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181st Meeting
of the Acoustical Society of America

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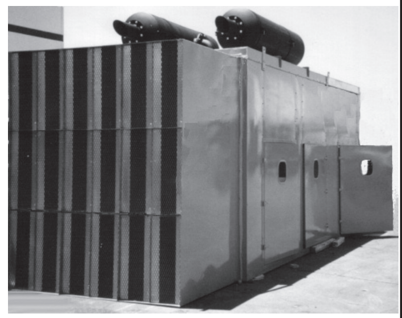

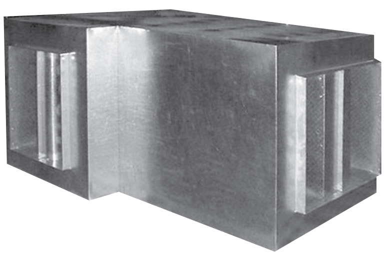

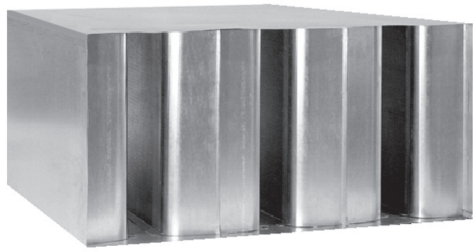

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181th Meeting: Acoustical Society of America

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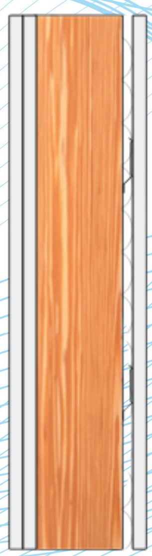

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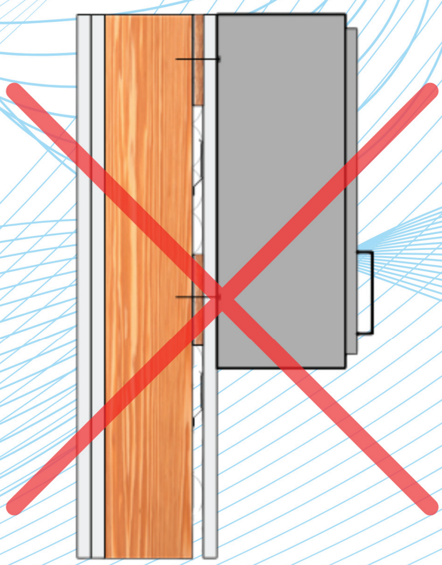

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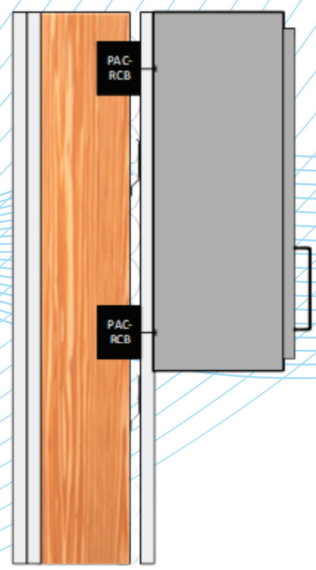

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CONTENTS

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 300, 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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SCHEDULE OF COMMITTEEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 29 November, 9:00 a.m.	Executive Council	Willapa
Mon, 29 November, 1:00 p.m.	Technical Council	Willapa
Tue, 30 November, 2:00 p.m.	Meetings Reimagined	305
Tue, 30 November, 5:00 p.m.	Women in Acoustics	Willapa
Wed, 1 December, 7:30 a.m.	Finance	305
Thu, 2 December, 7:30 a.m.	Editorial Board	Willapa
Thu, 2 December, 7:30 a.m.	Tutorials, Short Courses, Hot Topics	305
Thu, 2 December, 2:00 p.m.	Strategic Plan Champions	Willapa
Thu, 2 December, 5:00 p.m.	TCAA Healthcare Facilities	305
Fri, 3 December, 8:00 a.m.	Technical Council	Willapa
Fri, 3 December, 12:00 p.m.	Executive Council	Willapa

TECHNICAL COMMITTEEE OPEN MEETINGS

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Tue, 30 November, 7:30 p.m.	Acoustical Oceanography	Quinault
Tue, 30 November, 7:30 p.m.	Animal Bioacoustics	302
Tue, 30 November, 7:30 p.m.	Architectural Acoustics	Elwha A
Tue, 30 November, 7:30 p.m.	Physical Acoustics	402
Tue, 30 November, 7:30 p.m.	Psychological and Physiological Acoustics	Elwha B
Tue, 30 November, 7:30 p.m.	Signal Processing in Acoustics	502
Tue, 30 November, 7:30 p.m.	Structural Acoustics and Vibration	301
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Thu, 2 December, 4:30 p.m.	Computational Acoustics	402
Thu, 2 December, 7:30 p.m.	Musical Acoustics	Quinault
Thu, 2 December, 7:30 p.m.	Noise	Elwha A
Thu, 2 December, 7:30 p.m.	Speech Communication	Columbia D
Thu, 2 December, 7:30 p.m.	Underwater Acoustics	Elwha B

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

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Mon, 29 Nov 7:00 a.m. - 5:00 p.m. Tue-Thu, 30 Nov-2 Dec 7:30 a.m. - 5:00 p.m. Fri, 3 Dec 7:30 a.m. - 12:00 noon	Registration	3rd Floor
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Mon, 29 November 5:30 p.m. - 6:45 p.m.	Student Meet and Greet	Columbia B
Mon, 29 November 5:30 p.m. - 7:00 p.m.	Exhibit Opening Reception	Exhibit, 3rd Floor
Tue, 30 November 9:00 a.m. - 5:00 p.m.	Exhibit	3rd Floor
Tue, 30 November, 10:00 a.m. - 12 noon	Diversity Statement Workshop	Columbia B
Tue, 30 November 4:00 p.m. - 5:30 p.m.	Business Development Workshop	Columbia B
Tue, 30 November 6:00 p.m. - 7:30 p.m.	Social Hour	Columbia C/D
Wed, 1 December 9:00 a.m. - 12:00 noon	Exhibit	3rd Floor
Wed, 1 December, 8:00 a.m. - 12:00 noon	Solutions Showcase	Willapa
Wed, 1 December, 11:45 a.m. - 1:30 p.m.	Women in Acoustics Luncheon	Columbia B
Wed. 1 December, 2:30 p.m.	Women’s Roundtable	Willapa
Wed, 1 December, 3:30 p.m. - 5:30 p.m.	Plenary Session/Awards Ceremony	Columbia C/D
Wed, 1 December, 6:00 p.m. - 8:00 p.m.	Student Reception	Columbia B
Wed, 1 December, 6:00 p.m. - 8:30 p.m.	Listen Up	Willapa
Wed, 1 December, 8:00 p.m. - 12:00 midnight	ASA Jam	Columbia C/D
Thu, 2 December, 12:00 noon-2:00 p.m.	Society Luncheon and Lecture	Columbia B
Thu, 2 December, 6:00 p.m. - 7:30 p.m.	Social Hour	Columbia C/D

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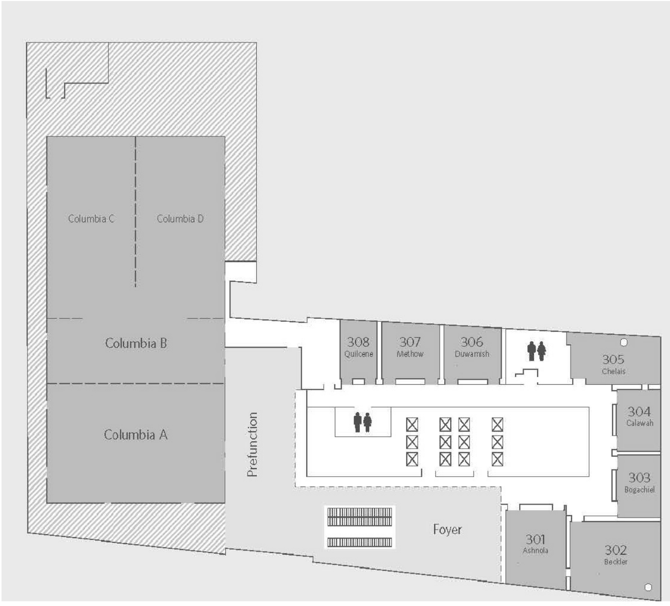

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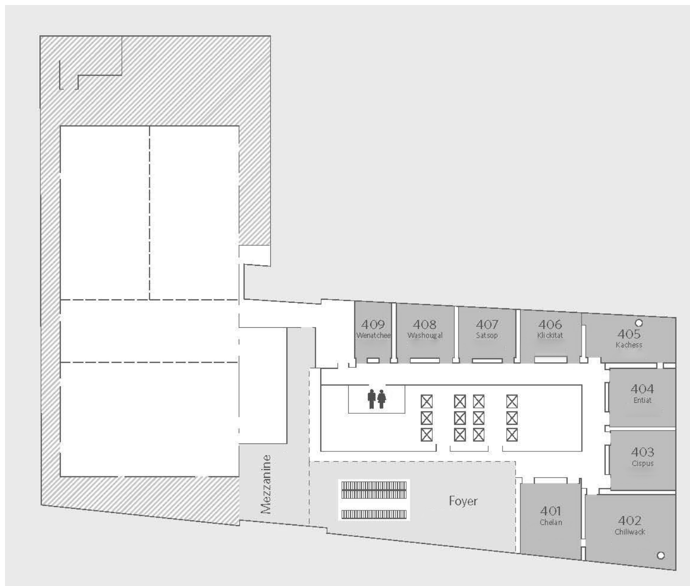

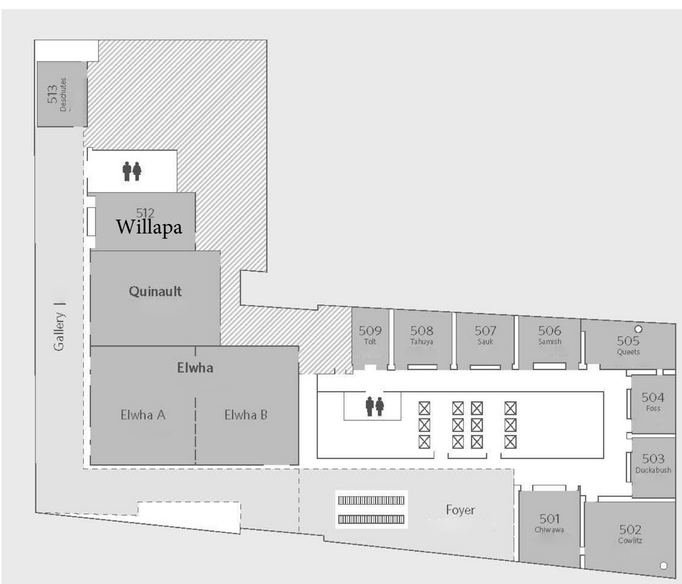

Level 5

ASA School 2022

Living in the Acoustic Environment

21-22 May 2022
Englewood, CO

ASA School 2022 is an Acoustical Society of America event for graduate students and early career acousticians in all areas of acoustics to learn about and discuss a wide variety of topics related to the interdisciplinary theme *Living in the Acoustic Environment*. ASA School 2022 follows on the success of four previous ASA Schools starting in 2012, and will provide opportunities for meeting instructors and fellow attendees, mentoring, discussing research topics, and developing collaborations and professional relationships within acoustics.

