were perceiving. This suggests heritage bilingual listeners adopt a different approach to heritage bilingual speakers, relying more on the acoustic signal and/or better attending to speech (reflecting a lack of stigmatization of L2 speech or talker-listener alignment).

4pSC8. Accent rating of noise- and tone-vocoded foreign-accented speech. Jaskirat Sidhu (UW-Milwaukee, 2400 E Hartford Ave. 8th Fl., Milwaukee, WI 53211, jksidhu72@gmail.com), Li Xu (CSD, Ohio Univ., Athens, OH), and Jing Yang (UW-Milwaukee, Milwaukee, WI)

There are abundant acoustic-phonetic cues in speech signals for listeners to encode talker identity. However, speech signals in the real world are always less optimal due to various adverse listening sources. Vocoded speech is one type of simplified signal that has less spectral and/or temporal information in comparison to normal speech. The purpose of this study is to examine whether and how listeners' judgment of talker accent is affected by noise and tone vocoding. Twelve Mandarin-accented English speakers with varying degree of accentedness and two native English speakers were recorded reading "The Rainbow Passage." The recorded speech samples from each talker were segmented into small sections that were randomly selected for noise- or tone-excited vocoder processing into 1, 2, 4, 8, and 16 channels. The vocoded and unprocessed speech samples were randomly presented to a group of normal-hearing, monolingual English listeners for accent rating. The listeners judged the degree of talker accent on a 9-point Likert scale with "1" representing no accent and "9" representing extremely strong accent. The data are still in the process of being collected and analyzed. Results and implications of the present study will be discussed.

4pSC9. Asymmetries in vowel perception align with focalization patterns, not peripherality: Acoustic evidence from Canadian French /e/ and /Ø/. Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., SCSD, Montreal, QC H3A 1 G1, Canada, linda.polka@mcgill.ca), Matthew Masapollo (Univ. of Florida, Gainesville, FL), and Lucie Menard (Linguistique, Universite du QC a Montreal, Montréal, QC, Canada)

The question of whether asymmetries in vowel perception stem from peripherality (location in F1/F2 vowel space) or focalization patterns (formant convergence) was examined. Prior studies with Danish and Dutch listeners reported asymmetric perception of /e-Ø/ such that discriminating a change from /e/ to /Ø/ was easier compared to the reverse direction (/Ø/ to /e/). Acoustic analyses of test stimuli suggest that focalization patterns predict this asymmetry whereas a peripherality-based prediction fails. Although this interpretation is compelling, it is based on analyses of several vowel tokens produced by a single talker. To confirm the reliability of these patterns we obtained acoustic and electromagnetic articulography recordings of Canadian French /e/ and /Ø/ productions from a broader sample of talkers ($n = 20$): 400 vowels/10 productions per vowel. Acoustic analyses revealed a robust pattern for all talkers: F1 and F2 (also F1 and F3) were spectrally closer for /Ø/ compared to /e/, and /e/ was also more peripheral than /Ø/ in F1/F2 space. These data align with the earlier (single-talker) analyses and confirm that /Ø/ is a more focal vowel than /e/. We will also discuss articulatory analyses of these French vowel productions that focus on how vocal-tract constrictions relate to focalization and peripherality.

4pSC10. Statistical regularities of accented speech influence listeners' own speech productions. Timothy K. Murphy (Psych., Carnegie Mellon Univ., Baker Hall, 3rd Fl., Frew St., Pittsburgh, PA 15213, tkmurphy@andrew.cmu.edu), Nazbanou Nozari, and Lori Holt (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Conversation partners' speech acoustics sometimes converge, providing evidence for the transfer of information from speech perception to one's own speech productions. The nature of this phonetic convergence has remained elusive, with variable experimental findings regarding the contexts under which phonetic convergence emerges, the acoustic speech features it affects, and the gender of talkers most influenced. Here, we approach phonetic convergence through the lens of *dimension-based statistical learning* whereby the statistical regularities of short-term speech input impact the perceptual weight of acoustic dimensions in speech categorization. Participants passively listened to a randomly ordered sequence of 4 *pier* and 4 *beer* utterances, sampled across voice onset time (VOT) and fundamental frequency (F0) in a manner that aligned with English norms (Canonical) or departed from typical English pronunciations as an 'accent' (Reverse). Immediately after, participants categorized and repeated aloud an ambiguous test stimulus varying only in F0, with ambiguous VOT. The Reverse input regularities led both male and female participants to down-weight F0 in both perceptual categorization and in the acoustics of word repetitions. The results indicate that statistical learning across passive listening to another speaker's voice can lead to detailed acoustic-phonetic adjustments in one's own speech productions.

4pSC11. Age differentially impacts monolingual and bilingual listeners' understanding of English speech in noise. Ian Phillips (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., America Bldg., Rm. 5400, 4954 North Palmer Rd., Bethesda, MD 20889, ian.phillips7.ctr@health.mil), Rebecca E. Bieber (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., College Park, MD), Gregory M. Ellis (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Evanston, IL), and Douglas S. Brungart (Audiol. and Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Some bilinguals may perform worse understanding speech in noise (SIN) in their second language (L2) compared to monolinguals. Poorer performance has been found mostly for late bilinguals (L2 acquired after childhood) listening to sentences containing linguistic context, and less so for simultaneous/early bilinguals (L2 acquired during childhood) and when testing context-free stimuli. However, most studies tested younger participants–little is known about interactions with age. This study addresses this gap by measuring context-free SIN understanding via the Modified Rhyme Test in over 2,000 normal-hearing young and middle-aged bilingual and monolingual adults (ages 18–58; 23% bilinguals, all L2 English). Data collection is ongoing. Interim analyses reveal an interaction of age and group. Word recognition accuracy decreased as age increased for simultaneous and early bilinguals, but was stable for monolinguals and late bilinguals (though worse for bilinguals than monolinguals). Response time was faster for monolinguals but all groups slowed with increasing age at similar rates. These findings suggest an exaggerated age effect for bilingual SIN understanding across early and middle adulthood. [The views expressed in this abstract are those of the author(s) and do not necessarily reflect the official policy of the Department of Defense or the U.S. Government.]

4pSC12. Acoustic characteristics of English/n, l, t, k/ produced by native speakers of American English and Japanese at and near the word boundary. Takeshi Nozawa (Ctr. for Lang. Education and Res., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu, Shiga 5258577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Ratree Wayland (Linguist., Univ. of Florida, Gainesville, FL)

The acoustic characteristics of English /n, l, t, k/ produced by native speakers of American English and Japanese at and near the word boundary are compared. The phonemes are produced in two-word sequences in sentence contexts, /n/ or /l/ appears at the end of the first word (e.g., an earth, you'll earn) or at the beginning of the second word (e.g., a nurse, you learn), while /t/ or /k/ appears at the beginning of the second word (e.g., stops talking, ice cream) or as the second segment of the initial cluster /sk/ of the second word (e.g., stop stalking, I screamed). Acoustic analysis revealed that English speakers' /n/ in coda position is shorter and creakier, while Japanese speakers' /n/ is as long as in onset. In addition, English speakers' vowels in /Vl#/ context are more velarized than vowels in /V#l/ context, whereas Japanese speakers' vowels in /Vl#/ context are less /l/-colored. English speakers produced initial /t/ and /k/ with much longer VOT than in /st/, /sk/ conditions, but because Japanese voiceless stops are weakly aspirated, Japanese speakers /t/ and /k/ not significantly different in VOT between the two contexts, leading potentially to syllable misparsing by English listeners

4pSC13. Multilingual spoken word recognition: A megastudy approach. Benjamin V. Tucker (Commun. Sci. & Disord., Northern Arizona University, 4-32 Assiniboia Hall, University of AB, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Scott J. Perry, and Annika Nijveld (Linguist., Univ. of AB, Edmonton, AB, Canada)

Bilingualism research has primarily focused on the perception and processing of individual sounds or word learning. Many studies have investigated how bilingual listeners perceive sound contrasts that don't create lexical distinctions in their native language. There is substantially less research that has investigated how word-level properties impact L2 auditory processing. The present study examines how auditory lexical processing differs between monolingual and bilingual listeners with different language backgrounds. We collected lexical decision accuracies and latencies for 26,793 words and 9600 pseudowords from the Massive Auditory Lexical Decision database from native and non-native listener responses. We compare the language backgrounds of our 1028 listeners and group them into four groups: native speakers, early, early-late, and late bilinguals. We report the findings of an analysis investigating how language background and proficiency modulate lexical effects to understand how language background influences spoken word recognition. We find differences in the effects of lexical frequency, phonological neighborhood density, and phonological uniqueness point between the different listener groups. We discuss the impact of bilingualism in a word recognition task and how these results inform models of spoken word recognition and second language acquisition.

4pSC14. The cognitive demands of adverse listening conditions for monolingual and bilingual listeners: A pupillometry study. Sita Carraturo (Psychol. & Brain Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Saint Louis, MO 63130, sita@wustl.edu), Samantha Chen (Philos.-Neurosci.-Psych., Washington Univ. in St. Louis, St. Louis, MO), and Kristin J. Van Engen (Psychol. & Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Bilinguals typically underperform relative to monolinguals in speech-perception-in-noise tasks. However, there is little data comparing bilinguals and monolinguals in other types of difficult listening conditions. Furthermore, such differences are typically investigated using off-line intelligibility accuracy scores, which do not necessarily reflect differences in on-line speech processing. In the current study, pupillometry was used during sentence recognition tasks to index cognitive processing load. Monolingual English listeners and English-dominant simultaneous bilingual listeners heard English sentences in four conditions: American-Accented in Quiet, Turkish-Accented in Quiet, American-Accented in Noise, and Face Mask-Attenuated American-Accented in Quiet. Differences in pupil dilation between groups are expected in the latter three conditions. Data collection is on-going, but preliminary results ($N = 40$) show bilinguals exert more cognitive effort than monolinguals across all conditions.

4pSC15. Effects of auditory context on nonnative recognition of reduced speech: Does meaning explain it all? Bihua Chen (Indiana Univ., Ballantine Hall 704, 1020 E Kirkwood Ave., Bloomington, IN 47401, bc1@iu.edu) and Isabelle Darcy (Indiana Univ., Bloomington, IN)

Previous studies have shown that acoustic cues in the sentential context improve native (L1) listeners' recognition of reduced speech (Janse & Ernestus, 2011). That is, when hearing a reduced target in casual speech (e.g., *too into* /t^{w}ntə/), L1 listeners would be more accurate in recognizing it

in its surrounding speech (e.g., *she's too into computers*) than if they merely see the text of its context (e.g., *she's ___ computers*), even though semantic/syntactic information is equivalent in both conditions. It is unclear, however, whether nonnative (L2) listeners make use of the acoustic cues in the context to facilitate reduced speech recognition to the same extent as L1 listeners do. This study investigates this question by comparing L1 and L2 listeners of English in their recognition of reduced words that were aurally presented in three conditions—Isolation, Textual (with surrounding text given), and Auditory (with both text and auditory forms of context)—in a repetition task. Results showed that L2 listeners were significantly more accurate in the Auditory than Textual condition, and such difference was not distinguishable from L1 listeners. These findings lend support to the hypothesis that the facilitative effects of context acoustic cues are language general.

4pSC16. The visual analogue scaling (VAS) task as a pre-instructional assessment in tailoring the L2 phonetic training paradigm. Jieun Lee (Linguist., Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, jieunlee@ku.edu) and Hanyong Park (Linguist., Univ. of Wisconsin-Milwaukee, Milwaukee, WI)

Our previous research has shown that we can predict which learners will benefit more from phonetic training to learn non-native phonological contrasts using the Visual Analogue Scaling (VAS). In this study, we adopted the VAS as a pre-instructional assessment to identify less likely-to-be successful learners and provided a pre-training to see whether such preparations would benefit those learners. We targeted naive English learners of Korean learning the Korean lenis-aspirated stop contrast who had to learn to use f0 cues (secondary cue in English stop voicing contrast) as a primary cue for successful learning. Using the VAS task with English stimuli, we identified participants with low f0 sensitivity. We gave the cue-attention switching training (CAST) to enhance their f0 sensitivity before providing three phonetic training sessions for the target contrast. Results showed the positive effects of CAST. Participants with low f0 sensitivity demonstrated a similar level of achievement in learning as participants with high f0 sensitivity and showed more native-like f0 cue utilization in identifying the target contrast than low f0 sensitivity participants who received training without CAST. Our study highlights the importance of considering individual differences with appropriate assessments and tailoring the training paradigm accordingly to maximize possible learning outcomes.

4pSC17. Phonetic learning and vocabulary knowledge in adolescents. Alissa Hopper (Communicative Disord. and Sci., Univ. at Buffalo, 122 Cary Hall, University at Buffalo, Buffalo, NY 14214, alissaho@buffalo.edu) and Christopher C. Heffner (Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY)

Individuals who perform well on tasks that measure their abilities to learn non-native speech sounds tend to also perform well on language measures such as vocabulary tests. This relationship is not well understood but may be driven by increased literacy skills among individuals with stronger phonetic learning skills. However, relatively few studies have examined the changes in performance that occur during adolescence, a time in which critical vocabulary knowledge is amassed. This is an especially interesting developmental period because studies comparing adults and children show that adults often outperform children on measures of non-native speech sound learning. In the present study, we examine the changes in non-native speech sound learning abilities that occur during adolescence. Our preliminary data confirm previous findings which show that non-native learning performance tends to increase with age. We also explore vocabulary development as a correlate of this effect. Our preliminary data suggest that the correlation between non-native phonetic learning and vocabulary grows stronger with age during adolescence. These findings support the hypothesis that phonetic learning abilities continue to develop during adolescence and that this development is relevant to language outcomes in the adolescent population.

4pSC18. The acoustics of the alignment of focus prosody and lexical stress: A production study of native speakers and second language learners of English. Hyunah Baek (Gwangju Inst. of Sci. and Technol., 123 Cheomdangwagi-ro, GIST College A, Rm. 420, Buk-gu, Gwangju 61005, Korea (the Republic of), hyunahbaek88@gmail.com)

In English, the tonal target of a pitch accent is aligned with lexical stress. Therefore, the phonetic realization of contrastive focus prosody, phonologically marked by a H*/L+H* pitch accent, is expected to vary depending on the stress pattern of the focused word. This study explored acoustic effects of the alignment of lexical stress and focus prosody and examined how it is learned by Korean learners of English, whose first language lacks lexical stress. Ten native English speakers (L1) and 19 Korean learners of English (L2) participated in a read-aloud task, in which they produced nine trisyllabic words with primary stress on one of the three syllables with contrastive focus (e.g., *Animal, vaNILla, engiNEER*). For each syllable, three acoustic measures were extracted: the height of pitch peak, periodic energy mass, and the direction of pitch movements (Albert *et al.*, 2021; Cangemi *et al.*, 2019). Results revealed that the height of pitch peak, a conventional measure of both lexical stress and contrastive focus, did not properly demonstrate their alignment. Instead, the stressed syllable of the focused words was marked by its overall prominence (mass) or the direction of pitch movements, depending on the stress pattern. The L2 speakers' productions were similar to the L1 productions, but there were indications of an influence of their L1 prosody and English proficiency. [This work was supported by the Ministry of Education and the National Research Foundation of Korea (NRF-2022S1A5A8052482).]

4pSC19. L1 category precision hypothesis in L2 production: Korean learners' English front vowels. Sujin Oh (Linguist., Univ. of Wisconsin-Milwaukee, 2522 E Hartford Ave. Milwaukee, WI 53211, sujinoh@uwm.edu) and Hanyong Park (Linguist., Univ. of Wisconsin-Milwaukee, Milwaukee, WI)

It has been hypothesized that learners with more precisely defined categories in their native language (L1) are better at learning nonnative sounds in a second language (L2). Some perception studies have supported this hypothesis, but not many production studies have been conducted. Our study aims to fill this gap. Focusing on front vowels in F1/F2 acoustic space, we calculated compactness scores of L1 vowels and distances between learners' and native speakers' productions to operationalize category precision and L2 accuracy, respectively. Then, we examined whether learners with a lower compactness score would produce L2 vowels with a closer distance. The learners' productions were taken from a corpus containing native Korean speakers' reading and retelling of the North Wind and the Sun in Korean and English and comparable native English speakers' productions from ALLSSTAR corpus. Results showed that learners with lower compactness scores produced more accurate English /i/ and /ɪ/, but not for /ɛ/ or /æ/, which partially supports the hypothesis's application to L2 production. We speculate that the merger between /e/ and /æ/ in Korean, the Northern Cities Vowel Shift in English, and/or perceptual mapping between native and L2 categories are responsible for the English non-high vowel results.

4pSC20. The neurophysiological responses of music processing in bilingual teenagers. Sharry Guo (Commun. Sci. & Disord., St. John's Univ., Queens, NY), Andreas Diaz (Neurosci., St. John's Univ., Queens, NY), Valerie Shafer (Ph.D. Program in Speech-Language-Hearing Sci., The Graduate Ctr., City Univ. of New York, New York, NY), Gavin Bidelman (Speech, Lang. and Hearing Sci., Indiana Univ., Memphis, TN), Tyreek Jackson (St. John's Univ., Queens, NY), and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

The association/dissociation between music and language processing has long been a matter of debate. Musicians and tonal language speakers are more sensitive to pitch differences than nonmusicians and nontonal language speakers. Bilingual experience modulates auditory processing of sounds, but it is unclear whether and how bilingual experience affects music

processing. This study examines music processing in bilingual teenagers from Mandarin (a tonal language) households and those from non-tonal language households. The central question is whether bilingual experience enhances auditory processing similarly regardless of the specific language, or whether the influence of language on music processing is language specific. Event-related potential (ERP) responses were recorded from 65 scalp sites. An oddball paradigm with six types of music changes (intensity, pitch, rhythm, timbre, slide and location) was presented. Preliminary results did not support the general bilingual advantage theory, but did support the language-specific enhancement of auditory processing.

4pSC21. Thing or sing: Modeling the phonetic variation of the interdental fricative /θ/ in accented English. Fenqi Wang (Linguist., Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, fenqi@ufl.edu), Delin Deng, and Ratree Wayland (Linguist., Univ. of Florida, Gainesville, FL)

Chinese learners of English tend to pronounce the English interdental fricative /θ/ as [s] because it is absent in Chinese (Rau *et al.*, 2009). To investigate the phonetic variation of the interdental fricative /θ/, this study extracted the first four spectral moments of the target fricatives [θ] and [s] from fricative-initial words in an L2 English speech corpus, and then analyzed them using generalized additive mixed-effects models (GAMM) and random forest classification. The corpus we used contains 31.6 hours of recordings from 389 Chinese speakers from different regions in China (Chen *et al.*, 2019). Classification accuracy rates suggest variation in the production of /θ/ among Chinese learners of English from different Chinese dialectal regions, and the computation of conditional feature importance sheds light on the relative impact of acoustic measurements in predicting the fricatives in accented English. The findings of this study can provide robust empirical evidence on the phonetic variation of the interdental fricative /θ/ in accented English by Chinese learners of English and shed light on a new way to probe into the phonetic variation in non-native speech.

4pSC22. Easy come, easy go: Examining the stability of lexically guided perceptual learning over time. Emma C. Hodges (Univ. of Connecticut, Storrs, CT), Shawn N. Cummings (Univ. of Connecticut, 2 Alethia Dr., Unit 1085, Storrs, CT06269-1085, shawn.cummings@uconn.edu), and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Listeners can use lexical information to accommodate ambiguity in speech input. Some evidence suggest that lexically guided perceptual learning persists over time. However, other evidence suggests that lexically guided perceptual learning attenuates throughout the test session, consistent with distributional learning that occurs given exposure to the test stimuli. Here we test the hypothesis that lexically guided and distributional learning may operate over different time scales. During exposure, listeners heard spectral energy ambiguous between /ʃ/ and /s/ in a lexically-biasing context. At test, listeners categorized tokens from an *ashi-asi* continuum. Test duration was manipulated between subjects to be either brief or long. Approximately 24 hours later, both groups completed a second test phase. The results to date show (1) robust lexically guided perceptual learning at the first test, (2) attenuation of learning for the long duration compared to the short duration test group at the first test, and (3) no robust evidence of lexically guided perceptual learning for either group at the second test. If these results hold in the full sample to be tested, then they would suggest that lexically guided perceptual learning is best characterized as dynamic adaption to recent input that dissipates within a day.

4pSC23. Linking lexically guided perceptual learning to statistical patterns in speech input. Shawn N. Cummings (Univ. of Connecticut, 2 Alethia Dr., Unit 1085, Storrs, CT06269-1085, shawn.cummings@uconn.edu), Jeung-Yoon Choi, Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA), and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Listeners use lexical information to modify the mapping between speech acoustics and speech sound categories. Despite convention to consider lexically guided perceptual learning as a binary outcome, the magnitude of the learning effect varies in the extant literature. We hypothesize that graded learning outcomes can be linked, in part, to statistical characteristics of the to-be-learned input, consistent with the ideal adapter theory of speech adaptation. Following standard methods (i.e., waveform averaging to create ambiguous variants), a lexically guided perceptual learning stimulus set for the /ʃ/-/s/ contrast was created for each of 16 talkers, yielding variability in the statistical cues specifying this contrast across talkers. Experiment 1 will (a) measure lexically guided perceptual learning for each talker, (b) identify input characteristics that are associated with learning magnitude, and (c) examine whether a computational instantiation of the ideal adapter theory can model the input-learning link. Experiment 2 will provide a confirmatory test of the patterns observed in Experiment 1 by manipulating the to-be-learned input holding talker constant. The results will provide a critical test of the ideal adapter framework for speech adaptation, thus informing an understanding of the mechanisms that allow listeners to solve the lack of invariance problem for speech perception.

4pSC24. Interpretation of speech rhythm: Speech error, speech rhythm, and speech proficiency. Seongjin Park (ETS AI Labs, Educational Testing Service, 660 Rosedale Rd., Princeton, NJ 08540, spark002@ets.org)

It has been well-known that second language learners are affected by their first language when producing their L2. For speech rhythm, it has been suggested that L2 speakers are affected by L1 speech rhythm (e.g., Korean learners of English produce English without reducing the duration of unstressed vowels), and the effect is greater when speakers are beginner or intermediate-level language learners. This study, however, suggests that the direction of the effect is not always the same as researchers expected, and shows how easily speech rhythm is influenced by speech errors. The result of this study shows the relationship between the type of speech errors and speech rhythm metrics, and how that affects the perceptual proficiency of L2 speakers as well as L1 speakers. Future studies will be conducted to examine the way to infer the type of speech errors using speech rhythm metrics.

4pSC25. Accommodating accents: Linking rapid adaptation and continuous learning over a month. Xin Xie (Univ. of California Irvine, SSPB 2223, University of California, Irvine, Irvine, CA 92617, xxie14@uci.edu) and Chigusa Kurumada (Univ. of Rochester, Rochester, NY)

Perceptual difficulties associated with unfamiliar talkers or accents are known to dissipate—sometimes within minutes of exposure. Two major questions remain open: (a) *How does adaptation commence within the initial moments of an encounter?*; and (b) *How does the adaptation continue via repeated exposure to the same talker/accent?* The current study addressed these questions by assessing incremental changes of native English listeners' recognition of L2-accented speech (Mandarin-accented English) over two timescales: within the first few minutes and over a month. Specifically, we tested how native listeners' recognition of a word-final /d/, which is initially confusable with a /t/, may change over these two time scales. The target versus control groups both heard Mandarin-accented speech, but only the target group heard words containing the critical contrast. Using a new repeated exposure-test paradigm, trajectories of adaptation were assessed three times within an initial session (Exp.1) as well as over five sessions spanning a month (Exp.2). In addition to mixed-effect analysis on the effect of exposure at a group level (i.e., target versus control), computational simulations of distinct mechanisms (cue-based normalization, Bayesian belief updating, changes of decision-making criteria) will examine whether adaptation seen across the timescales can stem from the same mechanism.

4pSC26. Acoustic convergence of voicing when shadowing an unfamiliar language and a non-native accent. Amy Hutchinson (Dept. of Linguist., Boston Univ., 621 Commonwealth Ave., Boston, MA 02215, ahut@bu.edu), Alexis N. Zhou, Yuhyeon Seo, and Olga Dmitrieva (Linguist., Purdue Univ., West Lafayette, IN)

Previous work has established that speakers are able adjust the acoustic properties of their speech in order to mimic their interlocuter or speech model with whom they share a native language (e.g., Pardo, 2013). However, it is less clear whether convergence to accented speech and/or unfamiliar languages is also a possibility. The goal of the present study is to investigate acoustic convergence between two groups of native English speakers (N = 15 in each group) and a speaker of Russian-accented English

versus a speaker of Russian. The study consisted of four experimental phases: baseline, exposure, shadowing, and post-test. Participants in both groups shadowed acoustically identical material, although one group was informed that they were hearing Russian-accented English words (ACC-ENG condition), while the second group was told that they were shadowing Russian words (RUS condition). Initial analysis of the VOT durations of participants' voiceless stops indicated that only participants in the RUS group converged towards the models' shorter VOT during shadowing, although this convergence did not generalize to participants' pronunciation of English words during the post-test. Additional data analysis is ongoing. This presentation will also discuss results with respect to their implications for theories of speech accommodation and non-native speech learning.

4pSC27. Exploring bilingual effects on iconic memory using 2D and 3D visuospatial recognition tasks. Cinty Chang Wu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, 1539 West 3rd St., Brooklyn, NY 11204, cwcinty@gmail.com) and Laura Spinu (Speech Commun., Kingsborough Community College, Brooklyn, NY)

Numerous studies indicate positive effects of bilingualism on cognition, mostly found in the domains such as attention, inhibitory control, auditory processing, and memory (Bialystok, 2018; Krizman *et al*., 2012; Spinu, 2022). While most studies focusing on memory have explored differences between monolinguals and bilinguals in working memory and, more recently, in auditory sensory memory, the question of whether bilingualism also affects iconic memory remains understudied. In this study, we explore whether bilingual experience results in enhanced iconic memory in the form of gestural, visuospatial recognition. We present bilingual and monolingual participants from NYC ($n = 40$) with thirty sets of foreign characters, one at a time, then ask them to recall which characters they have seen before from a list of characters that includes novel ones bearing high resemblance with previously shown characters. Second, we display to the same participants thirty sets of ASL signs (15 2D images and 15 3D short videos), with high resemblance as well, utilizing the same method to evaluate their gestural and visuospatial abilities in iconic memory. While the experiment is currently underway, the findings will shed more light on whether the benefits associated with additional language experience extend to non-linguistic domains, likely resulting from enhanced cognitive function mechanisms.

THURSDAY AFTERNOON, 11 MAY 2023 MICHIGAN/MICHIGAN STATE, 1:30 P.M. TO 3:50 P.M.