Program and Costs

ASA School 2022 will take place at the Hilton Denver Inverness in Englewood, CO, a resort and spa 30 minutes from Denver. Lectures and demonstrations followed by discussions will be given by distinguished acousticians in a two-day program covering topics in *architectural acoustics, animal bioacoustics, biomedical acoustics, engineering acoustics, musical acoustics, noise, psychological and physiological acoustics, and speech communication*. Although ASA School 2022 will focus primarily on these 8 technical areas, graduate students and early career professionals in all areas of acoustics are encouraged to attend to achieve a broader understanding of the diverse fields of acoustics.

The registration fee is $50. Hotel rooms at the Hilton Denver Inverness for two nights (single occupancy) and meals will be provided by ASA. Participants are responsible for their own travel costs and arrangements including transportation to the Hilton Denver Inverness. Transportation from the Hilton Denver Inverness to the ASA meeting location in Denver at the close of ASA School 2022 will be provided and paid by ASA. The COVID-19 vaccination and mask policies in effect in May 2022 will be followed at ASA School 2022.

Participants and Requirements

ASA School 2022 is targeted to graduate students and early career acousticians (within 5 years of terminal degree) in all areas of acoustics. Attendance is limited to 60 participants who are expected to attend all School events and the ASA meeting immediately following on 23-27 May 2022. ASA School attendees are required to be an author or co-author on an abstract for presentation at the ASA Denver meeting.

Application and Deadlines

The application form and preliminary program will be available online in December 2021, at www.AcousticalSociety.org.

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TECHNICAL PROGRAM CALENDAR
181st Meeting of the Acoustical Society of America
29 November–3 December 2021

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

Monday Morning

9:10 1aAA **Architectural Acoustics and Noise:** Back to Basics: Skill-Building Seminar I. Elwha A

8:20 1aAO **Acoustical Oceanography and Underwater Acoustics:** Upper Ocean and Mixed Layer Acoustics. Elwha B

8:00 1aBAa **Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics:** Cavitation Nuclei: Bubbles, Droplets, and More I. Room 401 (L)/404 (O)

8:30 1aBAb **Biomedical Acoustics:** General Topics in Biomedical Acoustics I. Room 405(L) / 406 (O)

9:00 1aED **Education in Acoustics:** Returning to Teaching Acoustics In-Person in the Post-COVID Era. Room 302 (L)/ 303 (O)

7:55 1aPA **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics I. Room 402 (L)/403 (O)

8:00 1aSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, Computational Acoustics, and Signal Processing in Acoustics:** Novel Techniques for Nondestructive Evaluation in Structural Acoustics and Vibration I. Room 501 (L)/504 (O)

9:00 1aSC **Speech Communication:** Second-Language Speakers and Listeners (Poster Session). Columbia A

Monday Afternoon

1:10 1pAA **Architectural Acoustics and Noise:** Back to Basics: Skill Building Seminar II. Elwha A

1:00 1pABa **Animal Bioacoustics:** Population Monitoring. Room 602 (L)/603 (O)

2:20 1pABb **Animal Bioacoustics:** Analysis and Classification of Animal Bioacoustics Signals. Room 602 (L)/603 (O)

1:00 1pAO **Acoustical Oceanography and Signal Processing in Acoustics:** Acoustical Oceanography Using Ocean Observatory Systems I. Elwha B

1:00 1pBAa **Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics:** Cavitation Nuclei: Bubbles, Droplets, and More II. Room 401 (L)/404 (O)

1:00 1pBAb **Biomedical Acoustics:** General Topics in Biomedical Acoustics II. Room 405 (L)/406 (O)

1:00 1pED **Education in Acoustics:** General Topics in Education in Acoustics. Room 302 (L)/303 (O)

1:20 1pNS **Noise, Architectural Acoustics, ASA Committee on Standards, and Structural Acoustics and Vibration:** Noise Issues in Modern Construction. Room 502 (L)/503 (O)

1:20 1pPA **Physical Acoustics, Biomedical Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration:** Acoustofluidics II. Room 402 (L)/403 (O)

1:30 1pPP **Psychological and Physiological Acoustics, Education in Acoustics, Speech Communication, and Biomedical Acoustics:** Acoustics Outreach to Student Scientists in Clinical and Physiological Research. Quinault

1:15 1pSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, Computational Acoustics, and Signal Processing in Acoustics:** Novel Techniques for Nondestructive Evaluation in Structural Acoustics and Vibration II. Room 501 (L)/504 (O)

1:00 1pSC **Speech Communication:** Speech Acoustics and Production (Poster Session). Columbia A

4:00 1pID **Interdisciplinary:** Keynote Lecture: An Unexpected Journey Through Sound. Columbia C

Monday Evening

7:00 1eID **Interdisciplinary:** Tutorial Lecture on Software Best Practices. Columbia D

Tuesday Morning

8:00 2aAAa **Architectural Acoustics, ASA Committee on Standards, and Signal Processing in Acoustics:** Recording and Production Spaces. Elwha A

8:00 2aAAb **Architectural Acoustics:** Student Design Competition. Columbia A

8:00 2aAB **Animal Bioacoustics and ASA Committee on Standards:** Classifying and Quantifying Natural Soundscapes I. Room 602 (L)/603 (O)

8:20 2aAO **Acoustical Oceanography and Signal Processing in Acoustics:** Acoustical Oceanography Using Ocean Observatory Systems II. Elwha B

8:00 2aBAa **Biomedical Acoustics, Signal Processing in Acoustics, Engineering Acoustics and Physical Acoustics:** Nonlinear Acoustic Propagation: Theory, Simulations, Experiment, and Applications I. Room 401 (L)/404 (O)

9:00 2aBAb **Biomedical Acoustics:** General Topics in Biomedical Acoustics III. Room 405 (L)/406 (O)

8:00 2aCA **Computational Acoustics and Signal Processing in Acoustics:** Validation and Verification (V&V) for Acoustics and Vibrations Simulations. Room 402 (L)/403 (O)

10:20 2aEA **Engineering Acoustics:** General Topics in Engineering Acoustics. Room 306 (L)/307 (O)

9:00 2aMUa **Musical Acoustics and Physical Acoustics:** Nonlinearities in Musical Acoustics. Quinault

10:45 2aMUb **Musical Acoustics:** Acoustics of Brass Instruments. Quinault

8:00 2aNSa **Noise and ASA Committee on Standards:** General Topics on Noise and Standards. Room 301 (L)/304 (O)

9:45 2aNSb **Noise, Architectural Acoustics, Strucutral Acoustics and Vibration and Engineering Acoustics:** Building Systems Noise and Vibration Control: Beyond ASHRAE Chapter 49 I. Room 301 (L)/304 (O)

8:00 2aPA **Physical Acoustics, Engineering Acoustics, Noise, Structural Acoustics and Vibration, and ASA Committee on Standards:** Wind Noise. Room 407 (L)/408 (O)

8:00 2aPP **Psychological and Physiological Acoustics and Animal Bioacoustics:** Auditory Perception of Stationary and Moving Sounds I. Columbia C

8:00 2aSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials I. Room 302 (L)/303 (O)

9:00 2aSC **Speech Communication and Psychological and Physiological Acoustics:** Development of Sensory-Motor Connections for Speech. Columbia D

9:00 2aSP **Signal Processing in Acoustics, Physical Acoustics, Underwater Acoustics,vBiomedical Acoustics, Noise, and Architectural Acoustics:** Signal Processing for Non-Specialists. Room 501 (L)/504 (O)

9:00 2aUW **Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics:** Acoustics of Underwater Explosions. Room 502 (L)/503 (O)

Tuesday Afternoon

1:00 2pAA **Architectural Acoustics, ASA Committee on Standards, Engineering Acoustics, and Signal Processing in Acoustics:** Current State-of-the-Art and Future Paradigms in Acoustic Measurement and Testing. Elwha A

1:00 2pAB **Animal Bioacoustics:** Classifying and Quantifying Natural Soundscapes II. Room 602 (L)/603 (O)

1:00 2pAO **Acoustical Oceanography and Signal Processing in Acoustics:** Acoustical Oceanography Using Ocean Observatory Systems III. Elwha B

1:00 2pBAa **Biomedical Acoustics, Signal Processing in Acoustics, and Engineering Acoustics:** Nonlinear Acoustic Propagation: Theory, Simulations, Experiment, and Applications II. Room 401 (L)/404 (O)

2:55 2pBAb **Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics:** Cavitation Nuclei: Bubbles, Droplets, and More III. Room 401 (L)/404 (O)

4:35 2pBAc **Biomedical Acoustics and Signal Processing in Acoustics:** Ultrasound for Brain Stimulation. Room 401 (L)/404 (O)

1:20 2pCA **Computational Acoustics, Physical Acoustics, Musical Acoustics, and Noise:** Computational Aeroacoustics. Room 306 (L)/307 (O)

3:00 2pID **Interdisciplinary and Student Council:** Introduction to Technical Committees. Quinault

1:00 2pNS **Noise, Architectural Acoustics, Structural Acoustics and Vibration and Engineering Acoustics:** Building Systems Noise and Vibration Control: Beyond ASHRAE Chapter 49 II. Room 301 (L)/304 (O)

2:30 2pPA **Physical Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics:** The Impact of Logan Hargrove on Physical Acoustics and Beyond. Room 402 (L)/403 (O)

1:00 2pPPa **Psychological and Physiological Acoustics and Animal Bioacoustics:** Auditory Perception of Stationary and Moving Sounds II. Columbia C

2:45 2pPPb **Psychological and Physiological Acoustics:** Psychoacoustics and Neural Processing (Poster Session). Columbia A

1:00 2pSA **Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics:** Acoustic Metamaterials II. Room 302 (L)/303 (O)

1:00 2pSC **Speech Communication:** Child Speech Production and Perception (Poster Session). Columbia A

1:00 2pSP **Signal Processing in Acoustics:** General Topics in Signal Processing I. Room 501 (L)/504 (O)

1:00 2pUW **Underwater Acoustics:** General Topics in Underwater Acoustics: Inversions and Parameter Estimation. Room 502 (L)/503 (O)

Wednesday Morning

9:00 3aAA **Architectural Acoustics and ASA Committee on Standards:** Restaurant Acoustics. Elwha A

8:20 3aAB **Animal Bioacoustics:** Animal Bioacoustics Poster Session (Poster Session). Columbia A

9:00 3aBAa **Biomedical Acoustics and Physical Acoustics:** Biomedical Acoustics Modeling Workshop. Room 401 (L)/404 (O)

8:40 3aBAb **Biomedical Acoustics:** Biomedical Acoustics in Ophthalmology I. Room 405 (L)/406 (O)

9:00 3aCA **Computational Acoustics, Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, Structural Acoustics and Vibration, Musical Acoustics, Architectural Acoustics, Physical Acoustics, Biomedical Acoustics, Psychological and Physiological Acoustics, Animal Bioacoustics, and Engineering Acoustics:** Showcases of High Performance Computing in Acoustics I. Room 502 (L)/503 (O)

8:45 3aEA **Engineering Acoustics and Signal Processing in Acoustics:** Acoustics and Human-Machine Interface. Room 602 (L)/603 (O)