Session 4pUW

Underwater Acoustics, Acoustical Oceanography, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics: Acoustic Methods for Unmanned Mobile Platforms: Sensing, Localization, and Communications II

Fumin Zhang, Cochair
Electrical and Computer Engineering, Georgia Insitute of Technology, 85 Fifth Street NW, Atlanta, GA 30332

Aijun Song, Cochair
Electrical and Computer Engineering, University of Alabama, 245 7th Ave., Tuscaloosa, AL 35487

Martin Siderius, Cochair
Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201

Invited Papers

1:30

4pUW1. An experimental study of mobile underwater acoustic channels and their impacts on mobile acoustic communications. Zhiqiang Liu (US Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375, zhiqiang@ieee.org)

Most underwater acoustic communication schemes have implicitly assumed limited platform mobility and their performance tends to suffer considerably when the platform is in high motion. To gain insight into the effects of high platform mobility, a sea-going experiment was recently conducted. In the experiment, two channels are set up. One is a stationary channel between a fixed source and a fixed receiver. The other is a highly mobile channel between a fixed receiver and a source towed in a circle at a speed of 6 knots. The two channels are directly compared in terms of four key channel parameters changes that matter most to the performance of acoustic communications, namely, channel coherence, path loss, Doppler scaling factor and carrier frequency offsets. A channel probing signal is judiciously designed to enable reliable estimation of those parameters at high resolution in time. This paper will provide the details of this sea experiment, including its design, setups and execution, and report interesting findings.

1:50

4pUW2. Optimizing wireless acoustic energy harvesting and communication with passive markers underwater. Ananya Bhardwaj (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr NW, J. Erskine Love Bldg., Rm. 131, Atlanta, GA 30332, ab22@gatech.edu), Ahmed Allam (MIT Media Lab, Massachusetts Inst. of Technol., Cambridge, MA), Alper Erturk, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

To enhance functionality of Acoustic Identification Tags (AID), passive markers designed as landmarks for Autonomous Underwater Vehicles (AUVs) [Satish *et al.*, *JASA* **149** (2021)], the development of an accompanying battery free active communication platform was explored [Bhardwaj *et al.*, *JASA* **152**, 4 (2022)]. This platform utilizes custom piezoelectric transducers impedance matched for broadband operation, enabling concurrent acoustic energy harvesting and fast backscatter communication [Allam *et al.*, *SPIE* **31**, 9 (2022)]. The combination of these active and passive devices can improve AUV localization and information transfer for specific applications such as short-range search missions or AUV homing and docking. The design of this platform for underwater operations requires a balance of parameters such as center frequency and piezoelectric element dimensions to achieve a high range of operation, power efficiency and data rates, while minimizing attenuation and spreading losses challenges. The design process for piezo selection, acoustic and electrical impedance matching, transducer casing design, and the experimental evaluation of the chosen 350 kHz design will be discussed. This approach was validated experimentally to quantify performance metrics such as the maximal range, data rates using various standard communication such as Frequency Shift Keying (FSK) and Amplitude Modulation (AM).

2:10

4pUW3. Multi-user transmit beamforming for broadband acoustic systems. Diego A. Cuji (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, cujidutan.d@northeastern.edu), Zhengnan Li, and Milica Stojanovic (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Multi-user transmit beamforming for acoustic systems based on orthogonal frequency division multiplexing (OFDM) modulation is studied and demonstrated. Transmit beamforming consists of assigning weight coefficients to multiple transmit elements, such that the signals transmitted from the array add constructively at the users' receive sides. In an optimal setting, the transmitter assigns weights that are proportional to the channel transfer functions to the users. However, the assumption of complete channel knowledge is not entirely accurate, as the transmitter obtains the channels via feedback from the users. Thus, the transmitter has partial knowledge of the channels, as their estimates are not only noisy, but can be outdated if the channels have changed during the time it took to close the feedback loop. These adversities are notably pronounced in time-varying acoustic channels, as the feedback delay is large as compared to the channel coherence time due to the low speed of sound propagation. To counteract this problem, a technique that targets only those features of the channels that can withstand the feedback delay is proposed. Such features are the users' angles of arrival of the principal propagation paths. The principal paths are stable paths that do not experience rapid fluctuations and are thus varying sufficiently slowly that they can tolerate long feedback delays. The design concepts are demonstrated using experimental over-the-air transmissions in an indoor environment of an acoustic communications testbed.

2:30

4pUW4. Exploring propagation delay and mobility in underwater acoustic networks. Ruoyu Wang (Elec. and Comput. Eng., Lehigh Univ., 400 Packard Lab, 19 Memorial Dr. West, Bethlehem, PA 18015, ruw220@lehigh.edu) and Yahong R. Zheng (Elec. and Comput. Eng., Lehigh Univ., Bethlehem, PA)

This work investigates the impact of mobility on underwater acoustic communication networks where the propagation delay is comparable to or larger than the packet duration. A new simulator is proposed to implement the acoustic communication timing model when the nodes are mobile or static. Synchronous and asynchronous Medium Access Control (MAC) protocols are employed with ALOHA, TDMA (Time-Division Multiple Access), and Artificial Intelligent (AI) agent nodes. Extensive simulations are conducted to study the throughput of a MAC network where a large number of source nodes transmit to a single sink node. A network of 2–15 source nodes are assumed to be randomly located in a 1500 m cubic space and some of the nodes are moving in random directions at 2–4 m/s speed. The simulation results show that the asynchronous MAC protocols achieve much higher throughput than the synchronous protocols by allowing time slots to be much shorter than the maximum propagation delay among nodes and allowing asynchronous transmission time. High mobility of a few mobile nodes also favors asynchronous protocols and improves the network throughput in a manner similar to an AI agent.

Contributed Papers

2:50

4pUW5. Passive underwater Acoustic IDentification (AID) tags for enhancing Autonomous Underwater Vehicle (AUV) navigation during docking or homing operations. Nizar Somaan (Mech. Eng., Georgia Inst. of Technol., Rm. 0306-C, 930 Spring St. NW, Atlanta, GA 30309, nsomaan3@gatech.edu), Ananya Bhardwaj, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Autonomous underwater vehicle (AUV) navigation requires accurate positioning information from the surrounding environment, especially for tasks such as AUV homing or docking operations. Previous literature has introduced a class of low-cost passive underwater acoustic markers, termed Acoustic IDentification (AID) tags [Satish and Sabra, *J. Acoust. Soc. Am.* **149**(5), 3387–3405 (2021)] built of multi-layer shells with different acoustic properties and thicknesses to generate a uniquely engineered acoustic signature, composed of the multiple reflections created by the layer interfaces. These passive AID tags can be detected by an AUV instrumented with a high-frequency sonar transducer at significantly greater distances than conventional optical methods, especially in turbid waters. Additionally using AID tags as navigation-aid can also alleviate the need of relying on active acoustic transponders. An implementation of a constellation of AID tags enabling fine underwater positioning an AUV towards a docking station or for homing purposes will be presented to provide proof of concept. Furthermore, the optimization of the design of the AID tags for this application as well as specific signal processing detection methodologies to improve the detectability of AID tags in the presence of interfering signals (e.g., clutter) will be discussed. [Work supported by the Office of Naval Research].

3:05

4pUW6. Low signal-to-noise ratio localization exploiting through-the-sensor *in situ* environmental information. Franklin H. Akins (Scripps, UCSD, 562 Arenas St., La Jolla, CA 92037, fakins@ucsd.edu) and William Kuperman (Scripps, UCSD, La Jolla, CA)

Drifting sensors/arrays typically may not have detailed information of the environment. A combination of modal-MUSIC [*JASA Exp. Lett.* **2**(7), 074802 (2022)] and synthetic aperture matched field processing [*J. Acoust. Soc. Am.* **150**(1), 270–289 (2021)] provides a method to localize a weak source in a shallow water environment without a priori knowledge of the bottom geoacoustic parameters. The success of the method hinges on its ability to extract the bottom geoacoustic information from noise. The technique is relevant for low-frequency (<500 Hz) localization with a drifting vertical line array. Localization of a quiet source without a priori geoacoustic information is demonstrated on data from the SWellEx-96 experiment. [Work supported by the Office of Naval Research.]

3:20

4pUW7. Improving passive acoustic target detection using machine learning classifiers. Ananya Bhardwaj (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, J. Erskine Love Bldg., Rm. 131, Atlanta, GA 30332, ab22@gatech.edu), Nizar Somaan (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Tillson Galloway (College of Computing, Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The ocean covers 70% of the Earth surface area, yet over 80% of it remains unexplored despite the advances in underwater acoustics and oceanography. Ocean exploration is critical for accurate climate model development, renewable energy applications, and in understanding the marine habitat. Further exploration necessitates improvements in underwater navigation with Autonomous Underwater Vehicles (AUVs). Utilizing acoustic landmarks can enhance AUV localization performance [Fula *et al.*, Oceans IEEE/MTS (2018)]. Passive markers called Acoustic Identification (AID) Tags have unique and identifiable acoustic reflection signatures designed to function as landmarks [Satish *et al.*, *JASA* **149** (2021)]. These targets can be detected by Match Filtering returns with template signals. Match Filtering performance is limited in the presence of strong interfering signals, and with changes in sound speeds that alters the temporal structure of these signatures. The application of a Machine Learning classifiers for detecting AID tag signatures can improve the localization performance. Through implementation of Logistic Regression, Deep Neural Networks, and Convolutional Neural Networks, the generalizability and superiority of these models is demonstrated. We report high accuracies (>90 %) in a multilabel (8 classes) classification task with signals with low SNR (–6 dB) and strong interference (+12 dB).

3:35

4pUW8. Lake testbed for mobile acoustic communications and networking. Aijun Song (Elec. and Comput. Eng., Univ. of Alabama, 245 7th Ave., Tuscaloosa, AL 35487, song@eng.ua.edu) and Fumin Zhang (Elec. and Comput. Eng., Georgia Insitute of Technol., Atlanta, GA)

We present a lake testbed that can support mobile acoustic communications and networking research. The primary assets of the testbed are two autonomous underwater vehicles equipped with software-defined acoustic modems. The acoustic modems are integrated with the user-installed autonomy software on the vehicle, MOOS-IvP. Through this integration, vehicle information can be passed to the acoustic modem. Or the acoustic modem can receive remote instructions to modify vehicle missions. With additional stationary nodes, the testbed can support two types of acoustic experiments. First, it supports acoustic waveform transmissions from two synchronized mobile nodes. The acoustic transmissions can be recorded in a multi-element receiving array for offline data processing. Second, the testbed can perform real-time acoustic communication and networking tests for different vehicle trajectories. We will present the testbed instrument and capabilities, field deployments, sample acoustic data, and communication test results. [The research is supported by the national science foundation (NSF).]

Session 5aAA

Architectural Acoustics: Heavy-Handed Recommendations: When Less Is More

Brandon Cudequest, Cochair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604

Shane J. Kanter, Cochair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604

Contributed Paper

8:00

5aAA1. Do we really need to do this? Jesse Ehnert (Arpeggio, 1947 Aspen Dr., NE, Atlanta, GA 30345, jehnert@arpeggiollc.com)

In the acoustical consulting profession, our task is usually to provide recommendations based on our expertise and experience on projects involving existing problems or new facilities looking to avoid problems and achieve sonic success. This does not occur in a vacuum as practicalities and budget usually loom in the shadows (or out in the open). Inevitably, clients often question and sometimes even push back on our guidance. This presentation will discuss certain areas where these situations are prone to arise, with examples, and will endeavor to explore the nuances of the question: "do we really need to do this?"

Invited Papers

8:15

5aAA2. A rigorous optimization of noise control with natural ventilation in an office with high-end sustainability goals. Arthur W. van der Harten (Acoust. Distinctions/Open Res. in Acoust. Sci. and Education, 400 Main St. Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com), Stefano Capra (Ramboll, Copenhagen, Denmark), and Irene Gallou (Specialist Modelling Group, Foster + Partners, London, United Kingdom)

The Bloomberg London headquarters, designed by Foster + Partners and built by Sir Robert McAlpine, espouses some of the world's loftiest sustainability goals for a large office building (more than 4000 employees). It is said to have achieved a 98.5% score against BREEAM sustainability assessment methodand was the winner of the 2018 RIBA Stirling Prize for Architecture. This talk will cover the design and implementation of the natural ventilation systemand discuss the delicate multi-disciplinary collaboration that was required in order to operate the natural ventilation effectively while still meeting requirements for background noise in an open office space in the heart of London.

8:30

5aAA3. Subjective success of exterior glazing with both a light and heavy hand. Joseph W. Myers (Kirkegaard, Chicago, IL) and Michelle Huey (Kirkegaard, 110 W. Ohio St. Ste. 3W, Chicago, IL 60654, mhuey@kirkegaard.com)

The subjective success of the isolation of exterior noise depends greatly on the nature of the site noise, the use of the isolated space, and the client's expectations. The architectural solution must respond to all of these factors, as well as budget and aesthetics. In recent years, Kirkegaard has completed several projects in which glazed walls separate a performance or rehearsal space from an urban or suburban exterior. Solutions have ranged from simple to "heavy-handed," with a varying degree of subjective success. This paper offers a range of case studies, including situations where a significant retreat from "heavy-handed" solutions was subjectively successful and one situation where a modest retreat from "heavy-handed" solutions was not subjectively successful.

8:45

5aAA4. Challenging rules of thumb to redefine flexibility. 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 Nicolaus T. Dulworth (Threshold Acoust., Chicago, IL)

The recently completed Lindemann Performing Arts Center on the Brown University campus is an exploration in all things flexible. To satisfy the programmatic needs of Brown Arts Initiative, the primary user of this new building, who needed five rooms but only got one, the Main Hall redefined the concept of multiuse hall. All sixsurfaces that define the major acoustically supportive surfaces (ceiling elements, walls, and floors) move to manipulate the otherwise beautifully simple architectural concept into five room configurations—Orchestra, Recital, End Stage Theatre, Experimental Media, and large Flat Floor. A mix of manual and motorized curtains and banners adds still more flexibility. As the paint still dries on the building, this paper will investigate the acoustic challenges, happy accidents, and areas where we might have done with less (or more) in a building that is sometimes heavy handed and sometimes a light touch. The paper will cover topics such as glass as a major reflecting surface, wall buildups that break rules of thumb to produce warm acoustic responses, use of variable acoustic solutions, and ensemble to audience size ratios that challenge conventional wisdom.

9:00

5aAA5. The sound of sustainable structures: Acoustic considerations for mass timber. Ben Loshin (Acoust., Audiovisual, and Theatre, Arup, 1191 Second Ave., Ste. 400, Seattle, WA 98101, ben.loshin@arup.com) and Denis Blount (Acoust., Audiovisual, and Theatre, Arup, Seattle, WA)

The use of mass timber in multi-family and commercial buildings poses a range of challenges for acoustic designers, due in part to its recent emergence as a structural technology and relative light weight compared to concrete. The spectral performance of mass timber assemblies differs from that of concrete and of lightweight double-panel systems, such as timber joist floors, even when comparing assemblies with the same single-number airborne and impact sound transmission ratings. Due to the inclusion of mass toppings and damping layers, mass timber floor designs may outperform lightweight double-panel systems at low frequencies, but they are often challenged at mid to high frequencies, resulting in a sound spectrum that differs from established "acoustically acceptable" benchmarks. This study presents initial findings from listening sessions conducted in the Arup SoundLab, as well as various design strategies to balance performance, cost, aesthetics, and sustainability.

Contributed Paper

9:15

5aAA6. Acoustic design trade-offs when reducing the carbon footprint of buildings. Jonathan Broyles (Architectural Eng., The Penn State Univ., 510 Toftrees Ave., Apt # 331, State College, PA 16803, jmb1134@psu.edu)

Architects and building engineers are increasingly tasked to consider the environmental performance of building designs to meet national and global targets to reduce carbon emissions. While minimizing the carbon footprint is needed to mitigate the effects of climate change, sustainability-driven design can have benefits or unintended consequences on secondary design objectives. Neglect of additional building disciplines could negatively impact the indoor environmental quality in a building and require costly change orders or retrofits. Recent research has shown that sustainability-driven design decisions can affect the acoustical performance in a building, though sustainable and acoustical goals are often unrelated. In response, this presentation reviews acoustic trade-offs when the building design is driven by sustainable goals. Acoustical ramifications and design solutions when minimizing the operational and embodied carbon are discussed. Lastly, areas of future research at the intersection of architectural acoustics and sustainability are provided. Overall, this presentation gives a brief overview on reducing carbon emissions in buildings, discusses the consequences sustainability-driven design can have on building acoustics, and provides sustainable alternatives that do not hinder acoustic performance.

Invited Papers

9:30

5aAA7. Vibration isolation of newer equipment typologies. Jay Bliefnick (Acentech, 33 Moulton St., Cambridge, MA 02138, jbliefnick@gmail.com), Mark Harlan, and Ben Davenny (Acentech, Cambridge, MA)

The vibration isolation of large mechanical/electrical/pluming equipment has become an industry standard to prevent unwanted vibrations from transmitting into building structures. ASHRAE, ISO, and other organizations have published recommended guidelines for vibration isolation based on the typical operation of this MEP equipment and "best practices" over the past several decades. However, recent advancements in mechanical equipment designs have resulted in situations where the traditional approach for vibration isolation is either not feasible or ineffective, such as air handling units with extremely well balanced fan arrays or low-speed cooling towers with variable-frequency drives. Changes in power distribution system designs have also resulted in a greater need for vibration isolation of electrical equipment, particularly utility transformers. This paper reviews these recent developments in MEP systems as well as the implications to vibration isolation for this equipment.

9:45–10:00 Break

10:00

5aAA8. Sound transparent assemblies in concert halls: Using simulation to balance acoustics and design aesthetic. 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 David Kahn (Acoust. Distinctions, Stamford, CT)

Variable acoustics finishes are often incorporated into concert halls to allow adjustment to the liveness of the space. Therefore. the visual appearance of the hall changes, depending on the settings of those variable acoustics finishes. Architects often prefer a consistent visual appearance, regardless of the positioning of those variable acoustics finishes. Acoustic Distinctions has made an effort to determine the relationship between visual opacity and sound transparency in order to facilitate a more successful collaboration between architect and acoustician without any compromise or guesswork in the acoustical impact of these finishes. This paper discusses several concert halls that incorporate sound transparent surfaces to hide variable acoustics finishes behind. We cite previous work done by Acoustic Distinctions and the University of Hartford to test and verify sound transparent construction and introduce more recent work using the Finite Volume Method to determine acceptability of sound transparent construction, and to inform application to larger geometrical acoustics models.

10:15

5aAA9. Auralization as a value management tool. Joshua Brophy (Acentech, 33 Moulton St., Cambridge, MA 02138, jbrophy@acentech.com), Kelsey Rogers, and Khaleela Zaman (Acentech, Cambridge, MA)

Acoustical consultants tend to rely on written reports, verbal explanations, and drawing markups to convey the scope and meaning of our recommendations. However, since hearing is an inherently perceptive phenomenon, these modes of communication are indirect: they require users and clients to imagine the implications of important design decisions rather than experience them directly. Auralizations can help us fill in those gaps by allowing users to "get a sense" of an acoustical environment before it is built, literally hear how design decisions influence the acoustical outcome, and feel more confident moving forward with the best approach for their project. Most often, these experiences convince decision-makers to devote more resources to the acoustic design than they may have done otherwise, occasionally even going above and beyond the "best practice" recommendation. However, sometimes the simulated outcome of the light touch approach is deemed acceptable to the client, and in these instances, the auralization enables them to avoid the expense of overdesigning. This presentation will explore the spectrum from "client invests more than we expected" to "client reduces costs based on auralization" with a number of case studies related to client decisions that follow the auralization experience.

10:30

5aAA10. Acoustical design prototyping with photo-enabled sound intensity. Erik Miller-Klein (Tenor Eng. Group, 11514 Dayton Ave. N, Seattle, WA 98133, erik.mk@tenor-eng.com)

This session will explore options to prototype and mock-up acoustical systems, assemblies, and materials to help optimize and right-size the design for your performance needs. Through years of use with a photo-enabled sound intensity wand, some of the commonly held beliefs about the root causes of acoustical performance defects have been shown to be over-designed. Project examples and observations from case studies are discussed in-depth to highlight some of these key conditions and ways for the industry to provide more efficient design solutions.

10:45

5aAA11. Assuming an engineering mindset in acoustical design: Helping clients get the most for their money. Paul H. Scarbrough (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, pscarbrough@akustiks.com) and Christopher Blair (Akustiks, LLC, Norwalk, CT)

Acoustical design for performance spaces is generally considered to be both an art and a science. Part of the science involves assuming an engineering mindset when deciding where to draw the line with respect to criteria for room acoustics, sound isolation, HVAC noise control, and other issues. When does sound from the outside truly become intrusive? How quiet is quiet enough when it comes to HVAC noise? Renovation projects often impose practical constraints that preclude aiming for extreme degrees of sound isolation or HVAC noise control. Even well-funded new construction projects sometimes require careful consideration of how to expend project resources to best effect. This paper will explore the authors' experiences making such decisions on a range of projects involving new construction (Las Vegas' Smith Center for the Performing Arts), historic restoration (Cincinnati Music Hall), adaptive re-use (Milwaukee's Bradley Symphony Center), and substantial reconstruction (David Geffen Hall at Lincoln Center).

11:00

5aAA12. Avoiding heavy-handed recommendations. Ronald Eligator (Eligator Acoust. Assoc., 825 Eighth Ave., 18th Fl., New York, NY 10019, religator@eligatoracoustics.com)

The avoidance of heavy-handed construction recommendations for achieving acoustic performance is important to ensure project success. Keys to achieving this goal include proper alignment of client expectations regarding acoustics, project schedule, and budgetas well as practice of an interdisciplinary approach to project design and construction. This presentation will provide several examples of how educating clients, design team members, and the constructorson the importance of integrated design and construction, and following through with appropriate construction materials and methods, can result in better projects at lower cost—and avoid heavy handed recommendations and construction.

Session 5aBAa

Biomedical Acoustics: General Topics in Biomedical Acoustics: QUS and Beamforming

Cameron Hoerig, Cochair
Radiology, Weill Cornell Medicine, 416 E 55th St., MR-007, New York, NY 10022

Federico Mento, Cochair
Department of Information and Communication Engineering, University of Trento, Via Sommarive, 9, Povo, Trento 38123, Italy

Contributed Papers

7:30

5aBAa1. Monitoring radiofrequency ablation in *ex vivo* human liver using 3D echo decorrelation imaging augmented by deep learning. Elmira Ghahramani Z. (Biomedical Eng., Univ. of Cincinnati, UC Bioscience Ctr., 3159 Eden Ave., ML 0048-04D, Cincinnati, OH 45221, ghahraea@mail.uc.edu), Peter D. Grimm (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Jiang Wang (Pathol., Univ. of Cincinnati, Cincinnati, OH), Syed A. Ahmad, Shimul A. Shah, R. Cutler Quillin, Sameer H. Patel (Surgery, Univ. of Cincinnati, Cincinnati, OH), Marepalli B. Rao (Environ. Health, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Monitoring of radiofrequency ablation (RFA) is desirable to improve safety and efficacy of liver tumor treatment. Three-dimensional ultrasound echo decorrelation imaging can successfully predict local ablation effects but has had limited success in mapping ablation zone margins and local tissue temperature. Here, a supervised deep learning approach is investigated to improve prediction of ablation zones and tissue temperature from 3D echo decorrelation images. RFA was performed on *ex vivo* human liver tissue, including normal, fibrotic, and cirrhotic liver ($N > 30$). During ablation, pairs of echo volumes were acquired with a 4.5 MHz transesophageal matrix array, 3D echo decorrelation images were computed for each volume pair, and temperatures measured by fourthermocouples integrated into the RFA probe were recorded. Tissue was then frozen, sectioned, scanned, and ablation zones were manually segmented. For prediction of ablation zones, B-mode and echo decorrelation images were input to a U-net convolutional neural network to segment ablation margins, with histology serving as ground truth for training and cross-validation. For prediction of temperature, echo decorrelation values at the thermocouple locations were used as input to train a dense network, minimizing mean-squared-error versus measured temperatures. The results indicate promise for improved mapping of tissue ablation and temperature.

7:45

5aBAa2. A preliminary numerical investigation of convolutional neural network (CNN) techniques for filtering high-intensity focused ultrasound (HIFU) noise in images. Grace Farbin (Univ. of Toronto, 38 Spotted Owl Crescent, Brampton, ON L7A0J9, Canada, gracefarbin@gmail.com)

High-intensity focused ultrasound (HIFU) is a minimally invasive medical procedure that uses ultrasonic waves to ablate or heat tissue with the aim of treating tumors and tremors. Diagnostic ultrasound imaging is the primary mode of imaging during HIFU treatments due to its real-time capabilities. However, HIFU noise, produced from therapeutic ultrasound components, interfere with the diagnostic ultrasound components and cause difficulty in monitoring changes to tissue during treatment. In a multitude of HIFU treatments, deep learning has been used as a tool to detect coagulation, monitor temperature, and segment tumors. Convolutional neural network (CNN) models are a series of deep learning algorithms that can assign importance to aspects of an inputted image and differentiate one from the other. Based on previous methods of filtering, CNNs too can be trained to filter raw RF signals received by an ultrasound probe for subsequent real-time treatment feedback with HIFU. Here, we were able to present a preliminary investigation of a CNN approach for HIFU noise reduction. To do this, we used acoustic wave simulations from k-Wave, a time-domain, full-wave model for ultrasound wave propagation, in combination with the Deep Learning Toolbox from MATLAB. Subsequent analyses studied the performance of noise reduction via the proposed regression model.

8:00

5aBAa3. Machine-to-machine transfer function: Transferring deep learning models between ultrasound machines. Ufuk Soylu (Elec. Eng., The Univ. of Illinois Urbana-Champaign, 405 N Mathews Ave., Urbana, IL 61801, usoylu2@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

In our previous work, we proposed a calibration method for deep learning (DL) to mitigate the effects of acquisition-related data mismatches in the context of tissue characterization. We showed that the “setting” transfer function can transfer deep learning models between imaging settings. We now extend the calibration method to transfer deep learning models between ultrasound machines. This can lead to reduced cost of model development and also improved understanding of the issues related to the security of deep learning based algorithms in biomedical ultrasound imaging. We gathered four datasets from three different scanners: (i) a SonixOne Machine with an L9-4 transducer, (ii) a Verasonics Vantage 128 Ultrasound Machine with an L9-4 transducer using line by line acquisition, (iii) a Verasonics Vantage 128 Ultrasound Machine with an L9-4 transducer using plane wave compounding, and (iv) a Siemens Antares Ultrasound Machine with an VF10-5 transducer. We used the first dataset as training data and the other datasets as testing data. The DL algorithm learned how to classify two tissue mimicking phantoms. The classification accuracy jumped to 90% from 50% for the second dataset, 70% from 50% for the third dataset, and 61% from 56% for the fourth dataset after incorporating the calibration method. Therefore, the results confirm that a transfer function approach can be used to transfer learning models between scanners for the purpose of classifying materials based on ultrasonic backscatter.

8:15

5aBAa4. Assessing machine independence of quantitative ultrasonography to evaluate vision degrading myodesopsia. Cameron Hoerig (Radiology, Weill Cornell Medicine, 416 E 55th St., MR-007, New York, NY 10022, cah4016@med.cornell.edu), Justin Nguyen (VMR Consulting, Inc., Huntington Beach, CA), Jonathan Mamou (Radiology, Weill Cornell Medicine, New York, NY), Cedric Venuat (Quantel Medical, Cournon d'Auvergne, France), J. Sebag (VMR Consulting, Inc., Huntingon Beach, CA), and Jeffrey Ketterling (Radiology, Weill Cornell Medicine, New York, NY)

Vision degrading myodesopsia (VDM) results from clinically significant vitreous opacities that reduce contrast sensitivity (CS) and visual quality of life (VQOL). Previous studies with a single clinical system for quantitative ultrasonography (QUS) found correlations with CS and VQOL as measured by the N.E.I. Visual Function Questionnaire (VFQ). Here, we evaluated a cohort of patients to determine if VDM as evaluated by QUS is machine independent. 28 eyes from 14 patients (age 56 ± 14 years) experiencing VDM were scanned with two different clinical ophthalmic ultrasound systems in succession: one with a 15-MHz single element transducer (23mm focal length, 7 mm aperture) and one with a 20 MHz annular array (five elements, 22 mm focal length, 9 mm total aperture). Images were acquired as a longitudinal plane through premacular vitreous in temporal gaze. Three QUS parameters and a composite score were computed from the vitreous region within each set of log-compressed envelope data and averaged over artifact-free frames. A linear regression comparing QUS parameters computed from each system showed statistically significant correlations ($R \geq 0.86$, $p < 0.001$). Moreover, QUS parameters from both systems were correlated with CS and VFQ scores. The results from this study suggest the QUS parameters are independent of scanner and could be universally applied to assess VDM.

8:30

5aBAa5. Quantitative characterization of the ultrasound thermal strains in tissue mimicking phantoms with different oil concentrations. Zhiyu Sheng (Medicine, Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 624, Pittsburgh, PA 15213, zhs41@pitt.edu), Ran Wei (Univ. of Pittsburgh, Pittsburgh, PA), Mengyue Chen (North Carolina State Univ., Raleigh, NC), Bohua Zhang (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Howuk Kim (North Carolina State Univ., Raleigh, NC), Xuecang Geng (Blatek Industries, Inc., Boalsburg, PA), Xiaoning Jiang (NC State, Raleigh, NC), and Kang Kim (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Ultrasound thermal strain imaging (US-TSI) has been known for the capability of tissue characterization according to distinct sound speed change in different tissues when temperature increases. US-TSI for detecting lipids in atherosclerosis plaques and fatty livers has previously been reported while some practical challenges were not fully addressed, especially due to physiological motions. To overcome such limitation, we recently developed an ultrasound transducer that combines an acoustic heating array and an imaging array to achieve US-TSI with heating performed in a region of approximately 10 mm by 5 mm by 2 mm within a very short time period of about 50 ms compared both cardiac and breathing motions. To characterize the new US-TSI probe, a thorough benchtop investigation was performed on the relationship among the threekey variables for TSI: thermal strain, temperature increase, and lipid concentration. In the experiments, homogeneous oil-in-gelatin phantoms of different oil concentrations were fabricated to simulate different lipid-plaque concentrations. Temperature curves were recorded by a thermal couple with millisecond-level time constant. Thermal strains were computed by developed US-TSI signal processing procedures. The results build a tissue-temperature-strain model and calibrate the new US-TSI probe for *in vivo* atherosclerosis plaque characterization.

8:45

5aBAa6. Quantitative ultrasound for preterm birth risk prediction—Part 1: Statistical evaluation. Mehrdad Mohammadi (Univ. of Illinois Urbana-Champaign, 616 E Green St., Ste. 212, Champaign, IL 61820, mehrdad3@illinois.edu), Barbara L. McFarlin, Michelle Villegas-Downs (Univ. of Illinois Chicago, Chicago, IL), Aiguo Han (Univ. of Illinois Urbana-Champaign, Urbana, IL), Douglas G. Simpson (Univ. of Illinois Urbana-Champaign, Champaign, IL), and William D. O'Brien (Univ. of Illinois Urbana-Champaign, Urbana, IL)

Hypothesis: Predicting the spontaneous preterm birth (sPTB) risk level is enhanced when using both historical clinical (HC) data and quantitative ultrasound (QUS) data compared to using only HC data. HC data defined herein includebirth history prior to that of the current pregnancy as well as, from the current pregnancy, a clinical cervical length assessment, and physical examination data. Study population included 248 full-term births (FTBs) and 26 sPTBs. Advanced statistical analyses were performed for supervised classification containing 53 scaled candidate features (48 QUS, 5 HC) using nested fivefold cross-validation of L1-penalized linear logistic regression with 1000 repetitions to identify potential predictors. Statistical models for HC data alone and HC + QUS data were compared with likelihood-ratio test, cross-validated receiver operating characteristic (ROC) area under the curve (AUC), sensitivity, and specificity. To assess performance, the ROC-AUC was estimated with 10-fold cross-validation logistic regression and 1000 repetitions. Averaged ROC curves plus AUCs were computed using threshold averaging. AUC confidence intervals and test statistics to test the two ROC curves' differences were constructed via DeLong method. Combined HC and QUS data identified women at sPTB risk with better AUC (0.68; 95% CI, 0.57–0.78) than those of HC data alone (0.53; 95% CI, 0.40–0.66). [Work supported by NIHR01HD089935.]

9:00

5aBAa7. Quantitative ultrasound for preterm birth risk prediction—Part 2: Clinical consequences and impact. Barbara L. McFarlin (Dept. of Human Development Nursing Science, University of Illinois at Chicago, Chicago, IL 60612, bmcfar1@uic.edu), Mehrdad Mohammadi (Statistics - Electrical Eng., Univ. of Illinois Urbana-Champaign, Champaign, IL), Michelle Villegas-Downs (Human Development Nursing Sci., Univ. of Illinois Chicago, Chicago, IL), Aiguo Han, Douglas G. Simpson (Univ. of Illinois Urbana-Champaign, Champaign, IL), and William D. O'Brien (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Globally, ~15 million babies are born preterm every year. In the United States, the preterm birth rate remains stubbornly high (10–15%) and refractory to interventions. Consequences of preterm birth (PTB) account for the second leading cause of infant mortality, with 1 million deaths annually. While the costs of PTB to society are more than for any other disease, the impact on families is devastating. Preterm babies suffer both immediate and lifelong physiological, cognitive, and developmental health problems. It is estimated that with the proper tools and technology, we could reduce the preterm birth and survival rates. Due to lack of technology, clinicians have had few interventions that have been rigorously studied to prevent preterm birth. Ineffective interventions have historically been based on opinion and patient symptoms rather than tissue based reliable and repeatable scientific studies. Our group has rigorously studied quantitative ultrasound's (QUS's) role for assessing PTB risk in animals and humans. QUS provided added value to currently available health and traditional ultrasound risk assessment methods. Basing clinical decision-making on tissue microstructure has the potential to reduce the PTB rate and provides a scientific basis for developing and objectively evaluating present and new treatments to prevent PTB. [Work supported by NIHR01HD089935.]

9:15

5aBAa8. Evaluating postpartum cervical remodeling with quantitative ultrasound technology. Michelle Villegas-Downs (Dept. Human Development Nursing Sci., Univ. of Illinois at Chicago, 845 S. Damen Ave., Chicago, IL 60612, mville20@uic.edu), Barbara L. McFarlin (Dept. Human Development Nursing Sci., Univ. of Illinois at Chicago, Chicago, IL), William D. O'Brien (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Douglas G. Simpson (Statistics, Univ. of Illinois Urbana-Champaign, Champaign, IL), Tara Peters, Judith M. Schlaeger (Dept. Human Development Nursing Sci., Univ. of Illinois at Chicago, Chicago, IL), and Aiguo Han (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Having a history of a previous spontaneous preterm birth (sPTB) is the strongest risk factor for recurrent sPTB. It is unknown if there are differences in postpartum cervical remodeling between women who have delivered spontaneous preterm (sPT) and full-term. No studies have evaluated the role of postpartum remodeling between the two groups. Quantitative ultrasound (QUS) is a noninvasive ultrasound technology used to quantify tissue microstructure and function. QUS biomarkers were used to evaluate postpartum cervical microstructure in women who delivered sPT and full-term. Data were collected from 54 women:14 who delivered sPT and 40 who delivered full-term. Transvaginal QUS scans were performed at 6 weeks ($\pm$2 weeks) after delivery. Attenuation coefficient (AC), backscatter coefficient (BSC), and shear wave speed (SWS) QUS biomarkers were collected. BSC was significantly higher at six weeks postpartum in women who delivered sPT versus full-term ($p = 0.01$), while the AC approached statistical significance ($p = 0.09$). QUS biomarker BSC was able to identify cervical microstructure differences at six weeks postpartum between women who delivered sPT and full-term. QUS technology may improve our understanding of postpartum cervical remodeling and has the potential to noninvasively direct precision-health approaches for recurrent sPTB. [Work supported by NIH 5F31NR019716.]

9:30–9:45 Break

9:45

5aBAa9. Automated region of interest placement on cervical ultrasound images for assessing preterm birth risk. Jingyi Zuo (Dept. of Bioengineering, Univ. of Illinois Urbana-Champaign, 1406 W. Green St., Urbana, IL 61801, jzuo5@illinois.edu), Barbara L. McFarlin (Dept. of Human Development Nursing Sci., Univ. of Illinois Chicago, Chicago, IL), Douglas G. Simpson (Dept. of Statistics, Univ. of Illinois Urbana-Champaign, Champaign, IL), William D. O'Brien, and Aiguo Han (Bioacoustics Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Spontaneous preterm birth (sPTB) is one of the leading causes of infant morbidity. Medical interventions can prevent death caused by preterm birth if the risk is predicted at early stages. Quantitative ultrasound (QUS) is found valuable for predicting sPTB risk with a limitation of requiring a medically trained image analyst to manually draw a region of interest (ROI) on the cervix of a B-mode ultrasound image. An automated ROI placement algorithm was designed and trained to reduce the reliance on human annotations. The algorithm utilized a deep neural network with an optimized U-Net architecture to locate cervical tissues. A total of 8670 ultrasound images with sonographer-drawn ROIs were used for algorithm training and testing, followed by several postprocessing steps to yield the final ROIs. Quantitative comparison between algorithm-generated and sonographer-drawn ROIs yielded an average pixel accuracy of 96% and a dice coefficient of 88%. In addition, the QUS's attenuation coefficient (AC) and backscatter coefficient (BSC) obtained from the algorithm-generated ROIs were highly correlated to those obtained from the sonographer-drawn ROIs with a Pearson correlation coefficient of 0.93 and 0.85, respectively. The results support the feasibility of automating QUS imaging for sPTB risk assessment. [Work supported by NIHR01HD089935.]

10:00

5aBAa10. The tradeoffs between grating lobes and larger array pitch using null subtraction imaging. Mick Gardner (ECE, Univ. of Illinois at Urbana-Champaign, 405 N Matthews Ave., Urbana, IL 61801, mickhg2@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Null Subtraction Imaging (NSI) is a new beamforming technique for producing B-mode images that results in high spatial resolution and low computational cost compared to other high-spatial resolution imaging techniques. Previous work has demonstrated that in addition to a narrow main lobe and low side lobes, NSI can also reduce or mitigate grating lobes. Grating lobes can appear when the wavelength of ultrasound is larger than the pitch of the array and these grating lobes can result in imaging artifacts. Lower grating lobes on a larger pitch array could allow larger linear arrays for abdominal imaging with lower element count. Experiments were conducted to quantify the tradeoff between grating lobe levels and pitch size when using NSI. NSI was able to reduce grating lobes by up to 19dB with an element pitch of five times the wavelength. The effects of the DC offset (a tunable parameter of NSI) on grating lobe reduction were also testedand found that NSI with a lower DC offset provides greater reduction in grating lobes, with up to 40dB reduction. The level of grating lobe reduction versus the depth of a target was also testedand found a slight decrease in grating lobe reduction at greater depths.

10:15

5aBAa11. Decorrelated compounding methods in synthetic transmit aperture ultrasound imaging and its application. Yuan Xu (Dept. of Phys., Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B2K3, yxu@ryerson.ca) and Na Zhao (Toronto Metropolitan Univ., Toronto, ON, Canada)

Speckles exist in most ultrasound images. They are generated due to the interference of the ultrasound waves scattered from multiple scattering particles in one resolution cell. Although they are very useful in many ultrasound applications, they limit the detection of low-contrast targets in a scattering medium. In conventional compounding, multiple correlated sub-images from various imaging apertures and frequency bandwidths are generated and then averaged incoherently to reduce the speckles. In this paper, we present our study on a method called Decorrelated Compounding to reduce speckles. In decorrelated compounding, a decorrelation procedure was applied to the correlated sub-images to further reduce speckle variance in synthetic transmit aperture (STA) ultrasound imaging. Decorrelated compounding was shown to improve the detectability of low-contrast lesions in terms of lesion signal-to-noise ratio (lSNR), visual detection, and statistical tests of the performance in detecting low-contrast lesions. The application of the proposed method to monitoring ultrasound thermal therapy and to measuring the attenuation coefficient will also be discussed.

10:30

5aBAa12. Improved image uniformity using optimized diverging-wave acquisition sequence for high frame rate pulse-echo ultrasound. Kashta Dozier-Muhammad (Dept. of Biomedical Eng., The Univ. of Memphis, 3720 Alumni Ave., Memphis, TN 38152, kndzrmhm@memphis.edu) and Carl Herickhoff (Dept. of Biomedical Eng., The Univ. of Memphis, Memphis, TN)

Ultrafast pulse-echo ultrasound imaging uses unfocused plane-wave transmit (PWT) or diverging-wave transmit (DWT) wavefronts and coherent compounding for image reconstruction. PWT imaging is more commonly utilizedbut has a limited region of overlapping insonification. This work characterizes the tradeoffs between PWT and DWT, to determine an optimal DWT transmit scheme for given constraints on imaging field of view (depth and width), frame rate, and resolution uniformity. Using Field II, the transmit energy field of a 64-element, 2.75-MHz linear phased array (Verasonics P4-2v) was analyzed for PWT and various active apertures and relative

virtual source locations for DWT. This was followed by a Field II calculation and analysis of point-spread functions (PSFs) at many locations in the field (to depth z = 35 mm and x = 20 mm laterally) for each PWT and DWT case, and several cases of PWT and DWT compounding. The amplitude and resolution of the PSF, and the uniformity (variance) of each of these metrics over the field of view, weremeasured in each case. Preliminary results suggest that a DWT scheme provides improved PSF amplitude uniformity over a broader field of view than PWT, with only a slight reduction in lateral resolution at greater depths.

10:45

5aBAa13. Application of a 2D interpolation technique to reduce the number of steering angles required for high-quality ultrasound images in plane-wave compounding. Sajjad Afrakhteh (Information Eng. and Comput. Sci., Univ. of Trento, Povo Summarive, Trento, Trentino 38122, Italy, sajjad.afrakhteh@unitn.it), Hamed Jalilian (Mathematics, Iran Univ. of Sci. and Technol., Tehran, Iran (the Islamic Republic of)), Giovanni Iacca, and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

Plane wave imaging (PWI) can reach frame rates in the kHz range. However, PWI suffers from reduced image quality due to the lack of focusing. To tackle this limitation, plane wave compounding (PWC) was introduced. PWC is based on recombining multiple PWI images generated using different steering angles. However, the number of steering angles required to achieve reasonable image quality significantly reduces the frame rate. In this study, we propose a two-dimensional (2D) interpolation technique, based on radial basis functions, which allows reducing the number of steering angles in PWC without degrading image quality. The idea is to reduce the number of steering angles and then apply 2D interpolation, along the angle dimension and directly to the RF data, to reconstruct the data related to those angles for which data were not collected. The full dataset is then utilized to generate an image. To compare this technique with standard PWC, we utilized the dataset from the Plane-wave Imaging Challenge in Medical UltraSound (PICMUS). The results show that the number of steering angles can be reduced by factor 3, thus tripling the frame rate, while achieving comparable performance with standard PWC (with 75 angles) in terms of contrast and spatialresolution.

11:00

5aBAa14. Ultrafast ultrasound beamformer for plane wave imaging with field programmable gate array. Zhengchang Kou (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, B420 Beckman Inst., 405 North Mathews Ave., Urbana, IL 61801, zkou2@illinois.edu) and Michael L. Oelze (Elec. and Comput. Eng.,Univ. of Illinois at Urbana-Champaign, Urbana, IL)

In this work, we propose a novel method of implementing an ultrafast ultrasound beamformer for plane wave imaging (PWI) on a field programmable gate array (FPGA). First, a modified delay calculation method was proposed to (1) separate the transmit and receive delay, (2) reduce the size of delay profile, and (3) enable parallel beamforming by delay reuse and data vectorization. Second, a parallelized implementation of beamformer on single FPGA was proposed by (1) loading pre-calculated delay profile from external memory instead of calculating delay on run-time, (2) vectorizing channel data fetching, (3) compensating transmit and receive delays separately, and (4) using fixed summing networks to reduce consumption of logic resources. The proposed method was also highly scalable, which was demonstrated by implementing the beamformer with different beamforming rates ranging from 2.4 G to 9.6 G samples per second to three different sizes of FPGAs ranging from entry-level FPGA to high-end FPGA. The power consumption was less than 3 watts for 2.4 G samples per second beamforming rate, which demonstrates the possibility of implementing ultrafast ultrasound imaging on handheld probe. The FPGA beamformer's results were compared with Verasonics CPU beamformer's result to verify that the image quality was not compromised for speed.

11:15

5aBAa15. Three-dimensional time-domain full-waveform inversion for ring-array-based ultrasound computed tomography. Fu Li (Bioengineering, Univ. of Illinois, 3146 Everitt Lab 1406 W. Green St., Urbana, IL 61801, fuli2@illinois.edu), Umberto Villa (Oden Inst. for Computational Eng. and Sci., The Univ. of Texas at Austin, Austin, TX), Nebojsa Duric (Univ. of Rochester Medical Ctr., Rochester, NY), and Mark Anastasio (Bioengineering, Univ. of Illinois, Urbana, IL)

Ultrasound computed tomography (USCT) is a promising imaging technique for breast cancer diagnosis. Full-waveform inversion (FWI)-based image reconstruction methods based on the acoustic wave equation can produce quantitatively accurate images of acoustic properties with higher spatial resolution compared to ray-based methods. One common USCT design employs a circular ring-array of elevation-focused ultrasonic transducers that can be translated orthogonally to the imaging plane to achieve volumetric scanning. Currently, slice-by-slice (SBS) reconstruction methods are often used in this setting. Such methods use a 2D wave physic model to reconstruct a cross-sectional image for each position of the ring-array and vertically stack them together to render the 3D volume. However, this approach neglects the 3D wave propagation physics and transducer's focusing properties, leading to out-of-plane scattering artifacts and degraded spatial resolution. To address this, a 3D time-domain FWI method that utilizes measurement data acquired at multiple adjacent ring-array locations is proposed. The reconstruction method was investigated using virtual imaging studies of ring-array-based USCT that employed realistic 3D numerical breast phantoms. The results demonstrated improved spatial resolution (both in-plane and off-plane) and reduced image artifacts compared to the SBS approach. Additionally, the impact of the number of ring-array locations on image quality is assessed.

11:30

5aBAa16. Improving attenuation imaging of the breast with the spectral log difference technique and full angular spatial compounding. Mingrui Liu (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 506 S. Wright St., Urbana, IL 61820, ml132@illinois.edu) and Michael L. Oelze (Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Ultrasound tomography is an imaging technique for the breast that can create maps of the sound speed and acoustic attenuation simultaneously. By use of wave-based methods, images of speed of sound are of high quality, but the attenuation images are of inferior quality. The Spectral Log Difference (SLD) is a technique based on the ultrasonic backscatter that can provide estimates of the attenuation coefficient slope (ACS). However, SLD requires a large amount of data to provide a low variance of the estimate. This estimate variance can be reduced through compounding, which decreases spatial resolution of SLD images. Due to the nature of tomography, the tradeoff between estimate variance of SLD and spatial resolution can be reduced by using full angular spatial compounding. Using the data from a breast tomography scanner, the results demonstrated that the SLD technique with full angular spatial compounding could generate an accurate attenuation map with low bias and variance. As a result, SLD with full spatial angular compounding can provide improved attenuation images for breast diagnostics.

11:45

5aBAa17. Holography-based measurement of sound speed and attenuation coefficient values for small samples of tissue phantoms. Oleg A. Sapozhnikov (Phys. Faculty, Moscow State University, Russia, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, olegs2@uw.edu), Dmitry A. Nikolaev, Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova (Phys. Faculty, Moscow State University, Russia, and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

For therapeutic ultrasound applications, design and planning of treatments often involve simulations to predict the *in situ* acoustic field, and these simulations rely on estimates of the acoustic properties of soft tissues. As defined in simulation models, these properties describe how a plane wave propagates in a homogeneous medium. However, because soft tissues inherently contain inhomogeneities, measurements are most appropriately performed on small, relatively homogeneous samples. In addition, mostly small samples of human tissue, either autopsy or post-surgical, are typically available for characterization. As published in a recent paper [J. Acoust. Soc. Am. 2021, **149**(1): 386–404], we have proposed a holography-based technique to extract plane-wave measurements from ultrasound beams passing through small (cm-sized) material samples. Here, we seek to adapt this technique for measuring small samples of deformable materials using an apparatus that holds such samples with two plane, parallel surfaces that can be interrogated by an incident ultrasound beam having a diameter smaller than that of the sample. Initial testing over a range of frequencies from 0.5 to 10 MHz is performed utilizing well characterized materials, such as hydrogels and silicone rubber. [Work supported by NIH grant R01EB025187.]

12:00

5aBAa18. Delivery of sirolimus with intravascular ultrasound to prevent vascular restenosis. Xiaoqian Su (Nanyang Technol. Univ., Singapore, Singapore), Huiying Ang (National Univ. Singapore, Singapore, Singapore), Uday Illinadala (San Francisco, CA), Khung Keong Yeo (National Heart Ctr. Singapore, Singapore, Singapore), and James Kwan (Dept. of Eng. Sci., University of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, james.kwan@eng.ox.ac.uk)

Restenosis is the re-narrowing of arteries. It typically occurs after angioplasty treatment for peripheral artery diseasesand results in the need for invasive surgical treatments or amputation of the limb. Sirolimus is a known anti-restenosis drug, yet it requires long exposure time in the artery despite being quickly inactivated in blood. As such, there is a need for a targeted drug delivery methodology to promote slow-releasing sirolimus drug delivery vehicles. Our previous work suggested that sirolimus can be loaded into sound sensitive microparticles (SSMPs) comprised of PLGA. Here, we seek to determine if a bespoke intravascular device (IVUS) can deliver SSMPs to the arterial wall of a porcine artery *in vivo*. In this study, porcine arteries of three pigs were occluded by a balloon on the IVUS device distal to the treatment site, while drug loaded microparticles were injected through a sheath prior to ultrasound emission to facilitate uptake. We observed that the presence of SSMPs was larger in the arterial walls compared the blood and untreated arteries. The IVUS device provided a marginal increase in SSMPs in the treated arteries for most of the pigs. Thus, the *in vivo* studies suggested that the IVUS parameters were not yet optimized.

Session 5aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics: Bubbles and More

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Contributed Papers

7:30

5aBAb1. The effect of gas composition on the color Doppler ultrasound twinkling artifact. Eric Rokni (Graduate Program in Acoust., The Penn State Univ., State College, PA 16801, ezr144@psu.edu) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The twinkling artifact, or twinkling, appears when imaging kidney stones and other mineralizations with Doppler ultrasound and is caused by scattering off crevice microbubbles. Previous work found that twinkling on *in vivo* kidney stones increased or decreased with elevated oxygen or carbon dioxide, respectively. However, it is unclear whether these results are from biological adaptions to the changing respiratory gas and thus if *in vitro* mineralizations are similarly affected. Here, cholesterol, calcium phosphate, and uric acid crystals were grown *in vitro* and imaged in deionized water using a Vantage-128 research ultrasound system with a Philips/ATL L7-4 transducer. Five of each crystal composition were imaged for one minute at different dissolved gas concentrations. By bubbling gas through water, dissolved oxygen ranged from 2 to 20 mg/L while dissolved carbon dioxide ranged from 30 to 290 mg/L. Twinkling on cholesterol crystals increased by 70% and 140% when oxygen and carbon dioxide levels were maximized, respectively. In contrast, twinkling on calcium phosphate and uric acid was not significantly affected by either gas. These results suggest that changes in twinkling from increased gas concentrations are dependent on crystal composition and are likely affected by biological adaptations *in vivo*. [Work supported by NSF-CAREER-1943937 and NSF-GRFP-DGE1255832.]

7:45

5aBAb2. The effect of elevated oxygen on kidney stone twinkling *in vivo* and *ex vivo*. Laura Brownstead (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lgb5113@psu.edu), Necole Streeper (Penn State Health Milton S Hershey Medical Ctr., Hershey, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The Doppler ultrasound twinkling artifact is a rapid color shift observed in ∼66% of kidney stones. Breathing oxygen at elevated pressures was shown to enhance twinkling on kidney stones in human subjects. Here, in humans and in *ex vivo* stones, we investigate the effects of elevated oxygen at ambient pressure. Recruited subjects with confirmed stones were scanned with a research ultrasound system and Philips/ATL C5-2 transducer while breathing ambient air for 2 min. Then, subjects were masked with 100% oxygen for 15 min and scanned again for the last 2 min. Similarly, *ex vivo* stones were imaged before and after bubbling oxygen into water. Average Doppler power was calculated for *in vivo* and *ex vivo* stones. In human subjects, Doppler power increased by 30% when subjects breathed pure oxygen compared to ambient air. Similarly, on *ex vivo* stones, an average increase in Doppler power of 98% was observed in elevated oxygen conditions; twinkling also became visible on stones that otherwise did not twinkle at elevated oxygen (50% more oxygen than control). These results suggest that elevated oxygen at ambient pressure may improve the consistency of twinkling to detect kidney stones. [Work supported by the PSU Center for Biodevices Seed Grant.]