9:00 3aMUa **Musical Acoustics:** Making Music During a Pandemic. Room 302 (L)/303 (O)

10:10 3aMUb **Musical Acoustics:** General Topics in Musical Acoustics I. Room 302 (L)/303 (O)

8:40 3aNS **Noise, Physical Acoustics, ASA Committee on Standards, Structural Acoustics and Vibration, and Signal Processing in Acoustics:** Jet and Launch Vehicle Noise. Room 301 (L)/304 (O)

8:00 3aPA **Physical Acoustics, Signal Processing in Acoustics, ASA Committee on Acoustics, Computational Acoustics, and Engineering Acoustics:** Infrasound I. Room 402 (L)/403 (O)

8:25 3aPP **Psychological and Physiological Acoustics and Speech Communication:** Acoustic Outreach to Early Career Scientists in Clinical and Physiological Research: Top-Down Influences on Auditory Processing. Columbia C

8:20 3aSA **Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, and Signal Processing in Acoustics:** Experimental Methods for Material Characterization in Structural Acoustics and Vibration. Room 501 (L)/504 (O)

9:00 3aSC **Speech Communication:** Articulation and Voice (Poster Session). Columbia A

8:20 3aSP **Signal Processing in Acoustics, Physical Acoustics, Underwater Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration:** Time Reversal Acoustics. Quinault

7:55 3aUW **Underwater Acoustics:** General Topics in Underwater Acoustics: Modeling and Measurements I. Elwha B

Wednesday Afternoon

1:00 3pAA **Architectural Acoustics:** AIA CEU Presenters Course Training Session. Elwha A

1:30 3pAB **Animal Bioacoustics:** Animal Bioacoustic Signals Transmission and Environment. Room 602 (L)/603 (O)

1:00 3pAOa **Acoustical Oceanography:** Acoustical Oceanography Prize Lecture I. Elwha B

2:15 3pAOb **Acoustical Oceanography:** Acoustical Oceanography Prize Lecture II. Elwha B

1:00 3pBAa **Biomedical Acoustics:** Biomedical Acoustics Student Poster Competition. Columbia A

1:00 3pBAb **Biomedical Acoustics:** Biomedical Acoustics in Ophthalmology II. Room 405 (L)/406 (O)

1:00 3pCA **Computational Acoustics, Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, Structural Acoustics and Vibration, Musical Acoustics, Architectural Acoustics, Physical Acoustics, Biomedical Acoustics, Psychological and Physiological Acoustics, Animal Bioacoustics, and Engineering Acoustics:** Showcases of High Performance Computing in Acoustics II. Room 502 (L)/503 (O)

2:20 3pID **Interdisciplinary:** Hot Topics in Acoustics. Quinault

1:30 3pMU **Musical Acoustics:** General Topics in Musical Acoustics II. Room 302 (L)/303 (O)

1:00 3pNS **Noise, Physical Acoustics, Signal Processing in Acoustics, Structural Acoustics and Vibration, and ASA Committee on Standards:** Military Jet Noise Measurements and Analysis. Room 301 (L)/304 (O)

1:00 3pPA **Physical Acoustics, Signal Processing in Acoustics, ASA Committee on Standards, Computational Acoustics, and Engineering Acoustics:** Infrasound II. Room 402 (L)/403 (O)

1:00 3pPP **Psychological and Physiological Acoustics:** Auditory Neuroscience Prize Lecture. Quinault

1:00 3pSA **Structural Acoustics and Vibration, Education in Acoustics, and Noise:** Perspectives from Senior Researchers in Structural Acoustics and Vibration. Room 501 (L)/504 (O)

Wednesday Evening

7:00 3eED **Education in Acoustics and Women in Acoustics:** Listen Up and Get Involved! Quinault

Thursday Morning

8:20 4aAA **Architectural Acoustics:** Architectural Acoustics Potpourri. Elwha A

8:30 4aAB **Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics:** Applications of Bioacoustics in Killer Whale Conservation I. Room 602 (L)/603 (O)

7:50 4aAO **Acoustical Oceanography, Signal Processing in Acoustics, ASA Committee on Standards, Computational Acoustics, and Engineering Acoustics:** New Data Sources and Processing Techniques in Fisheries Acoustics: Revolutionary or Incremental Progress? Elwha B

8:00 4aMU **Musical Acoustics:** General Topics in Musical Acoustics III. Room 302 (L)/303 (O)

8:15 4aNSa **Noise and ASA Committee on Standards:** Aircraft Sonic Boom and Community Noise Assessment. Room 402 (L)/403 (O)

10:15 4aNSb **Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics:** General Topics on Noise-Induced Hearing Loss and Protection. Room 402 (L)/403 (O)

8:00 4aPA **Physical Acoustics and Biomedical Acoustics:** Session in Honor of David T. Blackstock I. Quinault

8:20 4aPP **Psychological and Physiological Acoustics:** Open Source Audio Processing Challenge Results - Hackathon Challenge Presentations. Columbia C

8:00 4aSA **Structural Acoustics and Vibration and Engineering Acoustics:** Historical Development of Fuzzy Structures Concepts. Room 401 (L)/404 (O)

9:00 4aSC **Speech Communication:** Speech Communication in Challenging Situations (Poster Session). Columbia A

8:00 4aSP **Signal Processing in Acoustics, Physical Acoustics, Biomedical Acoustics, Noise, and Animal Bioacoustics:** Signal Processing Methods for Source Classification and Localization in Real Acoustic Environments I. Room 501 (L)/504 (O)

8:20 4aUW **Underwater Acoustics, Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics:** Waveguide Invariant Theory and its Applications I. Room 502 (L)/503 (O)

Thursday Afternoon

1:30 4pAB **Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics:** Applications of Bioacoustics in Killer Whale Conservation II. Room 602 (L)/603 (O)

1:30 4pBA **Biomedical Acoustics, Structural Acoustics and Vibration, Physical Acoustics, Underwater Acoustics, Engineering Acoustics, and Signal Processing in Acoustics:** Novel Ultrasound Imaging Approaches for Muscle Characterization. Elwha A

1:30 4pED **Education in Acoustics:** Preview of Next JASA Special Issue on Education in Acoustics. Elwha B

1:00 4pNS **Noise, ASA Committee on Standards, Architectural Acoustics, Education in Acoustics, and Psychological and Physiological Acoustics:** Sound Walks. Room 402 (L)/403 (O)

1:30 4pPA **Physical Acoustics and Biomedical Acoustics:** Session in Honor of David T. Blackstock II. Quinault

1:30 4pPPa **Psychological and Physiological Acoustics and Speech Communication:** Challenges and Opportunities in Technology for Remote and Virtual Testing. Columbia C

2:30 4pPPb **Psychological and Physiological Acoustics:** Binaural Hearing and Speech Perception (Poster Session). Columbia A

1:30 4pSA **Structural Acoustics and Vibration, Engineering Acoustics, Musical Acoustics, and Physical Acoustics:** Additive Manufacturing and Acoustics. Room 401 (L)/404 (O)

1:30 4pSC **Speech Communication:** Perception and Social Evaluation (Poster Session). Columbia A

1:00 4pSP **Signal Processing in Acoustics, Physical Acoustics, Biomedical Acoustics, Noise, and Animal Bioacoustics:** Signal Processing Methods for Source Classification and Localization in Real Acoustic Environments II. Room 501 (L)/504 (O)

1:10 4pUWa **Underwater Acoustics, Signal Processing in Acoustics, Physical Acoustics, Acoustical Oceanography, and Computational Acoustics:** Waveguide Invariant Theory and its Applications II. Room 502 (L)/503 (O)

2:20 4pUWb **Underwater Acoustics:** General Topics in Underwater Acoustics: Modeling and Measurement II. Room 502 (L)/503 (O)

Friday Morning

8:00 5aAA **Architectural Acoustics, Computational Acoustics, Education in Acoustics, Physical Acoustics, Noise, ASA Committee on Standards, Engineering Acoustics, and Psychological and Physiological Acoustics:** International Year of Sound Potpourri. Elwha A

9:00 5aABa **Animal Bioacoustics:** Animal Auditory Perception. Quinault

10:30 5aABb **Animal Bioacoustics:** Animal Bioacoustics in Noise. Quinault

8:00 5aAO **Acoustical Oceanography and Underwater Acoustics:** Topics in Acoustic Oceanography. Elwha B

8:30 5aBAa **Biomedical Acoustics:** Acoustics for Kidney Stone Management I. Room 401 (L)/404 (O)

8:00 5aBAb **Biomedical Acoustics and Physical Acoustics:** An Educational Session to Explore the History of Cavitation Bioeffects and How Lessons Learned are Shaping the Future of Cavitation in Medical Ultrasound. Room 405 (L)/406 (O)

8:40 5aPA **Physical Acoustics:** Physical Acoustics Potpourri. Room 402 (L)/403 (O)

9:00 5aPP **Psychological and Physiological Acoustics:** Development and Aging; Clinical Populations (Poster Session). Columbia A

8:20 5aSAa **Structural Acoustics and Vibration:** Novel Methods for Energy Dissipation in Structures. Room 501

10:15 5aSAb **Structural Acoustics and Vibration:** General Topics in Structural Acoustics. Room 501

9:00 5aSC **Speech Communication, Psychological and Physiological Acoustics, ASA Committee on Standards, and Signal Processing in Acoustics:** Speech and Machines. Columbia D

8:00 5aSP **Signal Processing in Acoustics:** General Topics in Signal Processing II. Room 501 (L)/504 (O)

Friday Afternoon

1:00 5pAO **Acoustical Oceanography:** Acoustic Sensing of Bio-Physical Interactions in the Water Column and Seafloor. Elwha B

1:00 5pBA **Biomedical Acoustics:** Acoustics for Kidney Stone Management II. Room 401 (L)/404 (O)

1:00 5pSC **Speech Communication:** Large-Scale and Remote-Platform Acoustic Analysis (Poster Session). Columbia A

181st Meeting of the Acoustical Society of America

The 181st meeting of the Acoustical Society of America will be held Monday through Friday, 29 November-3 December 2021 at the Hyatt Regency Seattle, Seattle, WA, USA.

SECTION HEADINGS

1. COVID-19 VACCINATION REQUIREMENT FOR ATTENDANCE

All meeting attendees are required to show proof that they are fully vaccinated.

To be fully vaccinated means that you have received the number of required vaccination shots for the type of vaccine that you have received.

2. HOTEL INFORMATION

The Hyatt Regency Seattle is the headquarters hotel where all meeting events will be held. The cut-off date for reserving rooms at special rates has passed. Please contact the Hyatt, 808 Howell St., Seattle, WA Hyatt Regency Seattle, 808 Howell Street, Seattle, WA 98101, https://www.hyatt.com/seattle for information about room availability.

3. TRANSPORTATION AND TRAVEL

Visit https://www.portseattle.org/sea-tac/ground-transportation for ground transportation options between the Seattle SeaTac airport and the Hyatt Regency Seattle

4. 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, 29 November, at 7:00 a.m. in the 3rd floor foyer (see floor plan on page A9).

Checks or travelers checks in U.S. funds drawn on U.S. banks and Visa, MasterCard and American Express credit cards 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 $725 for members of the Acoustical Society of America; $875 for non-members, $225 for Emeritus members (Emeritus status pre-approved by ASA), $415 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $175 for ASA Student members, $250 for students who are not members of ASA, $25 for Undergraduate Students, and $225 for accompanying persons.