8:00

5aBAb3. Using 3D printed structures to evaluate the potential causes of the color Doppler twinkling signature. Benjamin Wood (Graduate School of Biomedical Sci., Mayo Clinic, 200 1st St. SW, Rochester, MN 55902, wood.benjamin@mayo.edu), Renc Saracaydin (Div. of Eng., Mayo Clinic, Kent, WA), 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 occurring on objects like kidney stones, some breast biopsy markers, and sandpaper. The grooves of helical constructs that twinkle may provide sufficient locations for bubble retention and/or cavitation. We developed six half-cylinders that replicate the geometry of twinkling helical constructs with a micro 3D-printing process to explore how their characteristics relate to twinkling. Four copies of each design including a control were created. The cylinders had pitch (groove-to-groove distance) of 87.5–343 μm and amplitude (groove depth) of 41.5–209 μm. The cylinders were submerged in degassed water and optically imaged before and after ultrasound to visualize bubbles on the cylinders. The cylinders remained submerged while scanning with color Doppler at frequencies from 3.1 to 6.3 MHz with a General Electric Logiq E9 scanner and 9L linear array transducer. Two designs that showed twinkling characteristics were further examined. The presence or absence of bubbles on these designs was confirmed with microscopy, and the resulting twinkling behavior was investigated. This work shows strong evidence that both existing visible bubbles and either cavitation or ultrasound wave interactions with patterned or rough surfaces are significant factors in producing the twinkling signature.

8:15

5aBAb4. The subharmonic resonance of lipid-encapsulated ultrasound contrast agents. Hossein Yusefi (Phys., Concordia Univ., Montreal, QC, Canada) and Brandon Helfield (Phys., Concordia Univ., 7141 Sherbrooke St. West, L-SP 365.04, Montreal, QC H4B 1R6, Canada, brandon.helfield@concordia.ca)

A salient feature of ultrasound contrast agent echo is their subharmonic response, a property that can be exploited for diagnostic contrast imaging. Here, we aim to explore the subharmonic behavior, specifically the subharmonic resonance, of two identical lipid-encapsulated microbubbles in close proximity to each other (center-to-center $h = 6$–16 μm) using finite element modeling. We simulated the subharmonic resonance response for bubbles ($R_0 = 0.5$–1.5 μm) driven with 10 or 20 cycle tone bursts from 6 to 18 MHz under a peak negative pressure from 40to 180 kPa over a range of initial phospholipid packing values ($\sigma_0 = 0$–0.01 N/m). Our results demonstrate that for increasing pressure, the transmit frequency at which the peak in subharmonic response is observed shifts monotonically towards higher frequencies (1–10%) for bubbles close to their buckling point, while the opposite trend occurs for bubbles with less dense initial packing (1–15%). For intermediate values of initial packing, a two-stage response is observed whereby the transmit frequency first decreases and then increases (5–9%) with increasing pressure. These results have implications in subharmonic-based diagnostic imaging techniques.

8:30

5aBAb5. Delayed subharmonic response of a phospholipid-shelled ultrasound contrast agent: Effect of temperature and protein concentration. Anuj Kaushik (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat 382355, India, a.kaushik@iitgn.ac.in), Surabhi Menon (Phys., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), and Himanshu Shekhar (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India)

The subharmonic response of ultrasound contrast agents (UCAs) is important for imaging and therapy. Therefore, factors influencing the temporal evolution of subharmonic signals should be elucidated. We evaluated the influence of temperature (25°C and 37°C) and protein concentration in the fluid (0.5% w/v at 25°C and 37°C) on the subharmonic response of Definity, a phospholipid-shelled UCA. Definity was diluted to 10^6 microbubbles/ml and sonicated at 2 MHz frequency, 50 cycles, and 470 kPa peak rarefactional pressure. The subharmonic response demonstrated striking qualitative and quantitative changes with time. At 25°C, a delayed subharmonic onset was observed. The subharmonic signal appeared after 28 min, peaking at 18 dB above the baseline. At 37°C, the subharmonic signal appeared immediately but it peaked (5 dB increase) after 52 min. In the presence of dissolved protein, the subharmonic signal appeared at 16 min at 25°C and at 4 min at 37°C. The signal peaked at 24 dB and 10 dB above the baseline at 25°C and 37°C, respectively. These observations underscore the need to standardize acoustic measurements and may have implications for subharmonic imaging, non-invasive pressure sensing, and cavitation detection.

8:45

5aBAb6. Time evolution of ambient pressure sensitivity of microbubbles subharmonic response as a function of hydrostatic pressure and the filling gas. Roozbeh Hassanzadeh Azami (Mech. and Aerosp. Eng., The George Washington Univ., 800 22nd St. NW, Ste. 3000, Washington, DC 20052, roozbehazami@gwu.edu) and Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC)

The ultrasound contrast microbubbles have shown a great ability to produce nonlinear oscillations in response to acoustic excitation which results in the generation of harmonics and subharmonics of the excitation. The subharmonic response of ½ order changes significantly with a change in hydrostatic pressure in the physiological range. This property is being investigated for subharmonic aided pressure estimation (SHAPE), a noninvasive alternative to current methods for vascular pressure measurement in critical organs. In this study, the ambient pressure sensitivity of the subharmonic response of microbubbles with different gas cores and shells was investigated over 25–700 kPa acoustic pressure and the hydrostatic pressure range of 0–25 kPa. The subharmonic showed different trends of increase and decrease with hydrostatic pressure. The sensitivity has shown gradual change over time which was observed to be related to the hydrostatic pressure magnitude and the gas core. These findings could help to calibrate the SHAPE.

9:00

5aBAb7. Characterization of lipid shelled microbubbles designed for targeting post-operative nascent abdominal adhesions. Victoria V. Doheny (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, vdoheny@bu.edu), Colleen M. McCarthy, Joanna Chiu (Biomedical Eng., Boston Univ., Boston, MA), Phillip Anderson (Andersonics LLC, Belmont, MA), Jo Ann Buczek-Thomas, Joyce Wong (Biomedical Eng., Boston Univ., Boston, MA), and R G. Holt (Phys., Hampden Sydney College, Hampden-Sydney, VA)

Peritoneal adhesions are bands of fibrous tissue that can bind adjacent tissue or organs together and have a high probability of occurrence after a patient undergoes deep abdominal surgery. These adhesions can cause complications, such as chronic pain, intestinal obstruction or constriction, organ displacement, and even death, yet there exists no common diagnostic for these adhesions prior to symptomatic appearance. Furthermore, the gold standard treatment is ironically surgical removal. Nascent adhesions formed in the first 48 hafter surgery are primarily composed of fibrin. In this study, lipid shelled microbubbles have been designed as a potential theranostic agent to detect and treat adhesions using a fibrin targeting peptide called CREKA (Cys-Arg-Glu-Lys-Ala). Low amplitude through-transmission experiments were conducted to characterize the mechanical properties of the microbubble lipid shell, such as the shear modulus of elasticity and shear viscosity. Passive cavitation detection (PCD) experiments were conducted to determine the inertial cavitation threshold by which the microbubbles may break up fibrin under ultrasound exposure. [Work supported by NIH SBIR and BU Mechanical Engineering Department.]

9:15

5aBAb8. A method to determine the bandwidth of imaging pulses for chirp-coded excitation imaging of histotripsy bubble clouds. Katia Flores Basterrechea (Radiology, Univ. of Chicago, 5841 S. Maryland Ave., MC 2026, Chicago, IL 60637, floresk@uchicago.edu), Giselle Miralles (Univ. of Chicago, Chicago, IL), and Kenneth B. Bader (Univ. of Chicago, Chicago, IL)

Histotripsy is a noninvasive focused ultrasound therapy that utilizes bubble cloud activity for tissue ablation. Real-time ultrasound imaging is used to guide histotripsy, and subharmonic imaging with chirp-coded excitation has been shown to provide effective bubble cloud contrast for targets at depth (>5 cm). Choice of appropriate parameters for the chirped imaging pulse are dependent on a number of factors, including the imaging probe bandwidth. In this study, an analytic method was developed and tested to design chirped imaging pulses for fundamental and subharmonic imaging. *In silico* studies were conducted to estimate received signal based on the frequency response of a curvilinear imaging probe (C5-2v, Verasonics, Inc., Kirkland, WA). To enable assessment of both bubbles and anatomic information, criteria were set to maximize the probe sensitivity for both fundamental and contrast-specific signals. Based on these analyses, *in vitro* studies were conducted using a scattering tissue phantom for sequences that highlight fundamental, subharmonic, or equally between the two for bubble cloud visualization. Good qualitative agreement was observed between *in silico* prediction and *in vitro* assessment of bubble cloud generation, indicating the formalism developed here is a promising approach for the development of imaging sequences for histotripsy guidance.

9:30–9:45 Break

9:45

5aBAb9. Development of an optically transparent, tendon-mimicking hydrogel for histotripsy. Jacob C. Elliott (Graduate Program in Acoust., The Penn State Univ., Res. West, State College, PA 16801, jce29@psu.edu) and Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Collagenous, anisotropic tissues such as tendon are resilient to liquefaction by histotripsy, despite verification of cavitation activity using B-mode imaging. In previous published work, dehydrated fibrin gels were shown to mimic tendon bubble dynamics but are limited in thickness due to high opacity. Here, we aim to fabricate a thicker, more optically transparent hydrogel possessing similar cavitation dynamics to tendon. Highly transparent collagen hydrogels (TeloCol®, 10 mg/mL) were fabricated, and axial sound speeds were measured to determine anisotropy. Hydrogels were then exposed to 1.5-MHz focused ultrasound with 10-ms pulses at 1-Hz for 60 s with $p+=127$ MPa, $p-=35$ MPa. Cavitation activity was monitored with high-speed photography and passive cavitation imaging using a Philips/ATL L7-4 transducer and Vantage® ultrasound system. Despite exhibiting low degrees of anisotropy like other non-dehydrated gels (<1.2), peak cavitation energy was >20% lower than soft-tissue mimicking polyacrylamide. Additionally, both resilience to fractionation and transparency were notably improved compared to previously tested PureCol® collagen formulations. Different levels of collagen dehydration will be explored to further increase resilience to liquefaction and assess the role of water content in histotripsy fractionation. [Work supported by NIH R01EB032860.]

10:00

5aBAb10. Design of an electronic marking clip for tracking breast cancer lesions through neoadjuvant chemotherapy using ultrasound identification. Jenna Cario (ECE, Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Urbana, IL 61801, jcario2@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

During the treatment of breast cancer via neoadjuvant chemotherapy (NAC), radiological clips are used to track lesions. Marking lesions allows them to be located and distinguished from their surroundings post-NAC, but morphological changes to the treated regions due to NAC can affect the visibility of marking clips in ultrasound, sometimes to a degree which requires the use of alternative, less comfortable modalities to visualize the clips in preparation for procedures, such as surgical resection or biopsy. In previous work, we proposed an electronic clip design leveraging active communication with an ultrasound imaging probe, improving visibility and differentiation of clips in ultrasound. The transmitted signal in a prototype of this design was successfully localized and identified by the ultrasound system in post-processing. Presently, we refine the device design by using pseudorandom noise (PN) codes as the ultrasonic identification signals to improve localization, and adjust the system design to allow multiple clips to be active and identified with greater accuracy in the imaging field in real time. [Work supported by a grant from the NIH (R21EB030743) and a fellowship from the Cancer Center at Illinois (CST EP082021).]

10:15

5aBAb11. Automated identification of a radiological beads in breast tumors for purpose of calibrating quantitative ultrasound. Yuning Zhao (Dept. of Elec. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 1007 W Univ., 311, Urbana, IL 61801, yuningz4@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Quantitative Ultrasound (QUS) is an imaging method that has demonstrated the ability to detect the response of breast cancer to neoadjuvant chemotherapy. QUS is sensitive to tumor cell death, which is a hallmark of response. Traditionally, QUS requires an external reference material, e.g., reference phantom, to calibrate the system for backscatter coefficient based parameters. However, this approach does not account for losses *in vivo* and requires additional scans. To alleviate these issues, we used an *in situ* calibration target, i.e., a 2-mm diameter titanium bead, to obtain a reference signal. To automate the procedure to locate the bead in a B-mode image, we constructed a data mining method to distinguish the bead signal from other similar signals recorded from tumors having an embedded bead. The method was able to successfully locate the bead signal in a tumor image with good accuracy.

10:30

5aBAb12. Evaluating the feasibility of differentiating glioblastoma and normal brain tissue using H-scan imaging: An *ex vivo* study. Manik Kakkar (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), Jagruti M. Patil (Biological Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), Vishwas Trivedi, Anushka Yadav (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), Shilpa Rao (Neuropathology, National Inst. of Mental Health and Neurosciences, Bangalore, Karnataka, India), Vikas Vazhayil (Neurosurgery, National Inst. of Mental Health and Neurosciences, Bangalore, Karnataka, India), Anita Mahadevan (Neuropathology, National Inst. of Mental Health and Neurosciences, Bangalore, Karnataka, India), Himanshu Shekhar (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), and Karla Patricia Mercado-Shekhar (Biological Eng., Indian Inst. of Technol. Gandhinagar, Academic Block 6/207, Gandhinagar, Gujarat 382355, India, karlamshekhar@iitgn.ac.in)

H-scan imaging is a quantitative ultrasound technique in which raw radiofrequency (RF) echoes are matched to Gaussian-weighted Hermite polynomials of order n, which is related to the size of scatterers. This information is used to characterize tissue structures based on color-mapping large scatterers in the red channel and small scatterers in the blue channel of the H-scan image. Glioblastoma is a malignant tumor of the brain and has one of the most dismal prognoses of all tumors. In this study, we assessed the feasibility of differentiating glioblastoma and normal brain tissue *ex vivo* using H-scan imaging. Raw RF echoes from *ex vivo* brain tissues embedded in agarose phantoms were acquired using a Verasonics Vantage 128 system equipped with an L11-5v array. A total of 11 samples each of glioblastoma and normal tissues were evaluated. The intensity-weighted percentage of the red channel (IWP_{RED}), a scatterer size-dependent and concentration-independent parameter, was computed for each sample. The IWP_{RED} parameter estimates of glioblastoma and normal tissue samples were significantly different ($24.0\% \pm 0.7\%$ vs $34.2\% \pm 1.1\%$, respectively; $p < 0.001$). This work demonstrates the feasibility of differentiating glioblastoma tumors from normal tissue using H-scan imaging.

10:45

5aBAb13. Experimental detection of skull-based ultrasonic Lamb waves as an intracranial pressure monitoring method. Dan L. Nguyen (Simmons Univ., 300 Fenway, Boston, MA 02115, dan.nguyen2@simmons.edu) and Phillip J. White (Simmons Univ., Boston, MA)

Pressure within the cranial vault, which consists of cerebrospinal fluid, the central nervous system (CNS), and blood, is referred to as intracranial pressure (ICP). Severely high ICP can damage the CNS so its monitoring is crucial for patients deemed to be at high risk of elevated ICP, such as those with traumatic brain injuries or undergoing neurosurgery. The standard approach requires the insertion of a pressure probe into the ventricles through a burr hole on the skull. A noninvasive alternative to ICP monitoring using guided acoustic waves on the skull was investigated. Different modes of Lamb waves were generated in submerged acrylic skull models and their corresponding leaky components were detected. The behavior of Lamb waves upon asymmetrical pressure loading of the plate was examined to reveal metrics that were sensitive to pressure changes underneath the skull. Dispersion curves, including the anti-symmetrical and symmetrical modes of Lamb waves propagation in thin, isotropic acrylic plate comparable to thickness of cortical skull bone were computed using the Rayleigh–Lamb equation and the bisection method. Physical parameters, such as critical angle, frequency-dependence phase velocity, and time-of-arrival were analyzed.

11:00

5aBAb14. Effect of focused ultrasound peripheral nerve stimulation on muscle mechanical properties: An *in vivo* murine model. Jacob C. Elliott (Graduate Program in Acoust., The Penn State Univ., Res. West, State College, PA 16801, jce29@psu.edu), Zoe Moore, Meghan E. Vidt (Biomedical Eng., The Penn State Univ., State College, PA), and Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Electrical muscle stimulation is limited to surface muscle activation; however, focused ultrasound (fUS) has the potential to stimulate deep skeletal muscles. This study aims to evaluate the effects of fUS peripheral nerve stimulation on muscle mechanical properties in an *in vivo* murine model. The sciatic nerve of 19 Sprague-Dawley rats was insonified with 3.68-MHz fUS ($p_+ = 6.4$ MPa, $p_- = 5$ MPa) using 10-ms pulses (0.9-ms on/0.1-ms off, 10 times) repeated at 0.5-Hz for 30, 90, or 180 severy second day for two weeks. Stimulation was monitored using electromyography (EMG) with indwelling electrodes inserted in the gastrocnemius muscle belly. Bilateral muscles were harvested for tensile mechanical testing ($n = 15$) and histology ($n = 4$). Muscle contraction was confirmed via EMG in 16/19 rats. Elastic moduli of stimulated muscles (133.2 ± 42.1 kPa) and controls (101.4 ± 39.2 kPa) were not different (ANOVA, $p = 0.053$). A trend toward increased elastic modulus with number of muscle stimulations was observed until 450 stimulations (not significant: ANOVA, $p = 0.437$), after which elastic modulus began to decrease. No tissue damage was observed histologically. The results of this pilot study demonstrate that fUS can induce repeatable muscle contraction that may improve muscle strength and mechanical properties over time. [Work supported by PSU College of Engineering Seed Grant 2021.]

11:15

5aBAb15. Quantification of swallowing movements by ultrasound imaging in normal and disordered subjects. Nicholas S. Schoenleb (Biomedical Eng., Univ. of Cincinnati, UC Bioscience Ctr., 3159 Eden Ave., ML 0048-04D, Cincinnati, OH 45221, schoenna@mail.uc.edu), Anna K. Hopkins, Brittany N. Krekeler, Rebecca J. Howell (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Sarah Dugan (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Dayton, OH), Michael A. Riley (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Cincinnati, OH), AnnaKate Spotts (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Ultrasound imaging may be useful for tracking oral-pharyngeal structures of the vocal tract critical for effective swallowing, but use of this technology has been hampered by a lack of validation against standard clinical measures derived from x-ray video fluoroscopy (Modified Barium Swallow Studies or MBSS). Here, we compare quantitative tracking of hyoid bone and tongue motion from ultrasound imaging and simultaneously measured MBSS from a range of participants with disordered and healthy swallowing behavior. The proximal edge of the hyoid bone in MBSS and ultrasound recordings is tracked using CASM (Computational Analysis of Swallowing Mechanics) software, while tongue motion is tracked using TonguePART software, originally developed for analysis of speech. Hyoid trajectories and velocities measured from synchronized ultrasound and MBSS recordings are compared and standard deviations of differences between the two measurements are evaluated. Tongue motion during swallowing is characterized by analysis of root, dorsum, and blade displacement, velocity, and timingas well as tongue base retraction in the pharynx. Preliminary findings show quantitatively similar patterns in ultrasound and MBSS, suggesting that ultrasound has potential utility as an accessible tool for bedside diagnosis and biofeedback. Characteristic patterns separating normal from healthy swallowing, as well as potential therapeutic targets for biofeedback treatment of dysphagia, will be discussed.

11:30

5aBAb16. Numerical simulations for characterization of missing articulatory information in ultrasound imaging of speech. Sarah R. Li (Biomedical Eng., Univ. of Cincinnati, UC Bioscience Ctr., ML 0048-04D, 3159 Eden Ave., Cincinnati, OH 45219, lisr@mail.uc.edu), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

The air–tissue interface at the tongue surface allows submental (below the chin) B-mode ultrasound images to depict the shape of the tongue in real time, useful for measuring and understanding typical and disordered tongue articulation during speech production. However, parts of the tongue are obscured; depending on an individual speaker's anatomy and tongue movement patterns, the anterior tongue tip may be shadowed by the mandible bone and the sublingual airspace. To characterize this missing information about the tongue tip in ultrasound images, midsagittal ultrasound images were numericallysimulated. Three-dimensional anatomic models were constructed using magnetic resonance imaging (MRI) data to map tongue tissue, surrounding air, and an estimated position of the mandible. Pulse-echo ultrasound images were then numerically simulated using the open-source k-Wave toolbox. Acoustic properties of the medium and an imaging array model are identified that allow simulations to replicate features in ultrasound images recorded from the same speakers as the MRI data, while maintaining computational efficiency. The potential to recover missing information on the tongue tip position from reverberation artifacts is assessed.

11:45

5aBAb17. Voice recovery in eight Yucatan mini-pigs after vocal fold implantation surgery. Patrick Schlegel (Head and Neck Surgery, Univ. of California, Los Angeles (UCLA), 1000 Veteran Ave., Los Angeles, CA 90095, pschlegel@ucla.edu), Anthony Chen, Yazeed Al-Hiyari, and Jennifer Long (Head and Neck Surgery, Univ. of California, Los Angeles (UCLA), Los Angeles, CA)

As there currently exists no standardized framework to assess pig voice health, and voice parameters designed for humans do often not work on aperiodic pig voice, new approaches must be developed. We performed cell-based outer vocal fold replacement (COVR) surgery in six Yucatan mini-pigs (with two pigs in the control group) and tracked functional and structural voice recovery over up to six months. Recovery of voice function was assessed by tracking acoustic parameters and spectrograms in three pigs. Structural recovery was assessed by histologic analysis and Amino Acid Quantitation after harvest of vocal fold tissue in three pigs each. The results of functional analysis reflect expected voice worsening right after surgery and gradual recovery in one pig that was tracked the longest. Structural analysis shows small differences in fiber directionality between control group and pigs after surgery. Increasing amounts of detected amino acid markers for collagen and elastin over time after surgery may hint on gradual structural recovery. With this preliminary work, we form the basis for a larger long-term pre-clinical safety and efficacy study of COVR implantation.

12:00

5aBAb18. Non-contrast-enhanced ultrasound microvasculature imaging and quantification for differentiation of thyroid nodules. Soroosh Sabeti (Mayo Clinic, 200 1st St. SW, Rochester, MN 55902, Soroosh_Sabeti@yahoo.com), Melisa Kurti (Rochester, MN), Nicholas B. Larson (Mayo Clinic, Rochester, MN), Mostafa Fatemi (Physiol. & Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN), and Azra Alizad (Mayo Clinic, Rocheter, MN)

Observation of thyroid nodules in ultrasound examinations is a common occurrence. However, accurate diagnosis of the malignancy status of these nodules is a challenging task relying only on conventional ultrasound imaging. Therefore, increasing the sensitivity and specificity of non-invasive screening methods can be helpful in reducing the probability of missed diagnosis and decreasing the rate of unnecessary biopsies. In this feasibility study, a quantitative ultrasound microvasculature imaging technique is applied to ultrasound data collected from a cohort of 92 patients. This technique consists of first enabling the visualization of vascular networks inside the scanned nodules, followed by a morphometric analysis framework with a number of biomarkers to distinguish between benign and malignant thyroid nodules. Statistical tests are performed to find significant differences between the distribution of the values of the biomarkers. Subsequently, these biomarkers are utilized to train classification models that solely benefit from morphological biomarkers and those trained with additional clinical features. The receiver operating characteristic (ROC) curve analysis of these models, which helps us evaluate their predictive power, is presented. In our preliminary results, the model trained with combined morphological and clinical features achieves an AUC of 0.90, a sensitivity of 0.81, and a specificity of 0.79. [Work supported by R01CA239548.]

Session 5aED

Education in Acoustics, Physical Acoustics, Musical Acoustics, and Engineering Acoustics: Resources for Teaching Waves in a Physics Class (Hybrid Session)

Andrew C. Morrison, Cochair
Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Andrew A. Piacsek, Cochair
Dept. of Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926-7422

Chair's Introduction—9:00

Invited Papers

9:05

5aED1. ExploreSound.org—An Acoustical Society of America Educator Resource. Keeta Jones (Acoust. Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11747-4300, kjones@acousticalsociety.org)

The Acoustical Society of America Committee on Education in Acoustics (ASA EdCom) developed ExploreSound.org for young people and educators who are interested in learning about sound and the scientists who study sound. This educational website consists of open access material generated by ASA as well as externally created content found to be beneficial to teachers, librarians, professors, and anyone else interested in acoustics education. To effectively communicate the technical content of acoustics and its importance, ExploreSound.org is divided into three main audiences: educators, K-12 students, and university students and acoustics topics can be explored in all technical areas represented in the ASA from Animal Bioacoustics to Underwater Acoustics. Educators and students can freely access and search all games, science fair projects, acoustician profiles, lesson plans, and more. ExploreSound.org supports acoustics education at all levels and in all areas of expertise. All acousticians, professional and student alike, can visit the site to participate in the long tradition of acoustics education.

9:25

5aED2. Strategies for teaching waves in the next generation science standards. Marta R. Stoeckel (N St Paul - Maplewood - Oakdale Public Schools, 2520 12th Ave. E, North St. Paul, MN 55109, mstoeckel@isd622.org)

Waves and their applications are one of the disciplinary core ideas in the Next Generation Science Standards (NGSS), but the performance expectations include concepts that have not historically been an important part of teaching waves in high school science classrooms, with a particular focus on technologies that utilize waves and on students making connections between different types of waves including sound, seismic waves, and electromagnetic waves. In this talk, I will share resources and strategies that I use to teach some of these concepts, including both hands-on activities and digital resources. Many of these resources are also appropriate to other physics courses, including AP Physics and introductory college physics courses.

9:45

5aED3. Using an online graphing calculator as an aid in teaching the physics of waves. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

In teaching the physics of wave phenomena, it is frequently helpful to be able to interact with various representations of waves rather than showing static images to students. Through the online graphing calculator platform, Desmos.com, students can interact with animated graphics of different wave behaviors, including traveling waves, standing waves, wave interference, beats, and the Doppler effect. The platform also includes tools for instructors to design interactive activities for students to complete individually as homework or as part of a classroom activity. Instructors can share graphs, interactive simulations, and class activities with other instructors for reuse and modification in other classes. This online tool is an efficient way to bring more interactivity into the classroom for exploring wave behavior.

10:05

5aED4. Wave phenomena andthe high school AP physics classroom. Cameron T. Vongsawad (Sci. Dept., Herriman High School - Jordan School District, 5166 W Koppers Ln., Herriman, UT 84096, cameron.vongsawad@gmail.com)

Though a staple for modeling natural phenomena in much of modern science, oscillatory motion consists of at best 14% of any high school AP Physics exams focus and at most 6% in AP Physics 1. With such a minimal emphasis, wave phenomena can easily be quickly skipped over at the end of the school year for a typical AP Physics classroom rushing to wrap up before the exam in May. This presentation will share fun and simple ways to more fully incorporate major wave phenomena such as sound into an AP Physics curriculum, while supporting the College Board AP Physics units on Simple Harmonic Motion and Oscillations, with interactive demonstrations and hands-on activities. These demonstrations and activities provide for a deeper understanding of major curriculum ideas of oscillations as well as energy transfers in various systems, which the modern high school student can get excited about and easily relate to. With these small changes students have often reported this unit to be the most interesting and easily understood units of the year.