One-day registration is available at $415 for members and $490 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 $875 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 2022 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 $390 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.

5. ACCESSIBLITY

If you have special accessibility requirements, please indicate this by informing ASA (1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; asa@acousticalsociety.org) at a minimum of thirty days in advance of the meeting. Please provide a cell phone number, email address, and detailed information including the nature of the special accessibility so that we may contact you directly.

6. TECHNICAL SESSIONS

The technical program includes 124 sessions with over 1100 abstracts scheduled for presentation during the meeting.

A floor plan of the Hyatt Regency Seattle appears on page A9 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.

7. 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, 29 November
2-Tuesday, 30 November
3-Wednesday, 1 December
4-Thursday, 2 December
5-Friday, 3 December

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," "b," or "c" 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.

8. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 1 December, at 2:00 p.m. in the Quinault Room on the 4th floor. Papers will be presented on COVID-19 topics.

9. OTHER SPECIAL EVENTS

The Women in Acoustics Committee is hosting a facilitated round-table discussion session from 2:30 p.m. to 3:30 p.m. on Wednesday, 1 December in the Willapa Room. Discussion topics will include navigating careers in academia, government, and industry; mentoring at all levels; work-life balance; and navigating power differentials. While the discussions in this session will focus on women's experiences related to these topics, anyone interested in participating in these discussions is welcome to attend.

A workshop on writing a diversity statement will be sponsored by the Member Engagement Committee and the Committee to Improve Racial Diversity and Inclusivity Committee on Tuesday, 30 November, 10:00 a.m. to 12:00 noon in Columbia B. The workshop will provide strategies for developing a diversity statement. In addition to the workshop at the Seattle meeting, we will hold a virtual follow-up meeting where participants will learn about assessing diversity statements and will be encouraged to bring a draft of their statement for peer feedback. Contact Tessa Bent tbent@iu.edu or Dominique Bouavichith dbouavichith@gmail.com with questions.

The Member Engagement Committee will sponsor a Business Development Panel on Tuesday, 30 November, 4:00 p.m. to 5:30 p.m. in Columbia B. This panel workshop will discuss business development strategies for acousticians who are seeking funding from organizations, such as government offices or consulting firms, whose processes differs from traditional grant writing. Topics covered will include the importance of networking, mentoring, diplomacy, and how to write a statement of work.

A Wikipedia Editing session will be held on Monday, 29 November, 1:00 p.m. to 3:00 p.m. in Columbia D where attendees will discuss and edit Wikipedia. Participants will learn the basics of Wikipedia and how to edit followed by hands-on editing with guidance and support from the experts.

10. MEDWIN PRIZE IN ACOUSTICAL OCEANOGRAPHY AND ACOUSTICAL OCEANOGRAPHY PRIZE LECTURE

The 2020 and 2021 Medwin Prize in Acoustical Oceanography will be presented at the Seattle meeting during the Plenary Session on Wednesday, 1 December.

Dr. Kelly Benoit-Bird will present the 2020 Acoustical Oceanography Prize Lecture titled "A Sound Resolution to the Food Paradox in the Sea" in session 3pAOa, 1:00 p.m. to

2:00 p.m. and Dr. Ana Širović will present the 2021 Acoustical Oceanography Prize Lecture titled "Understanding Marine Life Through Long-Term Passive Acoustic Time Series" in session 3pAOb, 2:15 p.m. to 3:15 p.m. Both lectures will be held in Elwha B on the 5th floor.

11. TUTORIAL ON SOFTWARE BEST PRACTICES

Wu-Jung Lee (Applied Physics Lab., University of Washington, Seattle), Fabio Soares Frazao (Dalhousie University, Halifax, NS, Canada), and Valentina Staneva (University of Washington, Seattle) will present a tutorial titled "Software Best Practices" on Monday, 29 November at 7:00 p.m. in Columbia D on the 3rd level.

Software tools play an increasingly important role in modern day acoustics research. Core software best practices and tools that can help make research processes and outcomes more reproducible and at the same time facilitate collaboration will be shared. Coding best practices with emphasis on tests and documentation, which contribute to code that is more reusable and robust to errors and the use of code and data repositories for collaborative research and show open cloud-based platforms suitable for sharing research outcomes will be covered. The tutorial will include presentation, demos and hands-on exercises, so please bring your laptop to the tutorial. The registration fee is USD $25 (USD $12 for students with current student IDs).

12. SHORT COURSE ON ULTRASOUND CONTRAST AGENTS

A short course on Ultrasound Contrast Agents will be given in two parts: Sunday, 28 November, from 1:00 p.m. to 5:00 p.m. and Monday, 29 November, from 8:30 a.m. to 12:30 p.m. in Room 301.

The instructor is Tyrone Porter, Professor of Biomedical Engineering at The University of Texas at Austin.

13. 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 A8.

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.

14. EXHIBIT

An instrument and equipment exhibition will be located near the registration area and meeting rooms and will open on Monday, 29 November, with an evening reception serving lite snacks and a complimentary drink. Exhibit hours are Monday, 29 November, 5:30 p.m. to 7:00 p.m., Tuesday, 30 November, 9:00 a.m. to 5:00 p.m., and Wednesday, 3 December, 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.

Contact the Exhibit Manager for information about participating in ASA Exhibits: Dan Cooke, Director of Advertising and Exhibit Sales, AIP Publishing, LLC, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, Tel: 516-576-2629; E-mail: dcooke@aip.org.

15. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 1 December, at 3:30 p.m. in Columbia C/D.

ASA scholarship recipients and the David T. Blackstock Mentor Award recipients will be introduced. The Medwin Prize in Acoustical Oceanography, the Silver Medal in Animal Bioacoustics, Silver Medal in Biomedical Acoustics, Silver Medal in Psychological and Physiological Acoustics, Silver Medal in Signal Processing in Acoustics, Silver Medal in Speech Communication, and the Pioneers of Underwater Acoustics Medal will be presented, Certificates will be presented to Fellows elected at the Acoustics in Focus meeting. See page A220 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.

16. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will not be held at the Seattle meeting.

Meetings of selected advisory working groups are often held in conjunction with Society meetings and are listed in the Schedule of Committee Meetings and Other Events on page A8.

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 300, Melville, NY 11747-4300; T.: 631-390-0215; F: 631-923-2875; E: asastds@acousticalsociety.org

17. COFFEE BREAKS

Morning coffee breaks will be held from 9:45 a.m. to 11:00 a.m. near the Exhibit on the 3rd floor. On Tuesday there will be a coffee break near the Exhibit from 2:30 p.m. to 3:30 p.m.

18. A/V PREVIEW ROOM

Room 508 on the 5th floor will be set up as an A/V preview room for authors' convenience and 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 12:00 noon.

19. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Seattle 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.

20. INTERNET ZONE

Wi-Fi will be available in all ASA meeting rooms and spaces.

Tables with power cords will be set up in the 5th Floor Foyer for attendees to gather and to power-up their electronic devices.

21. SOCIALS

Socials will be held on Tuesday and Thursday evenings, 6:00 p.m. to 7:30 p.m. in Columbia C/D.

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

22. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 2 December, at 12:00 noon in Columbia B.

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 1 December. The cost is USD $30.00 per ticket.

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

Follow the student twitter throughout the meeting @ ASAStudents,

A New Students/First-Time Attendee Orientation will be held on Monday, 29 November, from 5:00 p.m. to 5:30 p.m. in Elwha A on the 5th Floor. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in Columbia B on the 1st floor where refreshments and a cash bar will be available.

The Students' Reception will be held on Wednesday, 1 December, from 6:00 p.m. to 8:00 p.m. in Columbia B. This reception, sponsored by the Acoustical Society of America and supported by the National Council of Acoustical Consultants, 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.

To encourage student participation, limited funds are available to defray partially the cost of travel expenses of students to attend Acoustical Society meetings. Instructions for applying for travel subsidies are given in the Call for Papers which can be found online at http://acousticalsociety.org. The deadline for the present meeting has passed but this information may be useful in the future.

24. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 1 December, in Columbia B. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 30 November. The fee is USD $30 for non-students and USD $15 for students.

25. JAM SESSION

You are invited to Columbia C/D on Wednesday night, 1 December, 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.

26. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Seattle meeting. The on-site registration fee for accompanying persons is USD $225. A hospitality room for accompanying persons will be open in Room 409, 8:00 a.m. to 10:00 a.m. Monday through Friday. This entitles you access to the accompanying persons room, social events on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

27. WEATHER

In most years, Seattle averages a daily maximum temperature for November that's between 50- and 55-degrees Fahrenheit (10 to 13 degrees Celsius). The minimum temperature usually falls between 38 and 43 °F (3 to 6 °C). Seattle can receive 4 to 7 inches (114 to 176 mm) of precipitation in November

28. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Aubrey Espana, Technical Program Chair; David Knobles, Acoustical Oceanography; Benjamin Taft, Animal Bioacoustics; Benjamin Bridgewater, David Manley, Architectural Acoustics; Kang Kim, Libertario Demi, Biomedical Acoustics; Jennifer Cooper, Amanda Hanford, Computational Acoustics; Daniel Russell, Education in Acoustics; Michael Haberman, Thomas Blanford, Engineering Acoustics; Chris Elmer, Taffeta Elliott, Musical Acoustics; William Murphy, James Phillips, Hales Swift; Noise; Kevin Lee, Samuel Wallen, Physical Acoustics; Ellen Peng, Virginia Best, Psychological and Physiological Acoustics; John Buck, Kai Gemba, Signal Processing in Acoustics; Christina Zhao, Rajka Smiljanic, Matthew Faytak, Matthew Winn, Speech Communication; Anthony Bonomo, Stephanie Konarski, Structural Acoustics and Vibration; Martin Siderius, Underwater Acoustics; Mallory Morgan, Student Council.

29. MEETING ORGANIZING COMMITTEE

Todd Hefner, Chair; Aubrey Espana, Technical Program Chair; Yashwanth Nanda Kumar, Mohamed Ghanem, Room Monitor Coordinators; Alex Douglass, Guangyu Xu, Signs.

30. PHOTOGRAPHING AND RECORDING

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

31. ABSTRACT ERRATA

This meeting program is Part 2 of the October 2021 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*.

32. 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. Generally, 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 adequately in the allotted time. Four elements to include are:
- Statement of research problem
- Research methodology
- Review of results
- Conclusions
- Generally, 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 very 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.

33. 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 relatively 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 likely 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.

34. GUIDELINES FOR USE OF COMPUTER PROJECTION

A PC computer with monaural audio playback capability and projector will be provided in each meeting room on which all authors who plan to use computer projection should load their presentations. Authors should bring computer presentations on a CD or USB drive to load onto the provided computer and should arrive at the meeting rooms at least 30 minutes before the start of their sessions. Assistance in loading presentations onto the computers will be provided.

Note that only PC format will be supported so authors using Macs must save their presentations for projection in PC format. Also, authors who plan to play audio during their presentations should ensure that their sound files are also saved on the CD or USB drive.

Introduction

It is essential that each speaker who plans to use his/her own laptop connect to the computer projection system in the A/V preview room prior to session start time to verify that the presentation will work properly. Technical assistance is available in the A/V preview room at the meeting, but not in session rooms. Presenters whose computers fail to project for any reason will not be granted extra time.