10:25–10:40 Break

Contributed Papers

10:40

5aED5. Building an electronic speckle pattern interferometer to visualize modal shapes in an educational setting. Eric Rokni (Graduate Program in Acoust., The Penn State Univ., State College, PA 16801, ezr144@psu.edu), Kyle S. Dalton, Dan Hendricks, Trent Furlong, John A. Case, Steven Todd (Graduate Program in Acoust., The Penn State Univ., State College, PA), Aaron Stearns (Appl. Res. Lab., The Penn State Univ., State College, PA), and Zane T. Rusk (Dept. of Architectural Eng., The Penn State Univ., University Park, PA)

An electronic speckle pattern interferometer (ESPI) is an optical system used to investigate vibrating objects and can be a great educational tool for visualizing mode shapes. At the Pennsylvania State University, the Acoustical Society of America student chapter acquired funds to build an ESPI for classroom and outreach use. The design of the ESPI was based on previously published work using equipment purchased for less than $5000. To approach this project, students were split into two groups where one developed a MATLAB application to collect and process images while the other arranged the optical components of the ESPI. To test the system, interferometry was performed on a banjo head which matched mode shapes found using a scanning Laser Doppler Vibrometer (LDV). Future work includes adapting our ESPI system to obtain images in real-time, developing educational demos involving the ESPI, and further refining the MATLAB application. This presentation provides a model for a multi-disciplinary project that could be implemented at the undergraduate or graduate level.

10:55

5aED6. Penn State acoustic wave kits (PAWKits). Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The increasing prevalence of blended resident and distance education courses makes it difficult to implement in-class demonstrations or hands-on activities to enhance learning. Our objective here is to develop and evaluate the use of at-home activity kits in fundamental acoustics and vibration courses for graduate students. First-generation PAWkits includedfive1-ft PVC pipes, four PVC couplers, a PVC end cap, a PVC T-branch, two dental dams, a tuning fork, a bluetooth speaker with passive radiator, a lapel microphone, a thick string, a thin string, ten magnets, four magnetic hooks, four springs (two spring constants), and an alligator clip. The total cost of the kit per student was estimated at $66 plus shipping. The first-generation PAWkit was designed for five main activities includingmeasuring directivity of the speaker and tuning fork, waves in pipes including T-branch and low-pass filters, waves in membranes, waves in strings, and mass-spring vibration systems. Activities were tested across two required Penn State first-year required courses for graduate students. PAWkits are currently being evaluated through student surveys and, in the future, by tracking the performance of Ph.D. students on the qualifying exam. [Work supported by Penn State Leonhard Center Seed Grant.]

11:10

5aED7. Educational high school physics demonstration apparatus for acoustic landmine simulant resonance detection. Rebecca D. Burge (Phys., U.S. Naval Acad., U.S. Naval Acad., 572C Holloway Rd., Chauvenet Hall Rm. CH040, Annapolis, MD 21402, deniseburge04@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

This landmine "simulant" resonance detection demo was created in conjunction with a high school physics mentorship, using a variety of easily-accessible equipment in order to demonstrate an important application of fundamental concepts of sound and vibration. A plastic tray (4 in. deep) contained a 3-in. diameter drum-like circular landmine simulant (1.5 in. high) with a thin acrylic top plate buried (1 in. deep) in a 3 in. layer of dry sifted masonry sand. A small accelerometer was place on the surface over the simulant. Airborne sound, a loud speaker chirp, with frequency slowly increasing over time, was created using a Mathematica® generated wav file played by the Raspberry Pi® computer. Sound excited the sand vibration that coupled to the top plate. The accelerometer generated a frequency response voltage, related to the sweep time. The Raspberry Pi® also played a role in implementing the A to D conversion, so that the frequency response near resonance was displayed as a voltage in a readable format. With this demo apparatus high school students will be excited by the fundamentals of wave propagation and concepts about resonance with regards to its real-life applications. [Specially inspired by James M. Sabatier's research on acoustic landmine detection.]

11:25

5aED8. Waves and fun acoustics. David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcousticsDB@gmail.com)

The presenter shares some experience with teachings and demonstration on waves, vibrations, and sound from classroom visits at kindergarten lovel, after school elementry school science enrichment, high school AP physics, to required graduate course in Acoustic and Electromagetic waves. Topics includestanding waves and lasagna, vibrating strings, coupled oscillators, parametric excitation, boundary conditions, effects of gravity, phonographs, and the Shive wave machine.

11:40

5aED9. Measuring and analyzing sound using phone microphone and free software. Andrew Wright (Mech. Eng., Univ. of Arkansas at Little Rock, 2801 South University Ave., Little Rock, AR 72204, abwright@ualr.edu)

In Fall 2022 acoustics class, the students developed class projects using their smart phones to measure sound. In a laboratory exercise, the phones were calibrated against a standard microphone. A microphone circuit was bread boarded and calibrated to demonstrate the inner workings of electret microphone technology. Students developed projects around the

measurement of sound, including measuring striking a cymbal, analyzing piano chords, analyzing guitar chords, characterizing reverberation time in a warehouse, and determining the amplification from a 3D printed human ear and bat ear. The sound files from the phone were edited to isolate the relevant sound and converted to WAV files using Garageband software. The WAV files were imported into Python for post-processing, such as windowing, filtering, and FFT. Most projects used a waterfall plot to demonstrate some time-frequency analysis. The ease with which the students mastered sound measurement and processing led to unexpectedly mature projects. The low cost of the equipment makes this approach a widely deployable educational paradigm.

FRIDAY MORNING, 12 MAY 2023 — CHICAGO F/G, 9:00 A.M. TO 11:15 A.M.

Session 5aPA

Physical Acoustics: General Physical Acoustics II: Application, Measurements, and Novel Effects

Gregory W. Lyons, Cochair
DEVCOM Army Research Laboratory, 2800 Powder Mill Rd., Adelphi, MD 20783

Carl R. Hart, Cochair
U.S. Army Engineer Research and Development Center, Cold Regions Research and Engineering Laboratory, 72 Lyme Rd., Hanover, NH 03755

Contributed Papers

9:00

5aPA1. Time domain response of a piezoelectric layer. 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 James Tressler (Acoust., Naval Res. Lab, Washington, DC)

A number of one-dimensional models have been developed to inform the design of piezoelectric transducers. The majority of these models are in the frequency domain. In this paper, we develop a one-dimensional time-domain model for the mechanical response of a piezoelectric layer. Secondary effects, resulting from feedback between the acoustic and electric variables, are included in the model. Our approach utilizes the Green's function for the Helmholtz equation with radiation boundary conditions and the methods of complex analysis. The model predictions are validated by comparison with a finite-difference time-domain numerical simulation of the governing acoustic equations in and outside the layer. This time-domain model enables efficient calculation of the secondary piezoelectric action effects and provides the mechanical response to an arbitrary electrical source.

9:15

5aPA2. Characterization of 3-D printed porous materials for use in a low cost impedance tube. Candace Tapia (Phys. and Astronomy, Univ. of Central Arkansas, 201 Donaghey Ave., LSC 171, Conway, AR 72035, ctapia@cub.uca.edu) and Carl Frederickson (Phys. and Astronomy, Univ. of Central Arkansas, Conway, AR)

With improvements in 3-D resin printing, it is now possible to design and print porous materials for testing in an acoustic impedance tube. The measured impedances of these materials can be compared to theoretical calculations. To do this, physical parameters, such as porosity and flow resistance, must be measured for the printed materials. Procedures for measuring these parameters will be discussed as well as preliminary measurements of the acoustic impedance. To measure flow resistance on the printed samples, a system based on the standard "Acoustics—Determination of airflow resistance Part 1: Static airflow method" (ISO 9053-1:2018) has been built and tested. These results will be compared to estimates of flow resistance through straight, round tubes. Porosity is measured in a simple volume comparison measurement.

9:30

5aPA3. Imaging based pore network modeling of acoustical materials. Arash Rabbani (School of Computing, Univ. of Leeds, Leeds, United Kingdom), Brittany Wojciechowski (Aerosp. Eng., Wichita State Univ., Wichita, KS), and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, bhisham.sharma@wichita.edu)

The acoustical behavior of porous materials is dictated by their underlying pore network geometry. Given the complexity of accurately characterizing the various pore network features, current acoustical models instead rely on indirectly incorporating these features by accounting for them within acoustical transport properties, such as tortuosity, viscous and thermal characteristic lengths, and flow resistivity. In turn, these transport properties are currently identified using inverse characterization techniques or using multiphysics modeling techniques. Here, we propose the use of advanced image processing methods to characterize the pore network of acoustical materials and allow the direct calculation of their transport and acoustical properties. To establish the feasibility of this idea, we create 3D printable CAD models of porous materials with controlled pore geometries and use a Matlab-based watershed segmentation technique to calculate their effective pore and throat size distributions. These distributions are then used to calculate their transport properties and predict their sound absorption coefficients using the Johnson–Champoux–Allard model. For comparison, we calculate the transport properties using the hybrid multiphysics modeling technique and the inverse characterization method. The predictions from the three different methods are then compared with experimental measurements obtained by printing and testing the models using an impedance tube.

9:45

5aPA4. Interferometric measurements of laser-induced shockwaves in air: Variable laser energy. 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), Gregory W. Lyons (DEVCOM Army Res. Lab., Adelphi, MD), and Michael J. White (U.S. Army Engineer Res. and Development Ctr., Construction Eng. Res. Lab., Champaign, IL)

When focused to a small spot size in air, a sufficiently energetic laser pulse initiates a rapidly expanding plasma. After a delay, a shockwave detaches from the plasma boundary and propagates. General features of the shockwaves can be deduced from condenser microphone measurements. However, the minimum range is limited by damage thresholds, and the presence of the microphone introduces a number of measurement artifacts. Distortion of the signal is caused by diffraction around the sensor, and the limited bandwidth does not allow rise times to be correctly quantified. In contrast, optical interferometry is a nonintrusive diagnostic for quantifying shockwave characteristics. In this study, a Nd:YAG laser is focused through a converging lens in order to generate laser-induced shockwaves. By using a variable attenuator, four laser energy outputs are examined: 25, 50, 75, and 100% of the maximum energy transmission. Heterodyne Mach–Zehnder interferometer measurements are made from 10 mm to 200 mm from the focal point of the lens. Virtual velocity signals, proportional to the time derivative of optical phase differences, are used to estimate density and pressure time histories, along with peak pressure as a function of distance.

10:00–10:15 Break

10:15

5aPA5. Synthesis of nonuniformly spaced arrays. Nguyen Le (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), Jenny Y. Au (Federal Aviation Assoc., Lowell, MA), Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, University Ave., College of Eng., Lowell, MA 01854, kavitha_chandra@uml.edu), and Charles Thompson (Elec. and Comput. Eng, Univ. of Massachusetts Lowell, Lowell, MA)

This work presents an analysis of beam formation from nonuniformly spaced planar microphone arrays. The effect of transducers' random and deterministic placement on the array's response characteristics is considered. Arrays that exhibit maximal sidelobe reduction are of particular interest. A solution for the transducer density as a function of the position is given.

10:30

5aPA6. Frequency-dependent surface wave suppression at the Dirac point of an acoustic graphene analog. 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), Alec K. Ikei, Amelia Vignola (Code 7160, U.S. Naval Res. Lab., Washington, DC), Jeffrey Baldwin, Steven Liskey, Aaron Edmunds, William Wilson (Physical Acoust., U.S. Naval Res. Lab., Washington, DC), Michael Boone (Jacobs Eng., Washington, DC), Gregory Yesner, Douglas Photiadis, and Bernard Matis (Physical Acoust., U.S. Naval Res. Lab., Washington, DC)

The dispersion of bound acoustic surface waves over hexagonal lattices of resonant cavities has been shown to be analogous to the dispersion of charge transport in carbon graphene. Of particular interest is the frequency range close to the acoustic Dirac point where novel physics is predicted to occur. In this study, we measure the dispersion curves of a single-layer acoustic graphene analogue with high resolution one-dimensional spatial scansand show how the curves can be suppressed (near and at the Dirac point) by strong variations in the impedance boundary conditions between the free field and surface wave regimes under certain experimental conditions. By systematically varying these impedance boundary conditions using different surface wave excitation techniques, we demonstrate that increased Rayleigh scattering and diffractive excitation can increase the dispersed surface wave pressure amplitude to an extent that the impedance-based wave suppression is circumvented. The improved conditions for observing acoustic Dirac points for two samples with two distinct operational frequency ranges are reported. The single-layer acoustic graphene analogue results discussed here are important for advancing the field of acoustic twistronics.

10:45

5aPA7. Classical analogous quantum coherent superposition through the Hertz-type nonlinearity of elastic granules. Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Ste. #2100, Detroit, MI 48201, hk2799@wayne.edu) and M Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI)

The theory of quantum superposition of states is one of the fundamental concepts related to quantum information science and technology (QIST). Two or more pure quantum states can be superposed to get a distinct quantum state, and one of the examples is the quantum bit, or qubit, state. But decoherence makes it easy for the current qubits, the most critical part of QIST platforms, to lose their superposition of states. Therefore, we theoretically propose and experimentally realize a classical analogous to the coherent superposition of energy states through the Hertz-type nonlinearity of elastic granules driven externally. The granules' nonlinear vibrations depend mutually through phase and form a coherent superposition when projected into linear modes of vibration. We demonstrate how the state vector components can emerge from the amplitudes of coherent states, spanning to the two-dimensional time-dependent parametric Hilbert space. These amplitudes represent an actual amplitude rather than a probability amplitude, which opens the possibility of exploring two-state quantum-like computations through the superposition of states without decoherence and wave function collapse. These characteristics make it possible to understand the material-based quantum analogous information technology through experimentation of elastic bit.

11:00

5aPA8. Quantum analogue information processing using "classical entanglement" of acoustic waves. M Arif Hasan (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, Hasan.Arif@Wayne.Edu), Pierre Deymier, Keith Runge (Mater. Sci. & Eng., Univ. of Arizona, Tucson, AZ), and Joshua Levine (Comput. Sci., Univ. of Arizona, Tucson, AZ)

Various quantum analogues have emerged due to recent advances in the design and fabrication of acoustic structures. Acoustic analogous to "entangled" states of quantum bits or qubits offer a transformative new solution to reach some of the goals of quantum information science. Beams of linear acoustic waves support degrees of freedom (DOF), analogous to quantum mechanics and light beams. The states associated with these DOFs—that we named phase-bits or phi-bits—are algebraically non-separable, like the DOF of a single quantum particle. The strong coupling and nonlinearity of acoustic waves in our waveguides offer new ways to realize non-separable states for classical multipartite systems that span exponentially complex Hilbert spaces with much larger dimensions [1,2]. These acoustic structures enable quantum analogue computing even at room temperature and for long periods of coherence. [Work supported by NSF.] [1] Hasan, M A, *et al*. "Experimental classical entanglement in a 16 acoustic qubit-analogue." *Scientific Reports* 11, 2021. [2] Runge, K, *et al*. "Demonstration of a two-bit controlled-NOT quantum-like gate using classical acoustic qubit-analogues." *Scientific Reports* 12, 2022.

Session 5aPP

Psychological and Physiological Acoustics and Speech Communication: Environmental Sounds: Perception, Cognition, Applications

Laurie M. Heller, Cochair
Psychology, Carnegie Mellon University, 5000 Forbes Ave., Baker Hall 332C, Pittsburgh, PA 15213

Valeriy Shafiro, Cochair
Communication Disorders and Sciences, Rush University Medical Center, 600 South Paulina Street, 1015 AAC, Chicago, IL 60612

Chair's Introduction—8:00

Invited Papers

8:05

5aPP1. Environmental sound research today: Perception, cognition, applications. Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 South Paulina St., 1015 AAC, Chicago, IL 60612, Valeriy_Shafiro@rush.edu)

Environmental sounds, defined as semantically rich acoustic signals different from speech or music, are ubiquitous in daily living. Environmental sound research in various forms has been long embedded into many areas of acoustics including noise control, bioacoustics, music, signal processing, speech communication, and psychoacoustics. However, environmental sounds have been typically investigated in the limited context of specific lines of inquiry unique to each of these areas. In this fractured research landscape, there is also a growing body of work which focuses on ecological aspects and elucidates the neurobiological bases of environmental sound perception and cognition. We will review main questions addressed in recent work, including existing theoretical and methodological challenges and opportunities, and suggest directions for developing a more comprehensive framework for future environmental sound research and applications. A better understanding of environmental sound perception and cognition can in turn have practical implications for many diverse areas of acoustics and related fields.

8:25

5aPP2. The causal properties of sounds predict their recognition and misidentification. Laurie M. Heller (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Baker Hall 332C, Pittsburgh, PA 15213, hellerl@andrew.cmu.edu), Urszula Oszczapinska, and Jessica M. Smith (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

One way to study human sound recognition is to investigate the reasons why sounds are sometimes misheard as coming from the wrong source. Understanding this cognitive process can not only help prevent undesirable sound confusions (e.g., auditory display design) but can also promote useful confusions (e.g., Foley effects, cognitive reappraisal for misophonia). We tested the hypothesis that sounds are more confusable if their source events share causal properties. In Expi.1, listeners assessed causal properties of everyday sounds (ESC-50 Dataset) by judging their actions (e.g., tapping), materials (e.g., metal), and causal agents (e.g., machines). Causal similarity between sounds was measured by the distance between their causal properties. In Exp. 2, new listeners identified these sounds with 90% accuracy. Using the distances obtained in Exp. 1, misidentifications were predicted with 91% sensitivity and 89% specificity. The causal properties that had the largest effect on recognition accuracy were predominantly actions (according to a Ridge regression). Additional experiments with our own recorded and synthesized sounds show that spectral degradation and temporal manipulation can cause significant changes in causal properties and source assignment. Altogether, these findings show that causal properties provide unique explanatory power for sound recognition and misidentification. [Work supported by REAM.]

8:45

5aPP3. Comparing acoustic, semantic, and deep neural network models of natural sound representation in perceived dissimilarity and cerebral fMRI responses. Bruno L. Giordano (Institut des Neurosciences de La Timone, UMR 7289, CNRS and Université Aix-Marseille, Campus santé Timone, 27, Boulevard Jean Moulin, Marselle, Foreign 13385, France, bruno.giordano@univ-amu.fr), Michele Esposito, Giancarlo Valente, and Elia Formisano (Dept. of Cognit. Neurosci., Faculty of Psych. and Neurosci., Maastricht Univ., Maastricht, Netherlands)

Recognizing sounds implicates the cerebral transformation of input waveforms into semantic representations. Although past research identified the superior temporal gyrus (STG) as a crucial cortical region, the computational fingerprint of these cerebral transformations remains poorly characterized. Here, we contrasted the ability of pre-published acoustic, semantic, and sound-to-event deep neural network (DNN) models to account for the behavioral estimate of the perceived dissimilarity, and for the 7T fMRI responses to natural sounds in the absence of explicit task demands. We find that both perceived dissimilarity and STG fMRI responses are better predicted

by sound-to-event DNNs, and, within the DNNs, by the layers intermediate between the input acoustic representation and the output semantic embedding. Our findings indicate that STG entails intermediate acoustic-to-semantic sound representations that neither acoustic nor semantic models can account for. These representations are compositional in nature and relevant to behavior.

9:05

5aPP4. Understanding auditory intuitive physics. Joshua H. McDermott (MIT, 77 Massachusetts Ave., 46-4065, Cambridge, MA 02139, jhm@mit.edu), Vinayak Agarwal (Mech. Eng., MIT, Cambridge, MA), Fernanda De La Torre (MIT, Cambridge, MA), and James Traer (Univ. of Iowa, Iowa City, IA)

Sound is often caused by physical interactions between objects. Humans have some ability to discern these interactions by listening. This talk will describe our lab's efforts to understand these perceptual inferences. Our research involves three inter-related approaches. First, we make physical and acoustic measurements of everyday objects and surfacesto characterize the sound of real-world objects. Second, we develop methods to synthesize sounds from physical interactions (impacts, scrapes, and rolls). Third, we use these sounds to investigate the perceptual processes and representations underlying auditory intuitive physics. The results reveal regularities in object sounds that have been internalized by humans and that are used to infer object properties and physical events from sound.

9:25

5aPP5. Categorization of sound events for automatic sound event classification. Benjamin Elizalde (Microsoft, One Microsoft Way, Redmond, WA 98052, benjaminm@microsoft.com)

To train Machine Listening models that classify sounds we need to define recognizable names, attributes, relations, and interactions that produce acoustic phenomena. In this talk, we will review examples of different types of categorizations and how they drive Machine Listening. Categorization of sounds guides the annotation processes of audio datasets and the design of models, but at the same time can limit performance and quality of expression of acoustic phenomena. Examples of categories can be simply named after the sound source or inspired by Cognition (e.g., taxonomies), Psychoacoustics (e.g., adjectives), and Psychomechanics (e.g., materials). These types of classes are often defined by one or two words. Moreover, to acoustically identify sound events we may require instead a sentence providing a description. For example, "malfunctioning escalator" versus "a repeated low-frequency scraping and rubber band snapping." In any case, we still have limited lexicalized terms in language to describe acoustic phenomena. Language determines a listener's perception and expressiveness of a perceived phenomenon. For example, the sound of water is one of the most distinguishable sounds, but how to describe it without using the word water? Despite limitations in language to describe acoustic phenomena, we should still be able to automatically recognize acoustic content in an audio signal at least as well as humans do.

9:45

5aPP6. Improving sound event detection with ontologies. Bhiksha Raj (School of Comput. Sci., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, bhiksha@cmu.edu)

Sound event recognition is the task of identifying and categorizing sounds in audio data. Automated algorithms for sound event recognition depend on having explicit models for individual sound event types to be recognized, which are trained on data tagged explicitly for those classes. The approach is data hungryand is fundamentally limited by the number of classes for which such data may be obtained. It also ignores the relationship between sounds being modeled. In this work, we attempt to address these deficiencies through the use of a human-generated sound ontology which represents sibling and parent–child relations between sound classes. We incorporate the relationships in the ontology through the design of an appropriate "loss" function (the objective function optimized to train sound-classifier models) that incorporates the relationships in the ontology, and through appropriate model update rules which utilize data from a class to update parameters (of both ontological siblings and parents). Through experiments run on the "Audioset" (a popular, large-scale dataset of 600 sound categories), we find that better-performing models can be trained for sound classes with a given dataset, and that the amount of new data required to train models for a novel sound class can be significantly reduced.

10:05–10:25 Break

10:25

5aPP7. Maximizing environmental sound recognition and speech intelligibility using time-frequency masking. Eric M. Johnson (Div. of Commun. Sci. and Disord., West Virginia Univ. School of Medicine, 375 Birch St., Morgantown, WV 26506, eric.johnson5@hsc.wvu.edu) and Eric W. Healy (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Environmental sound recognition is an essential part of the human auditory experience that not only provides a sense of connection to one's surroundings but also forecasts potential nearby safety hazards. Unfortunately, important environmental sounds can be rendered inaudible or otherwise unrecognizable by modern noise-reduction technology, leading to reduced environmental sound recognition. What is needed is a system that simultaneously provides listeners with access to audible, recognizable environmental sounds and intelligible speech. Many modern noise-reduction systems rely on some form of time-frequency masking, such as the ideal ratio mask. Restricting the output range of this mask by limiting the maximum allowable attenuation of any given time-frequency unit results in a mask that preserves environmental sounds to a certain extent while enhancing speech. In an experiment, subjects with hearing impairment and normal hearing listened to mixtures of sound + speech that had been processed by time-frequency masks with various levels of maximum attenuation, resulting in different amounts of environmental sound preservation. In a dual-task paradigm, environmental sound recognition and speech intelligibility were measured, and it was found that both types of listeners simultaneously attain high levels of performance on both tasks when the attenuation of time-frequency units is limited to 23 dB. [Work supported by NIH F32DC019314, NIH R01DC015521, and The OSU Graduate School.]

10:45

5aPP8. Environmental sound perception in adults with hearing loss and cochlear implants. Michael S. Harris (Otolaryngol. & Commun. Sci., Medical College of Wisconsin, 9200 W Wisconsin Ave., Milwaukee, WI 53226, msharris@mcw.edu), Nathan Luzum (Otolaryngol. & Commun. Sci., Medical College of Wisconsin, Milwaukee, WI), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL)

Perception of environmental sounds is crucial for safety, independence, and quality-of-life among adults with hearing loss. The objective of this study was (Experiment 1) to longitudinally assess environmental sound recognition (ESR) in a sample of postlingually deafened adults before and after receiving cochlear implants (CIs) and (Experiment 2) to cross-sectionally assess the ability of adults with CIs to identify safety-relevant environmental sounds. For Experiment 1, 20 postlingually deafened adults were tested with hearing aids on the Familiar Environmental Sound Test-Identification pre-CI and 6 months post-CI. A subset of 11 participants were also tested 12-months post-CI. Average ESR accuracy pre-CI ($M = 63.60\%$) was not significantly different from ESR accuracy 6-months ($M = 65.40\%$) or 12-months ($M = 69.09\%$) post-CI. For Experiment 2, 21 experienced adult CI users completed an ESR test consisting of 42 common environmental sounds, 28 of which were safety-relevant, along with 14 control sounds. Overall, ESR accuracy was 57% correct for safety-relevant sounds and 55% correct for control sounds. These findings suggest mediocre ESR in postlingual adult CI users for safety-relevant and other environmental sounds. Deficits in identification of environmental sounds may put CI listeners at increased safety risks and may require specific rehabilitation to improve outcomes.

Contributed Papers

11:05

5aPP9. A corpus-based approach to define Turkish soundscapes attributes. Semiha Yilmazer (Dept. of Interior Architecture and Environ. Design, Faculty of Art, Design and Architecture, Bilkent Univ., Ankara 06800, Turkey, semiha@bilkent.edu.tr), Donya Dalirnaghadeh, Ela Fasllija, Enkela Alimadhi, Zekiye Şahin, and Elif Mercan (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

Considering the challenges of translating soundscape attributes, this study aims to identify soundscape attributes in Turkish. To conduct this study, an online questionnaire with two parts was prepared and sent to 200 native Turkish bilingual speakers from all around Turkey. In the first part, the participants were asked to translate the eight soundscape attributes in ISO 12913-2:2018 from English to Turkish. In addition to direct translations, the participants were asked to describe the acoustic environment they were present in with five perceptual adjectives in their own words. The adjectives from direct translations and the open-ended question were combined to form a pool of 196 adjectives. Through a two-step elimination process by six soundscape experts, a total of 80 attributes were obtained. Twenty-four binaural sound recordings, using a 3Dio binaural microphone and Zoom H4N recorder, were collected from indoor and outdoor public spaces with different functions (e.g., concert hall, masjid, design studio, cafe). Each participant was asked to listen to six 30s-long sound recordings one at a time and rate the 80 attributes on a five-point scale for each recording individually. The results show that the opposite adjectives with similar perceptual meanings were placed together in two main clusters.