Guidelines

- Set your computer's screen resolution to 1024x768 pixels or to the resolution indicated by the AV technical support. If it looks OK, it will probably look OK to your audience during your presentation.
- Remember that graphics can be animated or quickly toggled among several options: Comparisons between figures may be made temporally rather than spatially.
- Animations often run more slowly on laptops connected to computer video projectors than when not so connected. Test the effectiveness of your animations before your assigned presentation time on a similar projection system (e.g., in the A/V preview room). Avoid real-time calculations in favor of pre-calculation and saving of images.
- If you will use your own laptop instead of the computer provided, connect your laptop to the projector during the question/answer period of the previous speaker. It is good protocol to initiate your slide show (e.g., run PowerPoint) immediately once connected, so the audience doesn't have to wait. If there are any problems, the session chair will endeavor to assist you, but it is your responsibility to ensure that the technical details have been worked out ahead of time.
- During the presentation have your laptop running with main power instead of using battery power to ensure that the laptop is running at full CPU speed. This will also guarantee that your laptop does not run out of power during your presentation.

Specific Hardware Configurations
Macintosh

Older Macs require a special adapter to connect the video output port to the standard 15-pin male DIN connector. Make sure you have one with you.

- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).
- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024x768 for an XGA projector or at least 640x480 for an older VGA projector. (1024x768 will most always work.). You should also make sure that your monitor controls are set to mirroring.

If it's an older PowerBook, it may not have video mirroring, but something called simulscan, which is essentially the same.

- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep.
- Depending upon the vintage of your system software, you may find that the default video mode is a side-by-side configuration of monitor windows (the test for this will be that you see no menus or cursor on your desktop; the cursor will slide from the projected image onto your laptop's screen as it is moved). Go to Control Panels, Monitors, configuration, and drag the larger window onto the smaller one. This produces a mirror-image of the projected image on your laptop's screen.
- Also depending upon your system software, either the Control Panels will automatically detect the video projector's resolution and frame rate, or you will have to set it manually. If it is not set at a commensurable resolution, the projector may not show an image. Experiment ahead of time with resolution and color depth settings in the A/V preview room (please don't waste valuable time adjusting the Control Panel settings during your allotted session time).

PC

- Make sure your computer has the standard female 15-pin DE-15 video output connector. Some computers require an adaptor.
- Once your computer is physically connected, you will need to toggle the video display on. Most PCS use either ALT-F5 or F6, as indicated by a little video monitor icon on the appropriate key. Some systems require more elaborate keystroke combinations to activate this feature. Verify your laptop's compatibility with the projector in the A/V preview room. Likewise, you may have to set your laptop's resolution and color depth via the monitor's Control Panel to match that of the projector, which settings you should verify prior to your session.

Linux

- Most Linux laptops have a function key marked CRT/LCD or two symbols representing computer versus projector. Often that key toggles on and off the VGA output of the computer, but in some cases, doing so will cause the computer to crash. One fix for this is to boot up the BIOS and look for a field marked CRT/LCD (or similar). This field can be set to Both, in which case the signal to the laptop is always presented to the VGA output jack on the back of the computer. Once connected to a computer projector, the signal will appear automatically, without toggling the function key. Once you get it working, don't touch it and it should continue to work, even after reboot.

35. 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 300, Melville, NY 11747-4300; Telephone: 516-576-2360; Fax: 631-923-2875; E-mail: asa@acousticalsociety.org

182nd Meeting, Denver, Colorado, 22-26 May 2022

183rd Meeting, Nashville, Tennessee, 5-9 December 2022

184th Meeting, Chicago Illinois, 8-12 May 2023

185th Meeting, joint with the Australian Acoustical Society, WESPAC, Sydney, Australia, December 2023

Session 1aAA

Architectural Acoustics and Noise: Back to Basics: Skill-Building Seminar I

Eric Reuter, Chair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

Chair's Introduction—9:10

Invited Papers

9:15

1aAA1. Sound isolation from a consultant's perspective. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

Sound isolation addresses transmission between adjacent spaces as well as non-adjacent spaces, related horizontally and vertically, interior/interior and interior/exterior, through flanking paths and weak links and crosstalk, via both airborne and structure borne "paths." Isolation can be characterized using single-number descriptors, such as Sound Transmission Class (STC) and Impact Insulation Class (IIC), but there are important limitations and caveats. Isolation and methods for improvement are all often misunderstood and misused, especially with an ocean of misinformation and confusion to navigate on the web and elsewhere. This presentation is intended to offer practical, helpful guidance from a consultant's perspective, with focus on fundamentals and simplicity, toward better overall understanding and effective interactions with colleagues and clients.

10:30–10:45 Break

10:45

1aAA2. Mechanical noise and vibration control. Jerry G. Lilly (JGL Acoust.., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

This presentation is intended to provide basic training for entry-level and early-career acoustical consultants in the area of mechanical equipment noise and vibration control in buildings. The seminar will address the following topics: (1) general design guidelines and noise and vibration criteria; (2) basics of fan, pump, and compressor noise generation and vibration isolation; (3) HVAC ductwork and piping design; (4) evaluating manufacturer sound data; (5) laboratory measurement of mechanical equipment; and (6) conducting field measurements and calculations. References to past projects from the speaker's 40 + years of consulting experience will be included to emphasize specific points, as appropriate.

Session 1aAO

Acoustical Oceanography and Underwater Acoustics: Upper Ocean and Mixed Layer Acoustics

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

Andone C. Lavery, Cochair
AOPE, Woods Hole Oceanographic Institution, 98 Water Street, Woods Hole, MA 02543

Chair's Introduction—8:20

Invited Papers

8:25

1aAO1. Sound speed, submesoscale, and spice: The role of rapidly evolving instabilities in setting upper ocean properties.
Jennifer MacKinnon (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., M/C 0213, San Diego, CA 92093, jmackinnon@ucsd.edu)

Sound speed characteristics in the upper ocean are sometimes thought to be set by a combination of mostly one-dimensional vertical mixing processes and (slow) mesoscale advection and stirring. However, a series of recent projects have showcased the role of small-scale (tens of meters to tens of km horizontally), rapidly evolving (hours to days) submesoscale instabilities in setting both stratification and sound speed properties. The observations also reveal that sound speed structure set by such processes is fundamentally three dimensional, with significant lateral gradients at a broad range of scales. Some relevant processes are frontogenetic and create sometimes rapid secondary circulations near lateral density gradients. Some are associated with wind-driven near-inertial oscillations or internal waves. Some of the more interesting recent observations concern the intersection of submesoscale and internal wave processes. Here, I will review a few highlights from the last decade of observations on unbalanced, nonlinear processes that conspire to set density, spice, and sound speed profiles in several contrasting oceans.

8:45

1aAO2. Mixed layers and surface ducts.
Michael B. Porter (Heat, Light, and Sound Res., Inc., 12625 High Bluff Dr., STE 211, San Diego, CA 92130-2054, Porter@HLSResearch.com)

The upper ocean is strongly effected by the sun and the wind. In an idealized view, this leads to a warm mixed layer with a nearly constant temperature and salinity. Then, the increase in pressure with depth causes an increase of about 1.7 m/s in sound speed for every 100 m in depth (i.e., a gradient of 0.017/s). This gradient forms an acoustic surface duct, which is a nearly universal feature of the world's oceans. They can be just tens of meters thick or several hundred meters in the Southern Ocean or the North Atlantic during periods of winter storms. Not surprisingly, the surface duct is very important in terms of sound propagation; However, as we shall describe, the surface duct has some peculiar acoustic properties and can produce vast shadow zones inside the duct. This simplified description is a good beginning; however, "mixed" layers are generally not well-mixed. It is more precise to speak of the upper ocean as a zone where there may be periods of mixing and heating driven by changes in winds and cloud cover. Upper ocean models predict this evolution and often lead to much more complicated sound speed structures. This, in turn, has lead to a wide variety of ways of characterizing the features of the surface duct. This talk will touch on the interesting acoustics of surface ducts and how variations in different (oceanographic) upper ocean models effect the sound propagation.

Contributed Papers

9:05

1aAO3. Sensitivity of mixed layer duct propagation to deterministic ocean features: The interaction parameter.
John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943, jacolosi@nps.edu) and William Zinicola-Lapin (Oceanogr., Naval Postgrad. School, Monterey, CA)

Surface duct propagation can be affected by deterministic ocean features such as fronts, eddies, filaments, and/or density compensated temperature and salinity anomalies (spice). Using the concept of duct modal structure and the Dyson series solution to the wave equation, we present a non-dimensional parameter Γ_{mn}, termed the interaction parameter, that quantifies the strength of mode coupling between a surface duct mode m and any other mode n as a function of ocean feature scales and acoustic frequency. Weak, first order coupling is given by $\Gamma_{mn} < 1$ where as strong, higher order coupling has $\Gamma_{mn} > 1$. Importantly, there is a frequency dependent resonance condition associated with the range width of the perturbations, Δ, such that Γ_{mn} goes to zero as Δ approaches 0 and infinite. The utility of the interaction parameter in estimating the stability of mixed layer propagation is demonstrated using numerical examples of a cold eddy and of a hot and salty spice feature.

9:20

1aAO4. Mixed layer acoustic propagation with small scale ocean dynamics. Edward Richards (Ocean Sci., UC Santa Cruz, 1156 High St., Santa Cruz, CA 95064, edwardlrichards@gmail.com) and John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA)

The effects of dynamic oceanographic processes in the upper ocean mixed layer on acoustic propagation are investigated using at-sea sound speed observations and acoustic transmission simulations. The sound speed environment is based on a 400 m deep and 1000 km long East-West salinity and temperature transect that was collected during April of 2005 in the North Pacific where there is a mixed layer acoustic duct with a mean depth of 89 m. The deterministic baseline of this sound speed measurement is first estimated from the observations using a spatial lowpass filter with a cut off length scale of 50-km. The remaining small length scale variations in sound speed are assumed to have two physically distinct contributions: one due to the vertical displacement of isopycnals caused by eddies and internal waves and another due to compensating temperature and salinity anomalies along isopycnals termed spice. These two contributions can be separated yielding three distinct sound speed fields: i(1) the deterministic background; (2) the displacement field; and (3) the spice field. Acoustic transmission loss calculations are carried out on these three fields for frequencies of 400 and 1000 Hz for sources both in and below the mixed layer duct. Statistical and deterministic results for acoustic normal modes and full field transmission loss are presented, which quantify the relative importance of displacement and spice for acoustic propagation in the upper ocean.