11:20

5aPP10. Observed changes in the perceived ambience of public spaces elicited by added sound. Andre Fiebig (Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, andre.fiebig@tu-berlin.de) and Cleopatra Moshona (Eng. Acoust., TU Berlin, Berlin, Germany)

New soundscape approaches consider plus sound design to create a more positive atmosphere and to reduce the audibility of unwanted sound sources. By adding water installations, sound art or simply music for example, urban noise can be efficiently masked to a certain degree. Although this design approach appears reasonable, scarce amount of data is available regarding the potential and limitations of additive sound measures in public spaces. In this context, the enrichment of the acoustic environment by music is frequently discussed and mainly applied in indoor settings. However, such measures raise ethical concerns when it comes to subconscious effects, because they might lead to behavioral changes. In Berlin, a variety of music pieces and genres were played in public transportation waiting areas to investigate the effect of the music on the atmosphere and passengers. In order to examine the effects of the music on different assessment parameters (e.g., perceived safety, cleanliness, mood, loudness) varying data collection tools were applied insitu to investigate the implications of additive sound design based on music streams in detail. The paper presents survey results with a special focus on how the perception of the soundscapes is changed by the added sound.

11:35–11:55
Panel Discussion

Session 5aSC

Speech Communication: Speech Production II: Speech Articulation | Sociophonetics (Poster Session)

Yoonjeong Lee, Chair
Linguistics, University of Michigan, 611 Tappan Street, 421 Lorch Hall, Ann Arbor, MI 48109-1220

All posters will be on display from 8:00 a.m. to 12:00 noon. Authors of odd-numbered abstracts will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even-numbered abstracts will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

5aSC1. Domain adaptation towards speaker-independent ultrasound tongue imaging based articulatory-to-acoustic conversion. Kang You (Tongji Univ., Shanghai, China), Kele Xu (National Key Lab. of Parallel and Distributed Processing (PDL), 107, Yanwachi, Changsha 410073, China, kelele.xu@Gmail.com), Jilong Wang, and Ming Feng (Tongji Univ., Shanghai, China)

In this paper, we endeavor to address an articulatory-to-acoustic issue which aims to estimate the mel-spectrogram of the acoustical signals, using midsagittal ultrasound tongue images of the vocal tract as input. Previous attempts employed statistical methods for the inversion between the articulatory movements and speech, while deep learning has begun to dominate this field. Despite the sustainable efforts that have been made, the mapping performance can be greatly varied for different speakers and most of the previous methods are constrained for the speaker-dependent scenario. Here, we present a novel approach towards speaker-independent mapping, which is inspired by the domain adaptation method. Specifically, we explore decoupling the spectrogram generation task and the speaker recognition task. Leveraging a novel designed loss function, we can improve the performance under the speaker-independent scenarios, through the adversarial learning strategy. To demonstrate the effectiveness of the proposed method, extensive experiments are conducted on the Tongue and Lips (TaL) corpus. Objective evaluation is conducted to compare the generated spectrograms and ground truth, using three evaluation metrics, including the MSE, SSIM, and CW-SSIM. The results indicate that our proposed method can achieve superior performance under the speaker-independent scenario, compared with competitive solutions. Our code is available at https://github.com/xianyi11/Articulatory-to-Acoustic-with-Domain-Adaptation.

5aSC2. Development of a normative baseline for speech stability in lingual articulation. Stefan A. Frisch (Rehabilitation Sci., Appalachian State Univ., Leon Levine Hall, 1179 State Farm Rd., ASU Box 32165, Boone, NC 28608-2165, frischsa@appstate.edu)

In typical speech production, speakers usually use the same articulatory gesture for a particular speech sound in a particular context (such as a /k/ sound after an /i/ vowel). However, since people are not machines, there is inherent variation in how precisely a speech gesture is repeated. Previous research on consistency of tongue articulation has found individual variation in articulatory consistency in tongue posture that may also provide insight into communication disorders, such as stuttering (Frisch *et al*., 2016, *Clinical Linguistics and Phonetics*). This presentation will report on pilot data for the development a normative measure for speech stability to establish how much variation is typical for different age groups and whether the complexity of speech materials influences the articulatory measure of speech stability. Included in the pilot methodology are (1) a wide variety of lingual consonants, (2) variation in vowel context, and (3) evaluation of a perceptual proxy measure for speech stability that could be practically used in a clinical environment.

5aSC3. Musicality measures and generalization of an artificial dialect of English. Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, 390 S 1530 E, Rm. 1218, Salt Lake City, UT 84112, liz.d.young@utah.edu) and Brett R. Myers (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

Wide individual variability exists in the ability to imitate and generalize novel speech and voice patterns, including unfamiliar dialects. Limited research exists examining possible underlying causes of such variability, particularly in the ability to generalized novel dialects. One possible source of individual differences is variability in musicality, either in terms of musical ability or degree of musical training. The current study examines the relationship between musicality and an individual's ability to generalize an artificial dialect of English to a novel context with minimal exposure. Thirty participants were tested, 15 self-classifying as non-musicians and 15 as musicians. All participants were given Gordon's Advanced Measures of Music Audiation (AMMA; a musical aptitude test) and the Goldsmith's Musical Sophistication Index (Gold-MSI, a survey of an individual's engagement with music). Participants then completed a dialect learning task using an artificial dialect of English (Spinu *et al*., 2020) in which they are first familiarized with the dialect, then asked to imitate it, then finally generalize it to novel sentences. The effect of both musicality measures on dialect generalization, as well as patterns of individual differences, will be examined.

5aSC4. Articulation differences of /s/ observed using inverse tongue atlas modeling. Ursa Maity (School of Biomedical Eng., Univ. of BC, 2366 Main Mall, Vancouver, BC V6T1Z4, Canada, ursa.maity@ubc.ca), Fangxu Xing (Radiology, Harvard Med. School, Boston, MA), Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD), Jiachen Zhuo (School of Medicine, Univ. of Maryland, Baltimore, MD), Georges El Fakhri (Radiology, Harvard Med. School, Boston, MA), Jerry L. Prince (Elec. and Comput. Eng., John Hopkins Univ., Baltimore, MD), Jonghye Woo (Radiology, Harvard Med. School, Boston, MA), and Sidney Fels (School of Biomedical Eng., Univ. of BC, Vancouver, BC, Canada)

Depending on the vowel context, the consonant /s/ can be articulated by engaging different functional units of the tongue. An inverse finite element (FE) tongue model comprising hexahedral elements generated from a 4D statistical MRI atlas of 22 speakers performing the speech tasks "a geese" (tongue moving forwards) and "a souk" (tongue moving backwards) is used to study this articulatory behavior. The model uses a state-of-the-art inverse tracking controller to simulate the motion of internal tissue points of the different speakers deformed into the atlas space. Motion tracking is successfully carried out by minimizing the *L2-norm* of velocity error of FEM nodes using the Cottle–Dantzig algorithm. The results show that for "a-geese," the utterance of /s/ in context of vowel /i/ showed increase in activation of tongue protruder muscles such as genioglossus posterior, floor muscle geniohyoid by ~5% more than in the case of "a-souk." In "a-souk" relative activity increased in the tongue retractor muscles mid- and anterior-genioglossus, and superior longitudinal by ~4%. Our findings are consistent with subject-specific state-of-the-art models

and with articulatory expectations. The inverse atlas tongue model can be further used to estimate such articulation behavior on extended datasets of subjects performing more variations of speech tasks.

5aSC5. Gender expression in productions of /s/ and /ʃ/. Nichole Houle (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 665 Broadway, New York, NY 10012, nh1473@nyu.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

The spectral features of /s/ and /ʃ/ carry important sociophonetic information regarding a speaker's gender. Gender is defined as man or woman, but this excludes people who identify as trans and gender non-conforming. In this study, we use a more expansive definition of gender to investigate the acoustics (duration and spectral moments) of /s/ and /ʃ/ across cis men, cis women, and transfeminine speakers in voiced and whispered speech. Additionally, we investigated the relationship between spectral measures and transfeminine gender expression. We examined /s/ and /ʃ/ productions in words from 30 speakers (10 cis men, 14 cis women, and 6 transfeminine) and 31 speakers (10 cis men, 13 cis women, and 8 transfeminine), respectively. In general, fricative center of gravity was highest in productions by cis women, followed by transfeminine, and then cis men speakers. Gender differences were found for /s/, but not /ʃ/. Cis women speakers produced /s/ with greater negative skew than cis men speakers. Transfeminine speakers did not differ from either group. Within transfeminine speakers, /s/ and /ʃ/ center of gravity were related to a speaker's gender related voice concerns. Taken together, /s/ and /ʃ/ differed by gender and may be related to transfeminine vocal concerns.

5aSC6. The style effect on the TRAP phonemic split in UP English. Wil A. Rankinen (Commun. Sci. and Disord., Grand Valley State Univ., 500 Lafayette Ave. NE, 303D, Grand Rapids, MI 49503-3360, wil.rankinen@gvsu.edu), Payton Westedt (Commun. Sci. and Disord., Grand Valley State Univ., Jenison, MI), and Lauren McKenzie (Commun. Sci. and Disord., Grand Valley State Univ., Grand Rapids, MI)

The short-a system, or the phonemic split of the TRAP vowel (i.e., /æ/), refers to the raised position of the vowel in pre-nasal and pre-velar phonological environments (Labov *et al*., 2006; Labov, 2007; Becker *et al*., 2010; Labov *et al*., 2015; Wagner *et al*., 2016). The present study builds upon these previous studies by examining the TRAP-system in rural, under-documented American English speech communities across Michigan's Upper Peninsula (UP). Previous work on this feature in reading passage data has revealed UP speech communities on the east- and west-sides of the peninsula have strikingly different systems (i.e., a two-way phonemic split in eastern UP, while a three-way split in western UP). The present sociophonetic talk examines the apparent-time change of the phonemic split of an 87-speaker sample as an interaction of task types (passage/word list/minimal pairs), area (east/west), age groups (18-39/40-59/60 +), and self-gender (female/male). The three-way phonemic split of TRAP only occurs in informal speech styles in western UP; in contrast, the two-phonemic split is preferred in eastern UP irrespective of speech style. The two-way phonemic split is preferred in more formal speech styles and is stable in the eastern UP suggesting an external influence from outside Michigan's Upper Peninsula.

5aSC7. Exploring macro-rhythm in African American English. A. C. Rochelle Thomas (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, acrthomas@u.northwestern.edu) and Jennifer Cole (Linguist, Northwestern Univ., Evanston, IL)

This study contributes to a growing body of research on quantifying the theory of macro-rhythm, a model of prosodic typology proposed by Sun-Ah Jun (2014) that looks at the global pitch patterns of an utterance, through comparing African American English (AAE) to Mainstream (white) US English (MUSE). Based on prior research on intonation, rhythm, and prosody in AAE (cf. Thomas, 2015 for a thorough overview), we predicted that AAE would have stronger macro-rhythm than MUSE, such that speakers would produce higher peaks and lower valleys, pitch excursions of greater magnitude, have a greater consistency in slope, a greater frequency of high (H) and low (L) pitch targets, and more regular spacing of these pitch targets. Data weretaken from the Valdosta group of the Corpus of Regional African American Language (CORAAL) for Black speakers and the Buckeye corpus for white speakers. Contrary to our expectations, Buckeye speakers proved to be more macro-rhythmic than their CORAAL counterparts in all but one metric, but these results confirm that macro-rhythm is a viable method for cross-dialectal analysis as it reveals quantifiable differences between speaker dialect groups of the same language.

5aSC8. Auditory feedback control of inter-articulator speech coordination: Evidence from tongue and jaw movements. Matthew Masapollo (Univ. of Florida, 1225 Ctr. Dr., Gainesville, FL 32610, mmasapollo@phhp.ufl.edu), Grant Oberle, Kayleigh Burge, Abigail Lebedeker, Nicholas Salazar, Kiana Ferdowsiepour, Ana Rodriguez, and Susan Nittrouer (Univ. of Florida, Gainesville, FL)

In order to test whether and how immediate auditory feedback is involved in the coordinated action of sets of speech articulators, the current research quantified changes in the temporal and spatiotemporal relations between jaw and tongue tip movements in response to noise masking. Normal-hearing talkers recorded /tV#Cat/ utterances using electromagnetic articulography, with alternative V (/ɑ/-/ɛ) and C (/t/-/d/), across variation in production rate (fast-slow) and stress (first syllable stressed-unstressed). Approximately 240 utterances were produced in two conditions: normal listening and auditory feedback masking. Two kinematic measures were obtained: (1) timing of tongue-tip raising onset for medial C, relative to jaw opening-closing; and (2) angle of tongue-tip raising onset, relative to the jaw phase plane. In the normal listening condition, any manipulation that shortened the jaw opening-closing cycle reduced both the relative timing and phase angle of tongue-tip movement onset. In the masking condition, specific changes were observed in how consistently the phase angle of tongue-tip movement onset scaled with jaw opening–closing across rate and stress variation, but not in the relative timing of tongue-tip movement onset. Collectively, these findings suggest that the spatiotemporal phasing between articulator movements relies more on immediate auditory feedback than the relative timing between ongoing articulator movements.

5aSC9. Acoustic distance effect on the perception of sibilants mergers between retroflexes and alveolars in Taiwan Mandarin. Baichen Du (Dept. of Linguist, The Univ. of Hong Kong, 9F Run Run Shaw Tower, Centennial Campus, Pokfulam Hong Kong, baichen@connect.hku.hk)

Studies in speech production and perception have found a tight link between the two processes. Sound properties in speakers' production repertoire strongly influence their perceptual spaceand exposure to another variant also shifts speech patterns in shadowing tasks. This suggests that diachronic sound change might be a result of it. Therefore, we tested whether categorical perception of sibilants is influenced by acoustic distance in the production of ongoing sibilants mergers in Taiwan Mandarin, where retroflexes are merging towards alveolars. Twenty-three native speakers produced 46 mono- or disyllabic Tone 1 words with onsets of /ts/, /ʈʂ/, /tsʰ/, /ʈʂʰ/, /s/, and /ʂ/, listened to 23 eight-step continuums of synthesized sibilants from retroflexes to alveolars pre-appended to vowels and codas, and answered a sociolinguistic questionnaire. As predicted, closer acoustic distances between categories predicted more high-CoG stimuli to be identified as retroflexes (more tolerance for high-frequency retroflexes). Step, sound, and social effects were significant. Sibilants' frequencies were lower than previously reported. Overall, this suggests that sibilant mergers diffuse to speakers with closer acoustic distances first, and the categories will collapse on language level in production before perception. They spread among speakers asynchronously and variably, depending on dialectal background and social attitude as well.

5aSC10. Self-made voice dosimeter. Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu) and Charles J. Nudelman (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Voice dosimeters are useful tools for gathering voice production data in the daily lives of individuals with voice disorders. In this tutorial, instructions for a low-cost (less than $100), easy-to-assemble voice dosimeter are

provided. This do-it-yourself (DIY) voice dosimeter is validated based on performance results. The DIY voice dosimeter was worn by 10 vocally healthy participants. The participants produced a sustained /a/ vowel and then read a text with three different vocal efforts. These tasks were recorded by the DIY voice dosimeter and a reference microphone simultaneously. The expanded uncertainty of the mean error in the estimation of these two quantities as measured by the DIY dosimeter was performed by simultaneously comparing signals acquired through the reference microphone and the dosimeter themselves. For measures of sound pressure level, the DIY voice dosimeter had a mean error of -0.68 dB with an uncertainty of 0.56 dB. For fundamental frequency, the mean error was 1.56 Hz for female participants and 1.11 Hz for male participants, with an uncertainty of 0.62 Hz and 0.34 Hz for females and males, respectively. The mean error and uncertainties for the DIY voice dosimeter are comparable to the most accurate voice dosimeters that were previously on the market.

5aSC11. Exploring speaker-specific behaviors and effects of electro-optical stomatography palates during speech breathing. Laura Koenig (Adelphi Univ., 300 George St., New Haven, NY 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Leibniz Ctr. for General Linguist, Berlin, Germany)

Speech inhalations, compared to vegetative breathing, are more likely to be associated with open oral postures, i.e., speakers may employ oral as well as nasal inspiration patterns. Past studies have suggested that there may be considerable individual variation for speech breathing behaviors as well. Recently, we have employed electro-optical stomatography (EOS) to assess lip apertures during speech breathing in healthy German-speaking women. The EOS system combines traditional electropalatography (contact sensors) with optical sensors that register distances. Consistent with past work, preliminary data on lip apertures showed that speakers frequently had open-mouth postures during speech breathing. However, the lip sensors of the EOS device, and the full coverage of the upper dentition that it involves, could have induced at least some speakers to adopt an open-mouth posture to a greater degree than usual. Thus, to explore the external validity of our data, this work will compare lip postures for speech breathing in the same speakers with and without the EOS device. We will also assess to what extent oral postures during speech inspirations vary across individuals, across normal and loud speaking conditions, and across speaking tasks that vary in their degree of naturalness.

5aSC12. Rhoticity in Black Boston: Examining the effects of ethnicity and ethnic orientation. Malachi Henry (Dept. of Speech, Lang., & Hearing Sci., Indiana Univ., 1579 S Renwick Blvd, Bloomington, IN 47401, mjhenry5519@gmail.com), Amalia Robinson (Dept. of Linguist, Indiana Univ., Bloomington, IN), Xiao Dong (Dept. of East Asian Lang. and Cultures, Indiana Univ., Bloomington, IN), Jeremy Miller (Dept. of Spanish and Portuguese, Indiana Univ., Bloomington, IN), Clara Miller-Broomfield (Dept. of French & Italian, Indiana Univ., Bloomington, IN), Erika Sosa (Dept. of Spanish and Portuguese, Indiana Univ., Bloomington, IN), and Monica Nesbitt (Dept. of Linguist, Indiana Univ., Bloomington, IN)

Recent studies have sought to more thoroughly examine rhoticity among non-white Bostonians (Nagy andIrwin, 2010; Browne andStanford, 2018), as the city becomes more racially and ethnically diverse. We build upon Browne and Stanford (2018), which found that Black Bostonians (African American [AA] and Caribbean American [CA]) were more r-ful than White Bostonians. We seek to account for variation in this speech community by considering the impact of ethnicity and the emic measure of ethnic orientation (Hoffman andWalker, 2010) on rhoticity in Black Boston. Six CA and 18 AA Bostonians' /r/ productions ($n = 2018$) were gathered from sociolinguistic interviews conducted as part of the Eastern Massachusetts Life and Language Project (Nesbitt andWatts, 2022), in addition to one CA and six AA participants from Browne and Stanford's (2018) dataset. Linear mixed-effects modeling revealed significant effects of ethnicity, such that Caribbean Americans (95% r-ful) are almost categorically r-ful ($p\text{-value} < 0.001$), while African Americans are more variable (10%–100% r-ful). Furthermore, ethnic orientation was a significant predictor of rhoticity ($p\text{-value} < 0.001$); among African Americans, high ethnic orientation led to more r-fulness. Rather than ethnic identification alone, factors such as social networks, migration history, and language attitudes influence rhoticity in Black Boston.

5aSC13. The role of the tongue back on contrasting voicing properties of English coda consonants: Evidence from an ultrasound imaging technique. Daejin Kim (Linguist, Univ. of New Mexico, 1 University of New Mexico, MSC03 2130, Albuquerque, NM 87131-0001, daejinkim@unm.edu)

This study explores how voicing properties of stop consonants at the coda of a syllable are reflected in the articulation of the tongue back in American English. The association between the laryngeal control of the voicing property of consonants and the positioning of the tongue back has been increasingly reported (e.g., Ahn, 2018; Coretta, 2020). A recent study (Coretta, 2020) reported that longer vowel duration indicating the following voiced stop consonant corresponds to greater tongue root advancement at the offset of the vowel in Italian and Polish. Since the vowel with a following voiced stop consonant has a longer duration than that with a voiceless stop consonant, the later offset of TB movement should be correlated with the offset of vocal folds vibration in American English as well. The preliminary result suggests that the later offset of TB advancement is positively correlated with the longer vowel duration of voiced consonants at the coda of a syllable; therefore, contrasting voicing property of coda consonants in English is phonological and phonetically evidenced in the articulation of the tongue back. Articulatory data from ultrasound images of the tongue from three native speakers of American English will be presented.

5aSC14. Limited attention shifts judgments in socioindexical perception. Jory P. Ross (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, ross.1589@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

This project investigated how limited attention influences the perception of socially meaningful details in speech with a focus on American English ING. American English ING has two variants, a canonically standard velar variant *-ing*, as in *cooking*, and a non-standard apical variant *-in'*, as in *cookin'*. Talkers who more frequently use the apical variant are judged to be less educated, less articulate, less formal, and more likely to be Southern. In a matched guise task talkers produced apical variants, velar variants, or a mix of the two. Participants rated the talkers on intelligence, clarity, dynamism, friendliness, and gender. In a distracted condition, participants completed visual search tasks while listening to limit their ability to attend to the talkers' speech. In a focused condition, participants listened with no concurrent task. The results revealed that apical talkers were rated as less cold and less articulate than velar talkers. Ratings for coldness, but not articulateness, were less affected by ING guise for distracted participants than focused participants. Additionally, ratings for articulateness and gender shifted between distracted and focused conditions independent of ING guise. These findings suggest that limited attention alters how listeners form judgments from indexical information in speech.

5aSC15. Effects of dialect familiarity on lexical processing dynamics. Kyler B. Laycock (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, laycock.21@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

When listeners map acoustically ambiguous forms to abstract lexical representations such that there is a potential for greater lexical competition, there is observable variation in listeners' lexical processing dynamics. As compared to non-mobile listeners, geographically mobile listeners exhibit less inhibition given competing prime-target pairsas well as less facilitation from matching prime-target pairs. This difference is attributed to mobile listeners' exposure to cross-dialect ambiguities. The current study seeks to more closely examine the role of geographic mobility and dialect-specific familiarity, and further address lexical processing dynamics by manipulating inter-stimulus-intervals (ISI). A lexical decision task using cross-modal priming and cross-dialect phonological contrasts was used to assess effects of facilitation and inhibition. Analyses show an interaction between familiarity, vowel contrast, and ISI as predictors of response time. While facilitatory effects increase and inhibitory effects decrease when participants have

higher familiarity with a contrast, high familiarity and longer ISI also increase the degree of individual variability. These results support predictions that processing differences are dependent upon familiarity with particular variants as opposed to a general strategic response, though it may be the case that a longer ISI allows for additional variability in processing and the adoption of general strategic responses to emerge.

5aSC16. Granger causality analysis of internal tongue muscles in tongue protrusion from diffusion and tagged MRI. Hyeonjeong Park (MGH/Harvard, Boston, MA), Fangxu Xing (Radiology, Harvard Med. School, Boston, MA), Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD), Xiaofeng Liu (Gordon Ctr. for Medical Imaging, MGH and Harvard Med. School, Boston, MA), Jiachen Zhuo (Univ. of Maryland, Baltimore, Baltimore, MD), Sidney Fels (Univ. of BC, Vancouver, BC, Canada), Tim Reese, Van Wedeen, Georges El Fakhri (MGH/Harvard, Boston, MA), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), and Jonghye Woo (MGH/Harvard, Boston, MA, jwoo@mgh.harvard.edu)

The human tongue is a muscular hydrostat, comprising a group of orthogonally oriented and interdigitated muscles without skeletal support. Key characteristics of the tongue as a muscular hydrostat include that the tongue's volume is constant, and any decrease in one dimension will be compensated by an increase in another dimension. This work aims to identify causal relationships among internal muscles in protrusive muscle contractions, which is vital to understand their underlying muscle coordination patterns. We investigated four muscles known to contribute to tongue protrusion, including transverse, verticalis, superior longitudinal, and genioglossus muscles. To capture tongue structure and function, we acquired diffusion MRI and tagged MRI to estimate the muscle fiber geometry of the four muscles and the motion patterns of tongue protrusion, respectively. We first computed Lagrangian strains along muscle fiber direction in each voxel, reflecting muscle fiber shortening. Average strain values in each muscle, defined by a vocal tract atlas, were then input into a linear vector autoregression Granger causality model, to investigate their causal relationships. Our findings on the strain patterns over time as well as the pairwise causal relationships were consistent in part with the muscular hydrostat theory, which helps elucidate the tongue muscle coordination patterns.

5aSC17. Breathing patterns of conversation partners. Megan Diekhoff (Speech, Lang. and Hearing Sci., Indiana Univ., Indiana University, Bloomington, IN 47401, meredieк@indiana.edu) and Steven M. Lulich (Speech & Hearing Sci., Indiana Univ., Bloomington, IN)

Previous research has shown that breathing patterns of conversation partners can become synchronized. For example, when one conversation partner is speaking, the listening partner's breathing patterns more closely resemble speech breathing than tidal breathing (McFarland, 2001). The present study aims to replicate and extend these findings using an unconventional instrumental setup consisting of a single inductive plethysmograph machine shared by both conversation partners. In addition to scripted and unscripted conversational speech, single-subject quiet breathing and speech breathing tasks of increasing length and complexity were also recorded using conventional inductive plethysmography, and used as baseline data for interpretating the conversational speech breathing data.

5aSC18. Does the articulatory modulation function encode stress? Jessica Campbell (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, jac95339@usc.edu), Dani Byrd, and Louis Goldstein (Linguist, Univ. of Southern California, Los Angeles, CA)

The articulatory modulation function, a signal quantifying global change in vocal tract posture over time, is highly rhythmic, with limited variability within and across speakers. It is unclear, however, if the stability and periodicity of this signal, whose peaks ("pulses") occur at a frequency of $\sim$8 Hz, is partially driven by quasi-oscillatory linguistic phenomena such as lexical stress patterning. In the present study, articulatory point-tracking data (XRMB) of connected speech for multiple subjects is used to assess the effect of linguistic stress on the spatiotemporal patterning of the vocal tract modulation function to determine if the patterning of acoustic vowel onsets—both stressed and in general—is encoded in the length of modulation function's inter-pulse intervals and/or in the magnitude (velocity) of the pulses. The results show no correlation between inter-pulse intervals and stress. However, stress is encoded as higher velocity magnitudes in the pulses immediately following acoustic vowel onsets in stressed syllables (compared to unstressed ones)and additionally encoded in higher magnitude than the immediately preceding pulse. While the temporal stability of the modulation function does not appear to be related to the general regularity of stressed syllables in English, the signal does retain linguistic stress information in modulation pulse magnitude.

5aSC19. Effects of dialect familiarity on word identification in a visual-world eye-tracking task. Marie Bissell (Ohio State Univ., Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, bissell.43@osu.edu), Larisa Bryan (Ohio State Univ., Columbus, OH), Precious-Janae Romain (Rutgers Univ., Columbus, OH), and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

The acoustic closeness of vowels across American English dialects results in cross-dialect lexical competition among minimal pair words. In the current study, we investigated how listener familiarity with specific dialects affects performance in a cross-dialect visual-world eye-tracking task. Listeners heard both acoustically close and control minimal pairs in each of three regional American English dialects (New England, Southern, and Northern) in a four-alternative forced-choice identification task. Listener dialect familiarity was assessed for each dialect based on each individual's self-reported residential history for themselves, their parents, and their significant others. Preliminary analyses revealed more accurate and faster responses to New England talkers as familiarity with the New England dialect increased, consistent with talker and accent familiarity effects in the literature. However, responses to Southern talkers were slower as familiarity with the Southern dialect increased, contrary to the typical pattern. Responses to Northern talkers were not affected by familiarity with the Northern dialect, consistent with the lack of enregisterment of this variety. Together, these findings suggest that listener dialect familiarity has different effects on the resolution of cross-dialect lexical competition, reflecting both the nature of the variation and the social stereotypes associated with the dialect. Analysis of the eye-tracking data is ongoing.

5aSC20. Classification of Kansai and Chugoku dialects spoken by old Japanese speakers. Kota Hattori (Faculty of Integrated Arts and Sci., Tokushima Univ., 2-24, Shinkura-cho, Tokushima-shi, Tokushima 770-8501, Japan, kota@tokushima-u.ac.jp) and Shinsuke Kishie (Dept. of Japanese Lit., Nara Univ., Nara-shi, Nara, Japan)

The present study examined how Kansai and Chugoku dialects in Japanese are classified using hierarchical density-based spatial clustering (HDBSCAN) and Random Forest (RF). We obtained the written-format pronunciations of 48 words from 1450 Japanese speakers over age 50 who had been residing in their birthplaces. We calculated phonetic distance (ALINE distance) between the dialectal pronunciations and standard Japanese ones, and ran HDBSCAN and RF models with 1000 bootstrap samples. The optimal HDBSCAN model demonstrated that there are two groups of speakers in the northern and southern pastoral areas of Kansai region. The RF models demonstrated that speakers from each prefecture were classified with a wide range of accuracies ($F1 = 0.73$). Kansai-region speakers in the urban area where Osaka, Kyoto, and Nara prefectures share borderlines were poorly classified. Similarly, Chugoku-region speakers living near the borderlines of Hiroshima, Shimane, and Tottori prefectures were poorly classified. The rest of the participants were generally classified well. These results suggest that each prefecture generally has its own dialect, but its distribution goes beyond prefecture borderlines. This is the first study to reveal such classification patterns of Japanese dialects using machine learning approaches.

5aSC21. Evaluating normalization accounts against the dense vowel space of Stockholm Swedish. Anna Persson (Swedish Lang. and Multilingualism, Stockholm Univ., Universitetsvägen 10D, Stockholm 10691, Sweden, anna.persson@su.se) and T. Florian Jaeger (Brain and Cognit. Sci. and Comput. Sci., Univ. of Rochester, Rochester, NY)

Talkers vary in the phonetic realization of their vowels. One influential hypothesis holds that listeners overcome this inter-talker variability through pre-linguistic auditory mechanisms that normalize the acoustic or phonetic cues that form the input to speech recognition. Dozens of competing normalization accounts exist —including both vowel-specific (e.g., Lobanov, 1971; Nearey, 1978; Syrdal and Gopal, 1986) and general-purpose accounts applicable to any type of phonetic cue (McMurray and Jongman, 2011). We add to the cross-linguistic literature by comparing normalization accounts against a new database of Swedish, a language with a particularly dense vowel inventory of 21 vowels differing in quality and quantity. We train Bayesian ideal observers (IOs) on unnormalized or normalized vowel data under different assumptions about the relevant cues to vowel identity (F0-F3, vowel duration), and evaluate their performance in predicting the category intended by talker. The results indicate that the best-performing normalization accounts centered and/or scaled formants by talker (e.g., Lobanov), replicating previous findings for other languages with less dense vowel spaces. The relative advantage of Lobanov decreased when including additional cues, indicating that simple centering relative to the talker's mean might be sufficient to achieve robust inter-talker perception (e.g., C-CuRE).

5aSC22. Hadza liquid production and acoustics. Jeremy R. Coburn (Linguist, Indiana Univ., Ballantine Hall 862, 1020 E. Kirkwood Ave., Bloomington, IN 47405-7005, jecoburn@iu.edu) and Sherman D. Charles (Speech & Hearing Sci. and Linguist, Indiana Univ., Bloomington, IN)

Liquid systems in the world's languages are relatively poorly understood. This study employs 3D/4D ultrasound methods to investigate articulatory characteristics of Hadza liquids produced by a single male speaker in a laboratory, analyzing simultaneously collected acoustic signals to determine relations between articulation and acoustics. Articulatory-acoustic relations are then compared to acoustic data obtained from other speakers collected in the field. Hadza is a language isolate spoken by approximately 1200 people (Brian Wood, pc.) in north-central Tanzania. Hadza is reported to have a single liquid phoneme which varies allophonically between a lateral approximant [l] occurring word-initially, and a tap [ɾ], occurring intervocalically (Tucker *et al.*, 1977; Sands *et al.*, 1996; Sands, 2013). The results of the current study confirm the production of both lateral approximant [l] and tap [ɾ] allophones; however, the results also show substantial variation between speakers. Furthermore, some speakers exhibit a categorical distinction between the allophones in given phonological environments, while others indicate a very similar articulation in both, and demonstrate no such distinction.

5aSC23. Laryngeal contrasts in Hadza occlusives. Jeremy R. Coburn (Linguist, Indiana Univ., Ballantine Hall 862, 1020 E. Kirkwood Ave., Bloomington, IN 47405-7005, jecoburn@iu.edu)

Laryngeal contrasts in consonants are found in nearly all languages. Two-way laryngeal contrasts are common in western European languages, and three- and four-way systems are attested, for example, in many South Asian languages. The current study presents acoustic data on laryngeal contrasts in Hadza, a language isolate spoken by ∼1200 people (Brian Wood, pc.) in Tanzania. Hadza contains multiple complex consonant types, including clicks and ejectives, as well as aspiration and prenasalization. Some previous descriptions suggested a three-way contrast in stops (e.g., /t t^{h} d/) and affricates (e.g., t͡s t͡s^{h} d͡z), and also a two-way contrast in clicks (e.g., /! !^{h}/) (Tucker *et al.*, 1977; de Voogt, 1992), but did not offer corroborating phonetic evidence. Sands *et al.* (1996) documented the expected distinction in Voice Onset Time (VOT) between voiceless unaspirated and aspirated stops, but they did not find VOT differences for clicks or affricates. Using wordlist data collected in 2022, this study presents temporal measurements, including VOT, to examine the phonetic properties of laryngeal contrasts in Hadza occlusives and, especially, to evince an aspiration contrast in affricates and clicks previously unsubstantiated in the literature. These results resolve discrepancies between past research and further elucidate the laryngeal system of Hadza.

5aSC24. Revisiting listener evaluations of 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)

In U.S. English, nonpathological creaky voice is a common and natural voice quality that contributes meaningful linguistic information. However, it is also a target of gender-based prejudice, with studies suggesting that speakers with creak sound less hirable and less pleasant. Unfortunately, the stimuli in these studies were often problematic (e.g., creak present in both creaky and modal stimuli; unusually high proportions of creak). To address these problems, we carefully selected naturallyproduced, low predictability sentences. Half of these contained creak on the final word (a common location for creak in both men and women speakers) and half did not. Listeners rated half of the sentences on a hirability scale and half on a pleasantness scale. They then completed an implicit association task regarding gender and career roles. Consistent with previous research, sentences with creak were rated as being produced by speakers who were less hirable than those without creak. Similar to previous findings, older listeners rated stimuli with creak as less pleasant than stimuli without creak. Implicit association scores did not predict ratings on either rating scale. Contrary to our expectations, less favorable evaluations of creak were found despite using naturallyproduced sentences with only sentence-final creak.

5aSC25. Studying the glottal vibration onset and offset using laryngeal high-speed videoendoscopy in connected speech. Maryam Naghibolhosseini (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, naghib@msu.edu), Trent M. Henry (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Stephanie R. Zacharias (Head and Neck Regenerative Medicine Program, Mayo Clinic, Scottsdale, AZ), and Dimitar Deliyski (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Laryngeal high-speed videoendoscopy (HSV) is a powerful tool to study the detailed vibratory characteristics of the vocal folds. The use of HSV in connected speech enables us to study the transitory behaviors of the vocal folds during different vocalizations. The onset and offset of voicing contain critical information about the health and function of the vocal folds. In this work, HSV is used to study how the voicing begins/ends at the laryngeal level at the onset/offset of phonation. HSV data were obtained during production of connected speech from participants with non-pathological voices and patients with different neurogenic voice disorders. The data collection was performed using a Photron FASTCAM mini AX200 high-speed camera coupled with a flexible nasolaryngoscope. The glottal attack time and glottal offset time were measured at the onset and offset of different vocalizations, respectively, from the HSV data. The findings show differences in the onset and offset measures between the two groups of participants. The vocal fold vibratory behaviors will be discussed qualitatively and quantitatively using these measures. The use of HSV in connected speech could help us develop accurate voice assessment procedures to assess the vocal folds function during connected speech.

5aSC26. Durational variations in Raleigh and Ocracoke vowel production: Methods for analyzing undershoot. Griffin Lowry (English, North Carolina State Univ., 2211 Hillsborough St., 210B, Raleigh, NC 27607, jglowry@ncsu.edu)

Dialectal and generational differences in undershoot between speakers from Raleigh and Ocracoke, North Carolina, are examined. Undershoot occurs when a speaker fails to reach the target formant values of a certain sound, especially for vowels. While this is commonly due to duration, other factors can cause undershoot (coarticulation, stress, speaking style). Through comparing these speakers, this paper aims to evaluate new and old ways of analyzing undershoot, including an examination of formant ranges and calculations of absolute undershoot values. Formant ranges are the difference between the maximum and minimum frequency values of a particular formant throughout the duration of the vowel. Absolute undershoot is a formula developed by Oh (2007, *J. Phon.* 36:361–384) that attempts to quantify the amount of undershoot a speaker shows in order to conduct inter- and intra-speaker comparisons. Furthermore, analysis of undershoot may illuminate causes of sound changes. For example, the older speaker from Raleigh showed undershoot in /u/—longer durations resulted in backer realizations. The younger Raleigh speaker, conversely, showed no undershoot but consistent /u/-fronting. /u/-fronting possibly began with undershoot in one generation, and then became a hypocorrected norm in the next generation.

5aSC27. Applying machine learning algorithms for stuttering detection. Piotr Filipowcz (Faculty of ETI, Gdansk Univ. of Technol., Gdansk, Poland, Poland) and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

This work aims to develop algorithms designed to detect stuttering in speech signals effectively. The background of this speech disorder is given with particular attention paid to its subcategories, i.e., extensions, blocks, sound repetitions, word repetitions, and interjections. A review of state-of-the-art machine learning-based approaches to detect and classify stuttering is presented. A deep model capable of detecting stuttering in the speech signal at the f1 measure level of 0.93 for the general class is built. Also, two baseline algorithms, i.e., Support vector machine (SVM) and k-nearest neighbors (kNN), are implemented to compare their efficiency with a residual neural network outcome. In addition, a series of experiments are conducted to study the impact of elements such as model architecture, data partitioning, amount of data, or use of individual features. The results obtained for the different categories of stuttering differ significantly due to the size of each subclass in the datasets used for the study and the annotation quality.

5aSC28. Using the Corpus of North American Spoken English to explore regional variation in millions of eɪ diphthongs. Steven Coats (English, Faculty of Humanities, Univ. of Oulu, Oulu 90014, Finland, steven.coats@oulu.fi)

Increases in the quality of automatic speech recognition (ASR) systems and forced alignment algorithms have opened up new horizons for the study of acoustic properties of naturalistic speech (Coto-Solano *et al.*, 2021). This paper describes the procedures used for the automatic extraction and alignment of audio and ASR transcript excerpts containing eɪ diphthongs from the Corpus of North American Spoken English, a 1.3-billion-word corpus of geolocated ASR transcripts of videos uploaded to YouTube channels of local government entities. In total, more than 2 million eɪ tokens were extracted from naturalistic speech in over 2000 locations in the US and Canada. The pipeline utilizes python code to extract audio segments from the DASH manifest of YouTube videos, then feeds them together with the corresponding transcript excerpt to the Montreal Forced Aligner. Targeted segments are then filtered using functions in Parselmouth, a python port of Praat. Because the underlying corpus is searchable and words are annotated with timing information, the extraction procedure is suitable for retrieving acoustic data from a range of naturalistic speech configurations (for example from discussions of particular topics or in the context of specific utterance types such as questions or replies). The corpus contains a variety of speech genres, making it suitable for studies of geophonetic and sociophonetic variation, and pipelines can be constructed to target specific phenomenona in YouTube videos, depending on the underlying research question.

5aSC29. Leveling of /o/ in Raleigh. Sean Lundergan (English, NC State Univ., 2211 Hillsborough St., Raleigh, NC 27607, slunder@ncsu.edu)

Beginning in the late 1950s, Raleigh, North Carolina, experienced high levels of in-migration from areas outside of the South. The resulting dialect contact led to leveling of many Southern features, including the Southern Vowel Shift. Change in the front vowel system of Raleigh has been the subject of many studies, but back vowels get less attention. Back vowel fronting is a widespread phenomenon in American English, but whereas back vowels in other regions are back-gliding, much of the South also fronted glides. This study investigates the change in /o/ (GOAT) in Raleigh during the 20th century, using sociolinguistic interview data from NC State's Raleigh corpus. The findings suggest that where previously /o/ was front- and upgliding, it now has a backward trajectory with little vertical movement. This is in line with previous findings of southern and non-southern variants of back vowels in the urban South and adds to our understanding of leveling in Raleigh. It also suggests future directions for research into the drivers of back vowel fronting.

5aSC30. Syllable position effects on English dark and light /l/ in comparison to /s/. Danilo A. Lombardo (Program in Speech-Language-Hearing Sci., The Graduate Ctr., City Univ. of New York, Speech Production, Acoust., and Percept. Lab., 365 Fifth Ave., New York, NY 10016-4309, dlombardo1@gradcenter.cuny.edu), D. H. Whalen (CUNY Graduate Ctr., New York, NY), Wei-Rong Chen (Haskins Labs., New Haven, CT), and Christine H. Shadle (Haskins Labs., New Haven, CT)

Effects of syllable position on the acoustic structure of speech sounds have been explored for different consonants, particularly for the lateral approximant /l/ in American English (AE). Sproat and Fujimura [*J. Phonetics* **21**, 291–311 (1993)] reported that /l/'s spectral properties vary between initial (light) and final (dark) position, as in [l]*ease* vs. *sea*[ɫ]; this has been replicated in other studies. A similar effect was found for the fricative /s/ [Lombardo *et al.*, *J. Acoust. Soc. Am.* **151**, A45 (2022)], though the acoustic measures necessarily differed. We explored whether the magnitude of a light/dark difference varies by syllable position in AE words with both /s/ and /l/. Preliminary results show that the magnitude of positional differences for /l/ was uncorrelated with that of the /s/. Like previous studies, the positional effect for /l/ occurred with all vowels. Our measures are distinct acoustically (formants during the lateral resonance versus both peak frequency and the amplitude of mid-to-high frequencies during the fricative); therefore, the inconsistency may be due to measurement factors. Alternatively, the physiological mechanism may differ enough that speakers themselves differ; or, indeed, /l/ differences may be phonologized while the /s/ differences are not.

5aSC31. Palatalization and tongue root harmony in Mongolian. Joshua Sims (Linguist, Indiana Univ., 2461 S Woolery Mill Dr., Bloomington, IN 47403, jodasims@iu.edu)

Khalkha Mongolian exhibits Tongue Root harmony, in which vowels in a word share specification of root features, as well as vowel fronting in the environment of palatalized consonants. This paper examines the interaction of the two processesof interest because the acoustic cues for tongue root harmony and palatalization oppose one another. Vowels in tongue root harmony systems are categorized by tongue root position, which triggers changes in F1. Advanced Tongue Root (ATR)corresponds to a decreased F1 (see Starwalt, 2008), while Retracted Tongue Root (RTR), active in Mongolian, corresponds to increased F1 (Svantesson, 1985). Meanwhile, a primary cue of palatalization is raised F2. In Mongolian, palatalized consonants contrast only in RTR words. Palatalized consonants front adjacent vowels, which exhibit increased F2 and decreased F1. These adjacent vowels are therefore targets for both F1 raising attributable to RTR and F1 lowering due to palatalization. The specific correlates of such vowels have not been previously reported, however. Thus, using data from a corpus of six Khalkha Mongolian speakers recorded in a laboratory setting, this paper reports on the acoustic correlates of RTR in Mongolian vowels in the context of palatalization.

5aSC32. Gasp! The reality of ingressive speech in contemporary French. Jane Gilbert (French & Italian, Indiana Univ., 355 Eagleson Ave., Bloomington, IN 47406, janegilb@iu.edu)

Pulmonic ingressive speech, the use of an inhalation airstream in speech, serves a set of similar paralinguistic functions in a wide variety of languages (Eklund, 2007). This phenomenon is most closely associated with Swedish and related languages (Eklund, 2008), affirmative particles (Eklund, 2007), and female speakers (Clarke andMelchers, 2005). It is known to be highly interactive and therefore difficult to elicit (Eklund, 2002). While acknowledged to exist in French, particularly used by women on the word *oui* "yes" when expressing "reticence" (Léon, 1992), this mode of articulation and its uses have yet to be described systematically in that language. In this study, we extracted all tokens of ingressive *oui* produced spontaneously over one

season (338 min) of the reality television series *L'Agence*, yielding 14 tokens. A detailed analysis of each in context, combined with observations from native speaker use and additional discourse-pragmatic observations from the study data, suggests that ingressive *oui* is used more often by men, is restricted to a semi-professional middle register, and is associated with final vowel devoicing, another known French phenomenon (Dalola, 2014). These data further indicate that, though rarer, ingressive *oui* is more perceptually salient when pronounced by a female speaker.

5aSC33. Compensatory articulatory behaviors after tongue reconstruction in production of English plosives: Establishing the range of lip movement patterns for control speakers. Natalie Hanas (Dept. of Commun. Sci. and Disord., Faculty of Rehabilitation Medicine, Univ. of Alberta, 116 St & 85 Ave., Edmonton, AB T6G 2R3, Canada, nhanas@ualberta.ca), Caroline C. Jeffery (Div. of Otolaryngol. Head and Neck Surgery, Dept. of Surgery, Univ. of Alberta, Edmonton, AB, Canada), and Daniel Aalto (Dept. of Commun. Sci. and Disord., Faculty of Rehabilitation Medicine, Univ. of Alberta, Edmonton, AB, Canada)

Head and neck cancer can have a devastating impact on speech and swallowing function. In particular, a tumor in the tongue can reduce the ability to produce articulatory gestures typical for English plosives. Previous case studies suggest that the lower lip can compensate for tongue tip gestures in speech after tongue reconstruction. The goal of this study is to establish typical lower lip movement patterns for alveolar and velar plosives for English speakers towards identifying compensatory lip movements in cancer speakers. Ten participants were recruited with no reported hearing or speech problems. A list of 40 minimal pairs beginning with the target plosives were created and embedded in carrier sentences. The participants read sentences in random order with and without babble noise in a standing posture. Lip motion was captured using a custom app on an iPhone 11 capturing the perioral surface area with the TrueDepth infrared camera. Preliminary results suggest that this app-based approach to lip tracking is a viable tool to capture typical lip movement patterns, ultimately enabling clinical research in more accessible settings. Mixed-effects linear regression analyses of the lip kinematics will be discussed.

5aSC34. The influence of co-speech gesture presence on the timing of F0 peaks in a tonal language. Kathryn Franich (Linguist, Harvard Univ., Boylston Hall, 3rd Fl., Cambridge, MA 02138, kfranich@fas.harvard.edu)

The timing of co-speech gestures has been shown to correlate with the location of pitch peaks in speech in several non-tonal languages. While little research has examined this relationship in tonal languages, existing work suggests that pitch peak timing does not show the same synchronous alignment with manual gesture timing. In this paper, we examine whether the presence of a gesture can have an influence on timing of pitch peaks in Medʉmba, a tonal Grassfields Bantu language. Manual gesture data were analyzed from spontaneous interview speech of six Medʉmba speakers. In line with prior findings on tonal languages, manual gesture apexes and pitch peaks did not align, and gesture-accompanied vowels were not realized with higher or more dynamic pitch profiles (though they were realized with longer duration and greater intensity; $ps < .01$). The findings did indicate, however, that f0 peaks—particularly for high, falling, and low tone syllables—were realized earlier in gesture-accompanied vowels than non gesture-accompanied vowels ($p < .05$). We interpret these findings in the context of an integrated model of speech and manual gesture in which tonal gestures and manual gestures are both crucially timed to vowels, but timed to occur sequentially with one another.

5aSC35. Japanese devoiced vowel tongue movement: Acoustics and articulory mapping. Marco Fonseca (Univ. of Illinois at Urbana-Champaign, 707 S Mathews Ave., Urbana, IL 61801, marcofon@illinois.edu)

The goal of this study is to evaluate the tongue movements of Japanese devoiced vowels /i/ and /u/ using acoustic and articulatory data. Four speakers of Japanese were recorded with an EchoBlaster 128Z ultrasound probe attached to their chin. Participants read a total of 22 tokens in a carrier sentence (12 repetitions). Duration (ms) and Center of Gravity (CoG, Hz) was fit in two linear mixed effects models as dependent variables, and the interaction between voicing (voiced/devoiced) and vowel (i/u) as predictors. For the duration values, there was an effect of both predictors. There was an effect of voicing and of the interaction between vowel and voicing. For the articulatory, the tongue splines were obtained from the last data frame of the segment before the vowel prone to devoicing were obtained through EdgeTrak. A total of 12317 x-y coordinates were fit into generalized additive models. In conclusion, devoiced vowels are shorter, and the CoG of devoiced vowels is higher than that of voiced ones. CoG results indicate that the former segments are more fronted than the latter. The tongue splines indicate that devoiced vowels are more fronted than voiced ones, which has also been indicated through the CoG data.

5aSC36. Lip posture of high and high-mid vowels in Hnaring Lutuv. Amanda Bohnert (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 504, Bloomington, IN 47405, abohnert@iu.edu) and Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, Bloomington, IN)

Lutuv is an under-documented South Central Tibeto-Burman language which is reported to contain six high monophthongs /i, y, ɨ, ʉ, ɯ, u/ and four diphthongal vowels with high onsets /iᵉ̯, yᵊ̯, uᵓ̯, ɯᵊ̯/ (Bohnert *et al.*, 2022). No articulatory work on Lutuv exists to date, but impressionistic observations suggest that while the contrast between /i, y/ and /u, ɯ/ is one of rounding (as implied by their IPA transcriptions), the contrast between /ɨ, ʉ/ is instead one of lip spreading. In addition, /ʉ/ seems to involve a degree of frication from labiodental contact. To further investigate these observations, the current paper utilizes facial landmark recognition through OpenFace 2.0 (Baltrusaitis *et al.*, 2018) to analyze the lip posture of these vowels, measuring horizontal, vertical, and lateral lip aperture over the duration of each vowel produced by one native speaker in a curated word list. These measurements are compared both within and across pairs to determine (1) the articulatory characteristics of Lutuv rounding contrasts, and (2) whether the central high vowels display these characteristics. These data complement existing acoustic analyses and provide a more detailed understanding of the articulation of a typologically rare contrast.

5aSC37. Quantifying tongue tip visibility in ultrasound images of /r/ tongue shapes using numerical ultrasound simulations. Sarah R. Li (Biomedical Eng., Univ. of Cincinnati, UC Bioscience Ctr., ML 0048-04D, 3159 Eden Ave., Cincinnati, OH 45219, lisr@mail.uc.edu), T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Midsagittal ultrasound images provide important articulatory information about tongue shape by showing much of the tongue surface from the root to the tip. However, portions of the tongue tip are often obscured due to shadowing from the mandible bone and the sublingual airspace. Information about tongue tip position can be important for understanding tongue movement patterns, e.g., classification of a retroflex or bunched shape for the American English rhotic /r/. Here, numerical simulations of acoustic wave propagation in tissue were performed using the open-source k-Wave toolbox to recreate ultrasound images of the tongue for a range of retroflex and bunched tongue shapes segmented from magnetic resonance imaging (MRI). Extent of the tongue tip visibly missing from the simulated ultrasound images is quantified by comparison with the known tongue shape from MRI. The fraction of the tongue surface visible in ultrasound images is compared between tongue shapes that appear retroflex versus bunched in MRI. Implications for ultrasound biofeedback therapy are discussed, including possible ambiguity of retroflex versus bunched classification.

5aSC38. Analyses of speech recordings from an acoustic head and torso simulator with and without face coverings using both spectrogram and transcription tools. Laura Ruhala (Mech. Eng., Kennesaw State Univ., Marietta, GA), Richard J. Ruhala (Mech. Eng., Kennesaw State Univ., 840 Polytechnic Ln., KSU - Mech. Eng. Dept., Rm. Q131, Marietta, GA 30060, rruhala@kennesaw.edu), Guillermo Brito, and Britt Cumbee (Mech. Eng., Kennesaw State Univ., Marietta, GA)

The COVID-19 pandemic caused the need for a wide assortment of face coverings to be adopted all over the globe. While reducing the transmission of the virus, the use of such face covering also reduced speech sound levels and intelligibly. Sound levels and recordings were made at a 2-mdistance from an acoustic head and torso simulator (HATS) with and without masks, including a KN95 mask, a surgical style mask, and a transparent plastic face shield. In this study, WAV recordings were made using short phases of prerecorded American English female and male talkers from the ITU-T P.50 database. Adobe Audition software was used to improve the playback quality of the speech recordings, trim the modified files into 10 sof speech, ensure that they all started with the same spoken words, and create spectrograms to identify the mask-related attenuations or amplifications. Adobe Premier Pro software was used for the transcription of these files. Documenting which words were transcribed correctly or incorrectly provided further indication of the effects of various face coverings on speech intelligibility.

5aSC39. The phrasal organization in speech gestures and co-speech head and eyebrow beats. Yoonjeong Lee (Linguist, Univ. of Michigan, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeol@umich.edu), Jelena Krivokapić, McLaren Lindsey (Linguist, Univ. of Michigan, Ann Arbor, MI), Vikas Tatineni, and Lynn Yambaye (Univ. of Michigan, Ann Arbor, MI)

Non-referential, beat-like gestures that accompany speech are often recruited as coparticipants for marking prosodic structure, especially signaling prominence. Only few studies have examined the coordination between vocal tract and co-speech gestures in relation to phrasal boundaries. Our recent study with eight Korean speakers (5F, 3M) has revealed that co-speech manual beat gestures and speech are synchronous with both phrase-edge segment and tone gestures [Lee *et al*., JASA **152**, A199 (2022)]. Building on this, the present study examines different gesticulators, namely, co-speech head and eyebrow movement using the same eight speakers. We specifically test how the spatiotemporal patterning of vocal tract actions and cooccurring body movements systematically varies at phrasal boundaries and under prominence. We hypothesize that co-speech head and eyebrow movement are also temporally coordinated with prosodic phrases in Korean. The movements of the lips, tongue, eyebrows, and head were point-tracked using EMA. Measurements taken included the duration of intervals from the timepoint of concurrent beat gesture onset and target to various phrasal landmarks. The results on inter-articulator organization of prosody will be discussed. [Work supported by NSF.]