9:35

1aAO5. The impact of internal tides and gravity waves on key acoustic parameters and acoustic transmission loss near the Amazon shelf. Martha Schönau (Appl. Ocean Sci., LLC, 11006 Clara Barton Dr., Fairfax Station, VA 22039, martha.schonau@appliedoceansciences.com), Emanuel Coelho (Appl. Ocean Sci., LLC, Fairfax Station, VA), Robert Helber, Jay Shriver (Code 7321, Naval Res. Lab., Stennis Space Ctr., MS), and Kathryn Verlinden (Appl. Ocean Sci., LLC, Fairfax Station, VA)

Upper ocean variability depends on tides and wind forcing. The barotropic tide creates baroclinic internal tides and internal gravity waves (IGWs) that can propagate long distances and dissipate, impacting mixed layer and thermocline depths, vertical thermal gradients, and mixing. In the last decade, there has been an effort to include tidal motions in global HYCOM (HYbrid Coordinate Ocean Model). Here, we look at the impacts of tidally driven motions on acoustic propagation along the Amazon shelf by comparing acoustic transmission loss and key acoustic parameters in 1/25 deg Global HYCOM model runs with and without tides. Acoustic runs were performed at mid-frequencies using Bellhop3D at depths above and below the sonic-layer depth (SLD). Key acoustic parameters assessed include the SLD and vertical sound-speed gradients. Although the impacts of the internal waves and tides on acoustics are intermingled with the effects of other stochastic ocean processes (e.g., mesoscale eddies, fronts, and atmospheric forcing), timeseries analysis indicates that there is acoustic sensitivity to tidally forced variability. These ocean-acoustic results can be used for future development of machine learning algorithms to model coupled oceanographic and acoustical processes.

9:50

1aAO6. Physics-informed neural networks for predicting ocean spatio-temporal fields. Kayla Thilges (Appl. Res. in Acoust. LLC, 1500 Westlake Ave. N, Ste 001, Seattle, WA 98109, kayla.thilges@ariacoustics.com), Meredith Plumley (Appl. Res. in Acoust. LLC, Seattle, WA), Jason E. Summers (Appl. Res. in Acoust. LLC, Madison, VA), Brian K. Arbic (Dept. of Earth and Environ. Sci., Univ. of Michigan, Ann Arbor, MI), and M. C. Buijsman (Div. of Marine Sci., Univ. of Southern MS, Stennis Space Ctr., MS)

Simulations of the ocean environment are important for assessing acoustic variability due to large and small scale changes in the propagation medium. By incorporating the Internal Gravity Waves (IGWs) partial differential equation (PDE) in ocean models, such as the HYbrid Coordinate Ocean Model (HYCOM) system, it is possible to model the impact of tides on acoustic propagation. This requires HYCOM simulations of the ocean environment, which are computationally expensive, especially at the high vertical resolutions required to capture IGWs. In this work, we develop a machine-learning approach to predicting the effects of IGWs without incurring the computational cost of high-resolution numerical modeling. We apply a deep-learning approach in the form of a Physics Informed Neural Network (PINN) that incorporates the IGW PDE system to make predictions about oceanographic quantities of interest with internal tides at high vertical resolution. Training data comprise HYCOM 4D fields (time, depth, latitude, and longitude) both with (IGW-HYCOM) and without tides in the Pacific subtropical/equatorial region. The developed framework is able to map a relationship between lower-resolution HYCOM inputs to highly resolved ocean fields that include internal tides.

10:05–10:20 Break

Invited Paper

10:20

1aAO7. Bubble effects on upper ocean acoustics under wind-driven seas. Grant B. Deane (Scripps Inst. Of Oceanogr., Code 0238, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu), Hari Vishnu, and Mandar Chitre (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore)

Dense clouds of bubbles—ranging in scale from 10s of micrometers to cm—are generated by breaking waves on wind-driven seas. Wave-induced bubbles increase ocean albedo and play an important role in air-sea exchange processes by enhancing the transfer of gases and generating sea spray aerosol. Because bubbles both scatter sound efficiently and generate pulses of sound on formation, they create important acoustical effects, including the generation of ambient sound, the attenuation and scattering of sound forward scattered from the sea surface, and enhanced surface backscattering at low grazing angles. Examples of these effects will be illustrated along with the treatment of a bubble cloud acoustical scattering regime characterized by low absorption but high phase distortion relevant to high-frequency, coherent acoustic communications. [Work supported by ONR Ocean Acoustics.]

10:40

1aAO8. Observations of high-frequency acoustic attenuation due to bubble entrainment at estuarine fronts. Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, NJ 98105, cbassett@uw.edu) and Andone C. Lavery (Woods Hole Oceanographic Inst., Woods Hole, MA)

Frontal boundary features with strong gradients in water properties and velocity are common features in coastal environments where fresh discharge meets ambient ocean waters. Convergence at these fronts results in two features that support the entrainment of bubbles. First, horizontal velocity gradients causes surface waves to steepen and break at these boundaries. Second, localized downwelling can further entrain and transport these bubbles. In experiments using broadband echosounders (45–420 kHz) at the Connecticut River ebb plume front and James River tidal intrusion front, bubble entrainment was observed to cause frequency-dependent excess attenuation. Seabed backscattering near fronts is used as a baseline, whereas a decrease in observed backscattering from the seabed under bubble plumes was attributed to excess attenuation due to bubble concentrations. Despite observing significant (>10 dB in some cases) excess attenuation in lower-frequency channels, we hypothesize that these fronts are unlikely to significantly affect propagation in the horizontal at these frequencies and that factors, such as refraction and three-dimension structure, are more operationally significant. [This work was funded by ONR.]

10:55

1aAO9. Predictions of acoustic backscattering from oceanic stratification interfaces. Elizabeth Weidner (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, eweidner@ccom.unh.edu) and Thomas C. Weber (Univ. of New Hampshire, Durham, NH)

The ocean is primarily composed of a series of increasingly dense fluid layers, defined by density stratification interfaces where medium properties, such as temperature and salinity, drive changes in medium density and compressibility. The transport of dissolved constituents (e.g., oxygen, heat, salt) between these oceanic fluid layers is strongly influenced by the intensity of the stratification interface. Active acoustics allow for high resolution observations of these interfaces, offering continuous data collection over broad spatial scales. In this work, we present a one-dimensional acoustic scattering model for predicting acoustic backscatter from stratification interfaces. The model is widely applicable to acoustic water column data collected with ship-mounted sonars and preliminary results, based on hydrographic profiles, suggest that, in many oceanic cases, sound speed perturbations drive the majority of acoustic scattering. Additionally, modeling results predict a frequency modulated backscattering intensity based on the sharpness of the stratification interface, suggesting the potential for remote estimation of medium properties through broadband acoustic inversion.

11:10

1aAO10. Observations of 11 to 12.5 kHz sea ice pulse reflections around 50–55 deg grazing angle in the Beaufort Sea from summer 2016 to summer 2017. Murat Kucukosmanoglu (Ocean Sci., Univ. of California, Santa Cruz, 604 Koshland Way, Santa Cruz, CA 95064, mkucukos@ucsc.edu), John A. Colosi, Christopher W. Miller (Oceanogr., Naval Postgrad. School, Monterey, CA), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Derek Olson (Oceanogr., Naval Postgrad. School, Monterey, CA), and Edward Richards (Ocean Sci., Univ. of California, Santa Cruz, Santa Cruz, CA)

Beaufort Sea ice has acoustic characteristics that change across time and space. This research aims to quantify sea ice scattering statistics with the goal of using them to predict physical ice characteristics. Over the time between summer 2016 and summer 2017, ice and ocean surface scattering in the 11–12.5 kHz frequency range between grazing angles 50 deg and 55 deg was studied using the navigation systems for a seven mooring 150-km radius acoustic array. Surface scattering has five primary epochs: open water (OW), early ice formation (IF) occurring for a few weeks, ice solidification (IS) occurring again for a few weeks, ice thickening (IT) occurring for about six months, and eventually, ice melting (IM) occurring for 1.5 to 2 months. Important changes in acoustic scattering behavior occurs between these epochs, here expressed in terms of the moments of reflected intensity, intensity probability density function, and pulse time spread. The eras of most considerable change in the observables belong to transition epochs of IF, IS, and IM. These results are interpreted physically and qualitatively in terms of the notions of partially and fully saturated wave fields, an incoherent Kirchhoff-like approximation for the rough surface, and a thin elastic layer reflection coefficient model.

Session 1aBAa

Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics: Cavitation Nuclei: Bubbles, Droplets, and More I

Stuart Ibsen, Cochair
Biomedical Engineering, Oregon Health and Science University, 3181 S.W. Sam Jackson Park Rd., KCRB 5001.41, Portland, OR 97239

Michaelann Tartis, Cochair
Chemical Engineering, New Mexico Institute of Mining and Technology, 801 Leroy, Socorro, NM 87801

Chair's Introduction—8:00

Invited Papers

8:05

1aBAa1. Lipid-stabilized hydrophobically modified silica-coated gold nanorods generate enhanced photoacoustic signal from cavitation with nonlinear dependence on laser fluence. Andrew Goodwin (Chemical and Biological Eng., Univ. of Colorado Boulder, 3415 Colorado Ave., 596 UCB, Boulder, CO 80303, andrew.goodwin@colorado.edu)

In this work, we report that gold nanorods coated with hydrophobically modified mesoporous silica shells not only enhance photoacoustic (PA) signal by more than 10-fold over unmodified mesoporous silica coated gold nanorods but also the relationship between PA amplitude and input laser fluence is strongly nonlinear. Mesoporous silica shells of $\sim$14 nm thickness and with $\sim$3 nm pores were grown on gold nanorods showing near infrared absorption. The silica was rendered hydrophobic with addition of dodecyltrichlorosilane, then re-suspended in aqueous media with a lipid monolayer. Analysis of the PA signal revealed not only an enhancement of PA signal compared to mesoporous silica coated gold nanorods at lower laser fluences but also a nonlinear relationship between PA signal and laser fluence. We attribute each effect to the entrapment of air in the mesopores: the air has a larger expansion coefficient than silica that enhances conversion to acoustic energy, and the air undergoes thermal size oscillations that increase with laser fluence, leading to a nonlinear trend even at fluences as low as 5 mJ cm^{-2}. At 21 mJ cm^{-2}, the highest laser fluence tested, the PA enhancement was >12-fold over mesoporous silica coated gold nanorods.

8:25

1aBAa2. Hemoglobin microbubbles for *in vivo* blood oxygen level dependent imaging: Boldly moving beyond MRI. Sugandha Chaudhary, Nasrin Akter, Akshay Rajeev (Bioengineering, UT Dallas, Richardson, TX), Misun Hwang (Dept. of Radiology, Childrens Hospital of Philadelphia, Philadelphia, PA), and Shashank Sirsi (Bioengineering, UT Dallas, 800 W. Campbell Rd., Richardson, TX 75080, shashank.sirsi@utdallas.edu)

Microbubbles are gas-filled, micron-sized particles stabilized by lipid, protein, or polymer shells. They scatter ultrasound energy efficiently due to their compressible gas cores, making them excellent blood pool contrast agents in ultrasound imaging. In this project, we have developed a novel oxygen-sensitive hemoglobin-shell microbubble designed to acoustically detect blood oxygen levels *in vivo*. We hypothesized that structural changes in the hemoglobin in response to varying oxygen levels will alter the mechanical properties of the bubble shell, resulting in detectable changes in the bubble acoustic signature. In this project, we have (1) optimized the hemoglobin bubble formulation for *in vivo* circulation, (2) demonstrated that the hemoglobin shell is still responsive to oxygen after formulation, and (3) characterized the acoustic response of the microbubbles at varying oxygen levels. Our preliminary results show a strong correlation between the oxygen concentration in the solution and the acoustic response of bubbles, suggesting that they would serve as excellent oxygen sensors. If successful, *in vivo* sensing of oxygen levels would have numerous benefits, including evaluating the hypoxic regions in tumors and in the brain, among other blood-oxygen-level-dependent (BOLD) imaging applications.

8:45

1aBAa3. Carbocyanine dye-patterned microbubbles as ultrasound-modulated fluorescent contrast agents. Carolyn Schutt Ibsen (Biomedical Eng. and Knight Cancer Inst. Cancer Early Detection Adv. Res. Ctr., Oregon Health and Sci. Univ., 2720 SW Moody Ave., KR-CEDR, Portland, OR 97201, ibsenc@ohsu.edu), Michael Benchimol, Mark Hsu (Elec. and Comput. Eng., Univ. of California, San Diego, San Diego, CA), and Sadik Esener (Biomedical Eng. and Knight Cancer Inst. Cancer Early Detection Adv. Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Combining fluorescence and acoustic techniques enables the characterization of biochemical environments with optical sensors while utilizing the penetration depth and spatial resolution of focused ultrasound. Localized light generation within a small tissue volume can yield spatially resolved chemical information useful for delineating tissue pathology such as cancerous lesions. Our work develops fluorescent microbubble contrast agents that generate light in the focal zone of an ultrasound beam through modulation of their fluorescence intensity. Microbubbles were designed with a self-quenching lipophilic dye patterned within the lipid layer, enabling the dye molecules to quench and dequench with the ultrasound-driven particle size oscillations. Distinct dye patterns were generated on the microbubble surfaces and investigated for their ultrasound-driven fluorescence response. By amplifying the microbubble fluorescence intensity modulations using a lock-in amplifier, the optical signal from these locally activated particles was detected within an optically scattering environment and at significant depth. We additionally observed that these microbubbles also displayed harmonic oscillations of fluorescence intensity beyond the ultrasound driving frequency which could be used to further improve the signal-to-noise ratio for detection. This technique could enable sensitive optical imaging with ultrasound-scale millimeter-level spatial resolution, providing a valuable tool to address the challenge of optical imaging in deep tissue.

9:05

1aBAa4. A novel microbubble platform for cancer immunotherapy. Sina Khorsandi (Radiology, UT Southwestern Medical Ctr., Dallas, TX), Xuefeng Li, Yifan Wang (Radiation Oncology, MD Anderson Cancer Ctr., Houston, TX), Robert Mattrey (Radiology, UT Southwestern Medical Ctr., Dallas, TX), Wen Jiang (Radiation Oncology, MD Anderson Cancer Ctr., Houston, TX), and Jacques Lux (Radiology, UT Southwestern Medical Ctr., 5323 Harry Hines Blvd, Dallas, TX 75390-8514, jacques.lux@utsouthwestern.edu)

The innate immune sensor cyclic GMP–AMP synthase–stimulator of interferon genes (cGAS-STING) has recently emerged as a potential therapeutic target to boost antitumor immune responses. However, STING is a cytoplasmic protein, and its natural activator cGAMP is a negatively charged dinucleotide that is difficult to deliver intracellularly. We have developed a new platform called **M**icrobubble-assisted **U**ltra**S**ound (US)-guided **I**mmunotherapy of **C**ancer (MUSIC) to activate STING with spatio-temporal control to treat cancer. MUSIC showed greater STING activation compared to cGAMP alone *in vitro* . In murine models, MUSIC showed dramatic tumor growth inhibition and increased survival in both syngeneic breast cancer models. The absence of antitumor immune responses in $STING^{-/-}$ mice shows that MUSIC action is STING-dependent. Furthermore, 6/10 MUSIC-treated mice were tumor-free versus 2/10 cGAMP-treated mice. All the rechallenged MUSIC-treated mice remained tumor-free 30 days after rechallenge. MUSIC also resulted in a 7-fold decrease in metastatic burden when compared to cGAMP alone. In summary, MUSIC showed efficient STING activation *in vitro*. In addition, local MUSIC treatment of primary tumors produced systemic antitumor immune responses *in vivo* as well as anti-cancer immune memory preventing tumor growth upon rechallenge. We are now exploring the use of nanobubbles for systemic administration. [Work supported by CPRIT Grants RR150010 and RP190233. R.F.M. is a CPRIT Established Investigator.]

Contributed Papers

9:25

1aBAa5. The crevice model of cavitation inception. Lawrence A. Crum (Appl. Phys. Lab, Univ. of Washington, 4662 175th Ave. SE, Bellievue, WA 98006, lacuw@uw.edu) and Julianna Simon (Graduate Program in Acoust., Penn State Univ., University, PA)

In Robert Apfel's Ph.D. dissertation and in subsequent published articles, he demonstrated that if one chose a volume small enough—say a microdroplet—cavitation could not be generated even at elevated temperatures. He expanded the work of Harvey (*J. Cell. Comp. Physiol.* **24**, 23 (1944) who proposed the crevice model of cavitation nucleation. Recently, Lu demonstrated that the phenomenon of twinkling was due to the presence of trapped bubbles on solid concretions (Lu, *et al.*, *U. S. Med. Biol.* **39**, 1026 (2013). Subsequently, Simon demonstrated that twinkling occurred in humans (Simon, *et al.*, *U. S. Med. and Biol.* **46**, 1802 (2020). In this presentation, the crevice model will be reviewed and asserted that it is the principal model for inception of cavitation in mammalian tissue.

9:40

1aBAa6. The impact of lipid shell composition of perfluorocarbon nanodroplets on size distribution and acoustic droplet vaporization and cavitation dynamics. Phoebe J. Welch (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA, pwelch8@gatech.edu), Craig R. Forest, and Chengzhi Shi (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Perfluorocarbon nanodroplets are ultrasound contrast agents that phase transition from liquid to gas when activated by lasers or insonated with a high intensity ultrasound pulse. The dynamics of perfluorocarbon nanodroplets can vary drastically depending on the nanodroplet shell and/or core composition. In this work, we investigated the role of varying lipid shell compositions and how this impacts the acoustic droplet vaporization and inertial cavitation dynamics of these nanodroplets when insonated with high intensity focused ultrasound. We also examined the role of lipid shell composition on the size distribution of these nanodroplets. Nanodroplets with a higher concentration of PEGylated lipids had larger diameters and exhibited greater variance in size distribution. Perfluorocarbon nanodroplets with a lipid shell composed of 50:50 PEGylated to non-PEGylated lipids yielded the highest intensity and duration ultrasound signal as well as the greatest pressure differential between acoustic droplet vaporization onset and inertial cavitation onset. These findings can help guide researchers to fabricate perfluorocarbon nanodroplets with optimized parameters for their specific applications, including ultrasound imaging, drug delivery, and tissue ablation.

9:55–10:10 Break

10:10

1aBAa7. Relationship between salinity of the liquid and shell composition on the resonance frequency of the C3F8 lipid coated microbubbles. Amin Jafarisojahrood (Physical Sci., Sunnybrook Health Sci. Ctr., 77 Harbour Square, Apt. 2103, Toronto, NS M5J 2S2, Canada, amin.jafarisojahrood@ryerson.ca), Celina Yang (Case Western Univ., Toronto, ON, Canada), Claire Counil, Pinuta Nittayacharn, A. l. C. deLeon, Agata Exner (Case Western Univ., Cleveland, OH), David Goertz (Physical Sci., Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), and Michael Kolios (Phys., Ryerson Univ., Toronto, ON, Canada)

Lipid-coated microbubbles (MBs) are used in contrast-enhanced ultrasound (CEUS) imaging and MB enhanced therapeutic ultrasound (US). Understanding the MB behavior and the influence of the surrounding medium on its response to the US is necessary to select the suitable US exposure parameters. The MB lipid coating is often charged, however the influence of the ions in the medium on MB behavior is not fully understood. In this work, the influence of the medium salinity on the pressure-dependent MB behavior is investigated for the first time. MBs of different lipid shell compositions are size isolated to achieve the same size distribution. The MBs linear and pressure-dependent attenuation are measured in deionized water, PBS 1×, PBS 2×, and PBS 10× using a system of aligned PVDF 100% bandwidth transducers with a center frequency of 10 MHz and exposures with peak to peak pressure range of 3–140 kPa. With increasing salinity, the linear resonance frequency decreases up to 50% for conventional lipid shell compositions, and the pressure dependence of the resonance frequency is inhibited. By modifying the shell PEG ratio, the salinity effects can significantly be altered. Moreover, the nonlinear pressure-dependent resonance frequency is restored with applications to increased CEUS.

10:25

1aBAa8. Ultrafast dynamics of acoustic vaporization and payload release of phase-shift emulsions. Mitra Aliabouzar (Univ. of Michigan, 3218-02 Med. Sci. I, 1301 Catherine St., Ann Arbor, MI 48109, aliabouz@med.umich.edu), Oliver D. Kripfgans, Jon B. Estrada, J. Brian Fowlkes, and Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI)

Acoustic droplet vaporization (ADV) is the phase-transitioning of perfluorocarbon emulsions via focused ultrasound. ADV has been utilized in many biomedical applications. For drug delivery and tissue regeneration, we developed composite hydrogel scaffolds, termed acoustically responsive scaffolds (ARSs), which here contained fluorescently labeled monodispersed emulsions (Ø: 12.3 ± 0.8 μm). Using ultra-high speed microscopy, we investigated the dynamics of ADV (2.5 MHz, P_-: 6 MPa, pulse length: 6 μs) in ARSs containing emulsions with perfluoropentane, perfluorohexane, and perfluorooctane. Vaporization and growth dynamics were studied in micro- to millisecond time scales at frame rates up to 10 Mfps. During ADV, the generated bubbles expanded 4.8-, 2.8-, and 2-fold greater than the initial diameter of the perfluoropentane, perfluorohexane, and perfluorooctane emulsions, respectively. Stable bubbles formed from ADV in perfluoropentane and perfluorohexane, though growth rates post-ADV were significantly different. Bubbles recondensed within perfluorooctane emulsions, which exhibited a 20% decrease in diameter following five repeated ADV events. Stable or transient bubble formation was further compared with classical nucleation theory predictions. Additionally, we compared the release dynamics of fluorescently labeled payloads from the emulsions. The results provide physical insight enabling the modulation of bubble dynamics with ADV and hence release kinetics, which can be used for therapeutic applications.

10:40

1aBAa9. Cavitation dynamics of histotripsy bubble cloud nucleated from echogenic liposomes. Aarushi Bhargava (Dept. of Radiology, Univ. of Chicago, 208 Broce Dr., Apt. 1, Blacksburg, VA 24060, aarushib@uchicago.edu), Shaoling Huang, David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Center-Houston, Houston, TX), and Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, Chicago, IL)

Deep vein thrombosis is a serious health concern worldwide. Administration of catheter-directed thrombolytics is the primary approach to achieve vessel patency for critical obstructions. Adjuvant techniques that localize thrombolytic delivery will improve treatment outcomes and reduce the potential caustic bleeding associated with systemic lytic administration. Histotripsy is a bubble-based focused ultrasound therapy that has been shown to act synergistically with lytic to promote clot degradation. Combining histotripsy with a targeted drug delivery agent may further improve treatment efficacy. Echogenic liposomes are vesicles formulated to entrain the thrombolytic, and can be triggered with ultrasound to release drugs locally. The objective of this study was to assess histotripsy bubble cloud dynamics nucleated from echogenic liposomes in an *in vitro* venous flow model. The change in liposome echogenicity following histotripsy exposure was assessed with high frame rate imaging. Hypoechogenicity indicative of liposome destruction extended beyond the focal region, with a strong decrease in pixel intensity immediately after the applied histotripsy pulse. Interestingly, echogenicity was restored to baseline levels over the course of ∼4 s, which was attributed to vigorous fluid mixing from bubble cloud oscillations. These results indicate an interesting range of bubble cloud behaviors when histotripsy is combined with echogenic liposomes.