5aSC40. Effect of speaking style and word length on V-to-V coarticulation in child and adult speech. Melissa A. Redford (Linguist, Univ. of Oregon, 1290 Linguist Dept., University of Oregon, Eugene, OR 97403-1290, redford@uoregon.edu) and Carissa A. Diantoro (Linguist, Univ. of Oregon, Eugene, OR)

Anticipatory vowel-to-vowel coarticulation indexes both the coherence of adjacent syllables in the speech plan and the relative "segmentalness" of articulation. In this study, speaking style and word length were manipulated to disrupt default production strategies and thereby identify age-related differences in planning and articulation between adults' and 8-year-old children's speech. Preliminary results indicate an anticipatory effect on schwa F1 and F2 in adults' speech across styles. The same effect was also observed on F1 in children's speech regardless of style, but the effect on schwa F2 was stronger when the target was in focus. There were also systematic effects of word length on F1 and F2 across both groups. For F1, unstressed vowels before one- and two-syllable words were more similar (more closed) than within four-syllable words; whereas, for F2, unstressed vowels were more similar (less fronted) before two-syllable words or within four-syllable words than before one-syllable words. Overall, neither speaking style nor word length appears to significantly disrupt the articulatory coherence of adjacent syllables in child or adult speech, but there is some indication of greater underlying coherence between syllables in children's speech compared to adults' speech. [Work supported by NICHD grant #R01HD087452.]

5aSC41. Examining vowel fronting and lengthening as indicators of feminine voices in Southern English speech. Irina Shport (Louisiana State Univ., 260 Allen Hall, Dept. of English, LSU, Baton Rouge, LA 70803, ishport@lsu.edu)

Fronting and lengthening of lax vowels /ɪ ɛ ɑ/ but not /ʊ/ was found to make words sound more feminine for Southern U.S. listeners (Shport, 2018). This study examined whether /ɪ ɛ ɑ/ are more fronted and are longer in female than male speech. Fifty speakers from Louisiana (27 Black, 23 White, 31 females, 19 males) participated in a word-reading task (4 vowels $\times$ 3 words $\times$ 4 repetitions). The effects of gender and ethnicity on Lobanov-normalized F2 and duration values were examined using generalized linear modeling. For duration, only /ɛ/ was significantly longer in female than in male productions. For F2, significant differences were found only in productions of the back vowels: /ɑ/ was fronted more by male than female speakers (the gender difference was larger in White speakers); /ʊ/ was fronted more by female than male speakers (and more so by White than Black speakers). These results suggest that Southerners' perception of relatively fronted and longer variants of /ɪ ɛ ɑ/ as sounding more feminine is not firmly grounded in the acoustic characteristics of these vowels in Southern female as compared to male speech. Possible influences of formant normalization and ethnolectal variation on result interpretation are discussed.

5aSC42. Improving signal-to-noise ratio in ultrasound video pixel difference. Pertti Palo (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, pertti.palo@taurlin.org) and Steven M. Lulich (Speech & Hearing Sci., Indiana Univ., Bloomington, IN)

Pixel difference is a holistic change measure for ultrasound videos and other image sequences. It is based on Euclidean distance (l2 norm) which is calculated by interpreting each frame as a vector of pixels. Recently, we found that 3D/4D ultrasound data producesignificantly worse signal-to-noise ratio in Pixel Difference curves, which appears to be primarily due to the lower frame rate of 3D/4D ultrasound when compared with 2D ultrasound. We also found that the signal-to-noise ratio is improved when we limit the analysis to tongue-internal pixels. In this study, we seek to quantify the timing behavior of tongue gestures obtained from whole-image and tongue-internal PD. If the timing behavior is similar in both cases, we can increase the signal to noise ratio of 3D/4D ultrasound at the same time as we increase its frame rate, because imaging to a shallower depth requires less time per frame. We also examine the signal-to-noise behavior of PD curves calculated with other norms (e.g., l1, and l3 norms).

5aSC43. Local stability and long distance reliability of tongue ultrasound spline metrics. Pertti Palo (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, pertti.palo@taurlin.org) and Steven M. Lulich (Speech & Hearing Sci., Indiana Univ., Bloomington, IN)

Tongue ultrasound data arecommonly analyzed by using different metrics. In this study, we evaluate the local stability and long distance reliability of Average Nearest Neighbour Distance (Zharkova and Hewlett, 2009), Median Point-by-Point Distance (Palo, 2020), Procrustes analysis and Modified Curvature Index (Dawsonet al., 2015). A metric which has good local stability should map a time-series of splines to a smooth scalar (or vector) time-series and not be very sensitive to small errors in the splining. Similarly, a metric with good long distance reliability should differentiate dissimilar tongue contours—such as different phoneme targets—reliably and without small splining errors significantly affecting the results. We test the metrics with simulated data generated from actual spline data. First, we select keyframes at phoneme target positions. Second, we vary the sampling frequency artificially by interpolating the splines between the keyframes. Third, we vary the level of splining noise by adding small perturbations to

the splines at varying magnitudes and frequencies. Based on the simulation results, we will calculate the distributions of each of the metrics as conditioned by sampling frequency and noise level. The code and simulation data will be publicly available.

5aSC44. An exploration of mixed race and the California Vowel Shift. Kaela Fong (North Carolina State Univ., 2211 Hillsborough St., Raleigh, NC 27607, kafong@ncsu.edu)

The California Vowel Shift (CVS) has been the study of much acoustic linguistic inquiry since it was first noted in the 1980s. It has since been grouped in with other like patterns which have arisen independently under the name, Low-Back-Merger-Shift (LBMS), though California will be the focus of the present study. The current study builds on previous literature by investigating acoustic vowel quality within a neighborhood in San Francisco, while also working to fill the gap of studies which take into account and specifically highlight mixed race peoples. Using vowel F1 and F2 measurements taken from informal interviews with 27 speakers from The Sunset district in San Francisco, a primarily white and Asian neighborhood, the vowel spaces of these speakers were examined. The aim of this project was to determine if racial/ethnic status as monoracial (Asian or White) or mixed-race had a significant effect on speakers' vowel quality and their level of participation in the CVS. Quantitative analysis found that speakers' vowel systems do exhibit the characteristics of CVS. For the back vowels /ɑ/, /ɔ/, /ʌ/, /ʊ/, /u/, and /o/ difference between speakers of mixed race and their white counterparts were found to be statistically significant.

FRIDAY MORNING, 12 MAY 2023 MICHIGAN/MICHIGAN STATE, 8:20 A.M. TO 11:35 A.M.

Session 5aUW

Underwater Acoustics: General Topics in Underwater Acoustics

Brian T. Hefner, Chair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105

Contributed Papers

8:20

5aUW1. Bottom-loss estimate from ship noise using a Ray-based Blind Deconvolution. Wonjun Yang (Marine Sci. & Convergence Eng., Hanyang Univ. ERICA, 55, Hanyangdaehak-ro, Sangnok-gu, Ansan-si15588, Gyeonggi-do, Republic of Korea, yangwon95@hanyang.ac.kr) and Jee Woong Choi (Marine Sci. & Convergence Eng., Hanyang Univ. ERICA, Ansan, Republic of Korea)

Ray-based Blind Deconvolution (RBD) is a technique that can estimate the channel impulse response of a receiver array even in the absence of acoustic source information. Combined with array invariant method, RBD has been applied in several studies, such as ship localization. In this talk, the channel impulse response was estimated through RBD for the ship noise of R/V *Onnuri* received by a 16-channel vertical line array moored at the bottom during the Shallow-water Acoustic Variability Experiment (SAVEX-15). The bottom loss is estimated as a function of grazing angle using the ratio of intensity impulse responses between direct path and bottom bounce path after compensating for the transmission loss for two paths. Finally, geoacoustic parameters for sediment at the SAVEX-15 site are estimated by comparing the estimated bottom loss to the Rayleigh bottom-loss model. Inversion results will be discussed in comparison with those presented in the literature. [Work supported by the Ministry of Oceans and Fisheries of Korea (PM63010, 2022).]

8:35

5aUW2. ALMA 2022 experiment : Source paramater estimation comparison betweenLFM and MLS waveforms. Gaultier Real (Underwater Warfare Dept., DGA Naval Systems, Ave. de la Tour Royale, Toulon 83000, France, gaultier.real@gmail.com), Kay L. Gemba (Dept. of Phys., Naval Postgrad. School, Monterey, CA), Kathrine Lamy, and Thomas Kacel (French Naval Acad., Lanvéoc, France)

For underwater source localization, accurate estimation of the time of flight (or travel time) is required and the observed waveform may include Doppler (waveform dilation). The source waveform, thus, needs to be sensitive to estimate both time of flight and Doppler. In this study, we compare two waveforms : Linear Frequency Modulated sweeps (LFMs) and Maximum Length Sequences (MLSs) with identical center frequency (6 kHz) and bandwidth (2 kHz). During an at-sea experiment conducted in March 2022 near in the Mediterranean Sea, near the coast of Toulon, a mobile source transmitted the two waveforms, recorded by a 192-hydrophone moored bilboard array. The source-array distance ranged from 2.5 to 8 km, and the source was towed at a constant speed ($\sim$1.5 m/s) and depth ($\sim$50 m). The analysis of the gathered data shows that the MLS waveform outperforms the LFM waveform for estimating time of flight (uncertainty reduced by a factor 4) and radial velocity (uncertainty reduced by a factor 20). Preliminary results focus on single hydrophone processing, but possible applications to 2D array processing are discussed for localizing the source in range-dependent environments.

8:50

5aUW3. An architecture for passive joint localization and environment learning in shallow-water underwater acoustic settings. Toros Arikan (EECS, MIT, 70 Amherst St., Apt 248, Cambridge, MA 02142, toros_arikan@hotmail.com), Amir Weiss (Elec. Eng. and Comput. Sci., MIT, Cambridge, MA), Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore), Grant B. Deane (Scripps Inst. Of Oceanogr., La Jolla, CA), Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL), and Greg Wornell (EECS, MIT, Cambridge, MA)

Passive localization and tracking of a mobile emitter, and joint learning of its reverberant 3D environment, is a challenging task in various application domains. In underwater acoustic monitoring with a receiver array, for example, a submarine may need to be tracked in a setting with natural and man-made obstacles, such as seamounts or piers. If such obstacles occlude the line of sight from this vessel to the receivers, then the non-line of sight reflected arrivals from the reverberant environment must be leveraged for localization. Hence, we need to precisely map these reflective features in order to deliver robust performance. We propose a multi-stage global optimization and tracking architecture to approach this problem. Each stage of this architecture establishes domain knowledge such as synchronization and occluder mapping, which are inputs for the following stages of more refined algorithms. This approach is generalizable to different physical scales, and improves on methods that do not exploit emitter motion. We further introduce a robust neural network-based reflector estimation method that outperforms its alternatives in realistic application settings. The performance of this holistic approach is analyzed and its reliability is demonstrated both in simulation and in a real-life reverberant watertank, which models shallow-water acoustic environments.

9:05

5aUW4. Spatial clustering of sound speed distributions. Duncan Williams (Defence Sci. and Technol. Lab., Dstl Porton Down, Salisbury SP4 0JQ, United Kingdom, dpwilliams@dstl.gov.uk)

It is increasingly important to our understanding of the effects of climate change on our oceans that we can model sound propagation in the future environments that are being predicted. This brings conflicting demands to increase the fidelity of sound propagation models, to account for a greater number of dynamic ocean properties, and their efficiency, in particular, to account for the spatial and temporal distribution of sound speed data. In this paper, different spatial clustering methods are developed to group similar sound speed data into a small number of clusters based on acoustically relevant properties of the sound speed structure. The clusters then represent the sound speed distribution, derived from thousands of individual sound speed profiles, with a small number of effective sound speed profiles which have similar properties. This is illustrated on data for the Norwegian Sea where the performance of each method is evaluated, using measures of the separation between clusters, and by comparing sound speed predictions using profiles sampled from each cluster and using the effective sound speed profile. We show that spatial clustering is an effective tool for reducing the number of sound speed profiles that are needed to represent their spatial distribution.

9:20

5aUW5. Mid-frequency sound intensity fluctuations on the Washington shelf. Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dtang@uw.edu), Brian T. Hefner, Guangyu Xu, Eric I. Thorsos, Ramsey R. Harcourt, John B. Mickett, and Kumar Prakash (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A joint Oceanography/Acoustics experiment was conducted 15 July–13 August, 2022 in Washington's shallow coastal shelf waters to investigate mid-frequency sound intensity fluctuations and the oceanographic mechanisms driving them, including subsurface ducts, internal tides, and internal waves. Acoustic pulses centered at 3.5 kHz and 6 kHz were transmitted upshelf from a stationary source and recorded by receivers on two moorings at 10 km and 20 km ranges, respectively. A ship-towed Shallow Water Integrated Mapping System (SWIMS) repeatedly sampled the ocean between the source and receiver moorings, providing sound speed measurements as a function of range, depth, and time. In addition, five oceanographic moorings strategically placed around the acoustic path took additional ocean data over time. The sound speed profiles feature a surface mixed layer underneath which a subsurface duct with local sound speed minima of 1–2 m/s is frequently present. Analysis shows that (1) when the sound source is inside the subsurface duct, transmission loss is reduced by 15 dB compared to when the source is outside and (2) the statistics of the acoustic intensity are consistent with strong scattering. [Work supported by the Office of Naval Research.]

9:35

5aUW6. Subsurface acoustic ducts in the Northern California Current System. Guangyu Xu (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, guangyux@uw.edu), Ramsey R. Harcourt, Dajun Tang, Brian T. Hefner, Eric I. Thorsos, and John B. Mickett (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A survey of sound speed profiles obtained from archived mooring and glider observations over the continental shelf, slope, and further offshore along the U.S. Pacific Northwest coastline reveals a shallow subsurface duct that appears at depths below the mixed layer. Sound propagation simulations using parabolic equation techniques demonstrate that the presence of the duct has a strong impact on sound propagation at 3.5 kHz. The duct is more prevalent in summer to fall than in winter to spring and extends to at least 200 km offshore from the shelf break. The depth of the minimum sound speed of the duct decreases onshore from between 80 m and 100 m offshore of the continental slope to less than 60 m over the shelf. The duct is also more prevalent offshore of the shelf break than over the shelf. Two processes potentially contribute to the formation of the subsurface duct: (1) differential advection of water masses driven by large-scale circulation within the northern California Current Systemand (2) wintertime surface cooling and convective deepening of the surface mixed layer. Data analysis conducted in this study suggests that the former is likely the dominant mechanism. [Work supported by the Office of Naval Research.]

9:50–10:05 Break

10:05

5aUW7. Modal structure of the sound field in a shallow-water waveguide with a gassy sediment layer: Experiment and theory. Marina Yarina (Marine Geoscience, Univ. of Haifa, Haifa, Israel), Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), and Oleg A. Godin (Phys., Naval Postgrad. School, Monterey, CA)

Motivated by a series of experiments in the Lake Kinneret (Israel), the paper investigates low-frequency sound propagation in the three-layer model of the shallow water waveguide, where a thin layer of gas-saturated sediment is situated between a homogeneous fluid sub-bottom and a continuously stratified water column. Typical values of the thickness and sound speed in the gassy layer are 1 m and 250 m/s. The layer is thin compared to typical water depth of about 40 m. Normal mode structure of the acoustic field is analyzed. It is shown that with increasing frequency each normal mode transforms into a mode trapped in the gassy layer. These modes have unusually small phase and group speeds that are significantly less than the sound speed in water. Theoretically predicted normal mode dispersion curves are compared to the dispersion curves retrieved from observations of shipping noise. Phase speeds of normal modes are measured by cross-correlating the noise recorded on two sparse vertical arrays. A technique is proposed for solving the inverse problem of determining the parameters of a gas-saturated layer by matching the measured and modeled frequency dependencies of the normal mode phase speed.

10:20

5aUW8. Scattering statistics of nonstationary seafloors: Mixture models and choosing the number of components. Derek Olson (Oceanogr., Naval Postgrad. School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943, dolson@nps.edu)

Modern high resolution synthetic aperture sonars are capable of resolving the structure of patchy seafloors, defined as seafloors having nonstationary geoacoustic and roughness properties. Within each patch, it is often possible to model the amplitude statistics using commonly used physics based statistical models, such as the K-distribution. However, it is impossible to use on a large scale because either the parameters change from patch to patchor the imaging sonar alters the scattering statistics at different ranges. Here, we propose a method to segment different patches in a sonar image using a mixture of physicallymotivated statistical models. We detail how the number of components can be chosen using Bayesian model selection techniques, such as the deviance information criterion. The trade-off between model accuracy and robustness is explored using a dataset provided by the Norwegian Defence Research Establishment.

10:35

5aUW9. Impacts of wind and river discharge on the variability of mid-frequency sonar in Dabob Bay. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), Dajun Tang (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA), and Grant Eastland (System Acceptance & Operational Readiness, Naval Undersea Warfare Ctr. Div. Keyport, Keyport, WA)

Dabob Bay is a long, narrow bay in Washington with a depth of 195 m at its center. In January, storms consistently come from the southwest creating waves that run along the long axis of the bay making it a natural laboratory in which to study their impact on sound propagation and reverberation. In January 2020, the Autonomous Reverberation Measurement System (ARMS) was deployed in the northern end of the bay and transmitted pulses every 20 s over 8-hperiods to a vertical line array deployed at a range of 5 km and in line with the prevailing wind direction. The sound speed profile during the experiment was upward-refracting with a layer of cold, fresh water at the surface that varied in thickness from 5 to 19 m due to rain-driven changes in river flow into the bay. The changes in the layer thickness produced 10–15 dB changes in transmission loss at time scales on the order of 30 min while the surface roughness produced ping-to-ping variations on the order of 5–10 dB. This talk focuses on efforts to capture these environmental impacts on propagation and reverberation in acoustic models. [Work supported by the Office of Naval Research.]

10:50

5aUW10. Comparison of imaging techniques for the analysis of bubble distribution in ship wakes. Ho Seuk Bae (Agency for Defense Development, Jinhae P.O.Box 18, Jinhae-gu, Changwon-si 51678, Korea (the Republic of), belfre@add.re.kr), Won-Ki Kim (Agency for Defense Development, Changwon, Korea (the Republic of)), and Su-Uk Son (Agency for Defense Development, Changwon, Korea (the Republic of))

Ship wakes, which are inevitably caused by ship movement, are a subject of great interest for military purposes. In this study, we created an underwater acoustic signal measuring device to characterize the distribution of wakes from a ship maneuvering in a real ocean environment. Two sets of multi-array sonar systems consisting of multiple projectors and multiple hydrophones were fabricated to acquire underwater acoustic signals. The systems were then placed vertically at a certain distance from each other in the sea. Afterward, we stably acquired the transmitted and scattered signals from the bubble wake generated by an actual ship. The distribution of bubbles in the ship's wake was then imaged by applying the RTM (reverse-time migration) and WI (waveform inversion) numerical analysis techniques to image the bubble distribution in the ship wake. Finally, the results obtained using the two aforementioned techniques were analyzed and compared.

11:05

5aUW11. An analytical framework for low-power underwater backscatter communications. Ahmed Allam (MIT Media Lab, Massachusetts Inst. of Technol., 771 Ferst Dr NW (Love Building), Rm. 204, Atlanta, GA 30332-0405, a.allam@gatech.edu), Waleed Akbar (Elec. Eng. and Comput. Sci., Massachusetts Inst. of Technol., Cambridge, MA), and Fadel Adib (MIT Media Lab, Massachusetts Inst. of Technol., Cambridge, MA)

Acoustic underwater backscatter enables ultra-low-power communication with applications in ocean exploration, monitoring, navigation, and aquaculture. Unlike traditional communication systems, acoustic backscatter does not require active signal generation. Instead, it communicates data by modulating existing acoustic signals, requiring few microwatts of power for operation. Backscatter communication enables ultra-low power sensors by switching between absorbing and reflecting acoustic waves transmitted from a central station. The energy burden is shifted to the transmitting station, and the acoustic power supplied by the station can power the sensor, allowing for battery-free operation. Initial demonstrations of underwater backscatter communication were encouraging; however, the theoretical and practical limits are still unknown. In this work, we develop a multiphysics analytical framework for the communication and power link budget of underwater backscatter. The framework calculates practical communication and power-up range, transmitter power budget, and signal-to-noise ratio, accounting for transducers' characteristics and the underwater communication channel. The analytical predictions are validated for practical transducers using high-fidelity piezo-acoustic finite element simulations and experimental measurements. The framework will guide future backscatter systems design, identifying practical operation ranges and optimal frequencies for data transmission and batteryless operation.

11:20

5aUW12. Focusing late-time returns from elastic targets in bistatic collection geometries. Kyle S. Dalton (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ksd5377@psu.edu), Thomas E. Blanford, and Daniel C. Brown (The Penn State Univ., State College, PA)

Targets encountered during synthetic aperture sonar (SAS) surveys may exhibit elastic scattering behavior and re-radiate sound after initial interrogation. These re-radiated returns are often described as "late-time" energy, as they reach the sonar after the initial geometric returns. Range-specific focusing methods can enhance acoustic features such as late-time energy, but these methods typically make a monostatic sonar approximation that may not hold in very shallow water sensing geometries. This presentation will discuss the development of a late-time reconstruction algorithm applicable to bistatic sonar geometries. Using both simulated and experimental sonar data from monostatic and bistatic collection geometries, this work will first quantify the differences between traditionally beamformed imagery and late-time-focused imagery using common image quality metrics. Next, we will quantitatively compare the impact of collection geometry on the performance of late-time focusing algorithms. Finally, the work will describe methods that use information from a multi-static collection to compress late-time energy down to a high-contrast, spatiallycondensed point. The analysis of these signal processing strategies will focus on their potential applications to automatic target recognition.

ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human vertebrate animals in research, and for publishing and presentations. The principles endorsed by the Society primarily follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed or modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS), and the American Institute of Physics Publishing (AIPP). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and its publication or presentation.

Authors of manuscripts submitted for publication in a journal of the ASA or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievance Committee of the ASA.

I. USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

The ASA endorses the view that all research involving human subjects requires approval by an existing appropriate governing authority (e.g., institutional review board [IRB], Health Insurance Portability and Accountability Act [HIPPA], or by other governing authorities used in many countries) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should have met the following criteria:

Informed Consent

When obtaining informed consent from prospective participants in a research protocol, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentiality.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. That office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s), if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

Dispensing with Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
   a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
   b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
   c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. Research involving the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

II. HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well-being of both human and non-human vertebrate animals often require the use of animals in research, education, and testing. The ASA remains committed to ensuring the health and welfare of vertebrate animals used for these purposes. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS) document: "International Guiding Principles for Biomedical Research Involving Animals-2012").

The ASA endorses the view that all research involving non-human vertebrate animals, hereinafter referred as "animals," requires approval by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should meet the following criteria:

1. Animals have been used only when necessary and when no alternative methods, such as non-animal approaches, mathematical models, or computer simulation, are available to achieve the scientific goals.
2. Investigators have handled all animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
3. Investigators have made all reasonable efforts to minimize the number of animals used in research to achieve the scientific goals.
4. Investigators are experienced in the care of laboratory animals, supervise all procedures involving animals, ensure all subordinates who use animals have received proper training in methodology and animal care, and assume responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
5. The health and welfare of animals are the primary considerations in making decisions of animal care including acquisition, housing, veterinary care, and final disposition of animals.
6. All surgical procedures have been conducted under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
7. Investigators have made all reasonable efforts to monitor and mitigate any possible adverse effects to animals as a result of the experimental protocol. Strategies to manage, mitigate, and minimize any pain and/or distress in animals should be developed in consultation with a qualified veterinarian or scientist. Animals that suffer chronic pain, distress or discomfort that cannot be relieved should be removed from the study and/or euthanized using a procedure appropriate for the species and condition of the animal.
8. Investigators proceed to rapidly and humanely terminate an animal's life when it is necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by a veterinarian and/or appropriate review board.

III. PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

Statement of Ethics and Responsibilities

The mission of the ASA is to generate, disseminate, and promote the knowledge and practical applications of acoustics. To that end, it is essential that all authors of papers in ASA journals and presentors at ASA-sponsored meetings conduct themselves in accord with the highest level of professional ethics and standards.

By submitting a manuscript to an ASA journal, each author explicitly confirms that the manuscript meets the highest ethical standards. The same is required for material presented at meetings. Authors submitting to ASA journals should also adhere to the policies included in the particular journals' Instructions for Contributors.

This section is mainly based on the policies of the American Institute of Physics Publishing.

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Plagiarism is the unauthorized and unacknowledged use of someone else's words, ideas, processes, data, or results in a manner that can mislead others into thinking the material is your own. Plagiarism can also be in the form of text recycling, also called self-plagiarism, where an author reuses portions of text from their own work that isn't properly credited. Plagiarism or self-plagiarism constitutes unethical scientific behavior and is never acceptable.

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Authorship should be limited to those who have made a significant contribution to the concept, design, execution or interpretation of the research study. All those who have made significant contributions should be offered the opportunity to be listed as authors. The author who submits a paper for publication or an abstract for presentation and publication should ensure that all coauthors have seen the final version of the paper or abstract and have agreed to its submission. Other individuals who have contributed to the study should be acknowledged, but not identified as authors.

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The results of research should be recorded and maintained in a form that allows analysis and review, both by collaborators before publication and by other scientists for a reasonable period after publication. Exceptions may be appropriate in certain circumstances in order to preserve privacy, to assure patent protection, or for similar reasons.

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All coauthors have an obligation to provide prompt retractions or correction of errors in published works.

Fabrication of Data and Selective Reporting of Data

Fabrication of data is an egregious departure from the expected norms of scientific conduct, as is the selective reporting of data with the intent to mislead or deceive, as well as the theft of data or research results from others.

Disclosure of Conflicts of Interest

A conflict of interest is anything that interferes with, or could reasonably be perceived as interfering with, the full and objective presentation of articles in the ASA journals and presentations at the ASA meetings. Author(s) have the obligation to disclose any personal interest or relationship that has the potential to be affected by publication of the submitted manuscript or presentation at ASA meeting:

1. The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.
2. If the publication or presentation of the research would directly lead to the financial gain of the author(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or

in a footnote of a paper. Authors must report any financial interest in corporate or commercial entities dealing with the subject matter of the manuscript or presentation.

3. If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Authors must submit corrections if conflicts of interests are revealed after publication.

Approved by the Executive Council on 9 December 2019.

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The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

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ACOUSTICAL · SOCIETY · OF · AMERICA

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Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

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JASA Online–Vol. 1 (1929) to present	*	*	*	*
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Corresponding Electronic Associate: Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $50 per year.
Member: Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $115 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

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