10:55

1aBAa10. Bubble cloud progression in fibrous tissue mimicking hydrogels at different histotripsy sonications. Yashwanth Nanda Kumar (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynandaku@uw.edu), Zorawar Singh (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Yak-Nam Wang (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Wayne Kreider, Matthew Bruce (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Eli Vlaisavljevich (Dept. of Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), Tatiana D. Khokhlova (Dept. of Gastroenterology, Univ. of Washington School of Medicine, Seattle, WA), and Adam Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Polyacrylamide-alginate hydrogels (85/15, 90/10, 95/5 by weight ratio) have been developed with similar mechanical and acoustic properties to human benign prostatic hyperplasia tissue. Although the phantoms are similar to other gels in stiffness, they have much greater toughness and require >10 times the dose to liquify by histotripsy. The purpose of this study was to evaluate differences in bubble dynamics and bubble cloud behavior with different histotripsy sonication in the gels, and how these might affect lesion formation. All gels were treated with a 700 kHz histotripsy transducer at pulse repetition frequencies of 10 or 500 Hz and pulse-durations of 20 or 3 cycles at identical acoustic pressures P+ 108 MPa, P− 20 MPa. The bubble cloud generated over 2000 pulses was captured using high-speed camera and were analyzed for bubble density and temporal changes from pulse to pulse. In low-toughness agarose gel, there was a dense cloud formation, leading to a defined lesion as pulse count progressed. In the polyacrylamide-alginate gel, a dense focal cloud surrounded by a sparse uniform cloud towards the transducer was observed throughout, with both PRFs. This sparse cloud potentially explains the increased dosage required to treat hybrid gels. [Work supported by R01-DK119310 and L30-DK122509.]

11:10

1aBAa11. Static magnetic fields dampen focused ultrasound-induced cavitation and blood-brain barrier opening outcome. Yaoheng Yang (Dept. of Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO 63108, mack.yang@wustl.edu), Christopher P. Pacia, Dezhuang Ye, Yimei Yue, Chih-Yen Chien, and Hong Chen (Dept. of Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO)

MRI-guided focused ultrasound combined with microbubbles has been used in clinical studies for blood-brain barrier (BBB) opening. However, the impact of the static magnetic field generated by an MRI scanner on the BBB opening outcome has not been evaluated. The objective of this study was to determine the relationship of the static magnetic field of an MRI scanner on focused ultrasound combined with microbubble-induced BBB opening. Mice were divided into four groups which were sonicated by FUS in different static magnetic fields B_0 (∼0T, 1.5T, 3.0T, and 4.7T) with all other experimental parameters kept the same. Microbubble cavitation activity was

monitored by passive cavitation detection during FUS after microbubble injection. After sonication, contrast-enhanced T1-weighted MRI and Evans blue were used to evaluate the BBBO outcome. The microbubble cavitation dose decreased by an average of 2.1 dB at 1.5T ($P = 0.05$), 2.9 dB at 3.0T ($Pi = 0.01$), and 3.0 dB at 4.7T ($P = 0.01$) compared with that outside the magnetic field. It also decreased Evans blue trans-BBB delivery 1.4-fold at 1.5T ($P = 0.009$), 1.6-fold at 3.0T ($P < 0.001$), and 1.9-fold at 4.7T ($P < 0.001$). These findings demonstrated that static magnetic fields dampened microbubble cavitation and decreased trans-BBB delivery by FUS.

11:25

1aBAa12. High intensity focused ultrasound and microbubbles induce targeted mild hyperthermia suitable for enhanced drug delivery. Eric Juang (Bioengineering, Univ. of Washington, Seattle, WA), Lance H. De Koninck (Bioengineering, Univ. of Washington, 1400 NE Campus Parkway, Seattle, WA 98195, lancedek@uw.edu), Aswin Gnanaskandan (Worcester Polytechnic Inst., Worcester, MA), Chao-Tsung Hsiao (Dynaflow, Inc., Jessup, MD), and Michalakis A. Averkiou (Bioengineering, Univ. of Washington, Seattle, WA)

High intensity focused ultrasound (HIFU) can elevate the temperature leading to tissue ablation (>60 °C). Mild hyperthermia (39–45 °C) can enhance drug delivery such as in the activation of thermosensitive liposomes for chemotherapy and in radiotherapy. The addition of microbubbles during HIFU could greatly enhance the heating process and increase the treatment area while operating at lower intensities. Here, we present a parametric study of microbubble-enhanced mild hyperthermia with HIFU in *ex vivo* porcine liver. A 30-s treatment at 0.9 MHz, 372 000 cycles, 82% duty cycle was used, and at estimated non-derated (water) pressures of 0.63, 1.25, 2.50 and 3.75 MPa. The microbubbles were injected locally in the treatment area. The measured temperature rise was greater with microbubbles than without at low acoustic pressures (0.63 and 1.25 MPa). At higher pressures (2.50 and 3.75 MPa), the presence of microbubbles did not have an effect on the measured temperature rise possibly due to inertial cavitation. Our results agree with predictions from a numerical model that solves the coupled acoustic-bubble flow fields with a two-way coupled Euler–Lagrange model. Future work will utilize the numerical model and new measurements to optimize targeted mild hyperthermia before applying it to enhanced drug delivery.

11:40

1aBAa13. Reliable and safe blood–brain barrier opening by closed-loop feedback control of focused ultrasound. Chih-Yen Chien (Biomedical Eng., Washington Univ., Saint Louis, MO, c.chien@wustl.edu), Yan Gong, Yaoheng Yang, Yimei Yue, and Hong Chen (Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO)

Real-time cavitation monitoring and feedback controller is critical to safe and effective brain drug delivery by focused ultrasound (FUS)-mediated blood-brain barrier disruption (FUS-BBBD). However, existing controllers were either open-loop or using a generalized target cavitation level (TCL). This study aimed to develop a closed-loop feedback controller with TCL defined based on the baseline stable cavitation level (SCL) of each subject. A single-element FUS transducer with a coaxially aligned passive cavitation detector (PCD) was used. PCD signals were first acquired with "dummy" sonication to determine TCL defined to be 0.5, 1, 2, 3, or 4 dB above baseline SCL. A custom closed-loop feedback controller was developed to control SCL to be at each TCL. After sonication, wild-type mice were intravenously injected with Evans Blue (EB) to quantify FUS-BBBD using *ex vivo* fluorescence imaging. The fluorescence intensity of delivered EB increased in an average of 1.4-fold, 2.6-fold, and 3.9-fold at 1, 2, and 3 dB respectively compared to the 0.5-dB group. No apparent hemorrhage was found in lower TCLs, and hemorrhage was consistently found in the 4-dB group. This study demonstrated that the individualized feedback controller achieved reliable and safe FUS-BBBD at selected TCL.

Session 1aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics I

Tatiana D. Khokhlova, Chair
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Contributed Papers

8:30

1aBAb1. Ultrasonic intra-body communication using semi-guided waves through human body tissues. John O. Gerguis (Elec. and Comput. Eng., Purdue Univ., 516 Northwestern Ave., West Lafayette, IN 47906, jgerguis@purdue.edu), Mayukh Nath, and Shreyas Sen (Elec. and Comput. Eng., Purdue Univ., West Lafayette, IN)

Body Area Network (BAN) demands a secure and low-power technology. Traditionally, RF signals were used, which suffer from high losses and lack of security. Recently, electro-quasistatic human body communication has emerged, enabling low-power communication, but somewhat susceptible to leakage. Ultrasound is presented as a promising alternative owing to its ability of propagating in water-dominated media, like the human body, besides being safe. Existing studies employ ultrasound devices for powering or communicating across tissues; either through transmitter-receiver aligned configurations (not suitable for non-line-of-sight communication) or using omnidirectional transducers, hence limiting the communication distance to only a few centimeters. This paper presents a theoretical study on confining ultrasound waves inside the human tissues (muscle, fat and skin). Through oblique incidence on the bone/muscle interface, the waves can bypass the bone (highly attenuative tissue) through total internal reflection, and be guided through water-rich body tissues. In addition, the high reflection at the skin/air interface ($\sim$99.9%) confines most of the signal inside the body, ensuring secure communication. Simulations are performed on a simplified cylindrical-shaped human body model at 100 kHz, demonstrating the possibility of transmitting a signal over a distance of $\sim$1 m with losses $<$50 dB and leaked signal attenuation of additional $>$20 dB.

8:45

1aBAb2. Micro-elastography on spheroids using a magnetic pulse as a shear wave source and the impact of cavitation on their mechanical properties. Gabrielle Laloy-Borgna (INSERM U1032, 151 Cours Albert Thomas,69428, Lyon, France, gabrielle.laloy-borgna@inserm.fr), Thomas Lambin, Gabrielle Lescoat, jacqueline ngo, Magali Perier (INSERM U1032, Lyon, France), Julie Guillermet-Guibert (INSERM U1037, Toulouse, France), Cyril Lafon, and Stefan Catheline (INSERM U1032, Lyon, France)

In the context of development of a new treatment against pancreatic adenocarcinoma combining chemotherapy and cavitation, it is interesting to be able to mechanically characterize spheroid tumor models. Spheroids, which are clusters of cancerous cells (KPC) and fibroblasts (iMEF), are made using magnetic nanoparticles which are linked to the cells. Then, by using magnetic cell culture plates, cells aggregate, until forming a spheroid. Hence magnetic nanoparticles are incorporated in the spheroid, and can be used to generate elastic waves inside. A 5-ms magnetic pulse is induced in a coil, allowing to induce vibrations inside the spheroid placed in its center. It is observed using an optical microscope and an ultrafast camera. Then, tracking algorithms allow to measure the displacements at each point of the image, and for each time step. Next, noise correlation algorithms are used to retrieve the local velocity of shear waves inside the spheroid. Therefore, we obtained shear wave velocity maps of spheroids, using a non-destructive method. The influence of trypsine (known for disrupting the bounds between the cells) on shear wave velocity maps has been studied. As expected, a decrease in the wave velocity has been observed, attesting that this method is sensitive to elasticity. Furthermore, the impact of cavitation inside the spheroid has been investigated. Cavitation designates the bubble generation using ultrasound. It has been shown that cavitation also leads to a substantial decrease in the velocity of shear waves.

9:00

1aBAb3. Effects of crosslinking density on the acoustic properties of hydrogel scaffolds. Megan Anderson (Mech. and Aerosp. Eng., George Washington Univ., 800 22nd St. NW, Washington, DC 20052, andersonm@gwmail.gwu.edu), Jenna Osborn, Lijie Grace Zhang, and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC)

Hydrogels are popular materials for various biomedical applications due to their large water content which resembles that of biological tissues. Given the relevance of hydrogels in therapeutic applications, the non-invasive characterization of hydrogels is highly desirable for *in vivo* monitoring, whereas traditional testing methods are destructive, ultrasound is a safe and readily available tool that could be used for non-invasive characterization. In the present study, gelatin methacrylate (GelMA) hydrogel scaffolds were produced with a range of crosslinking densities by varying both GelMA concentration and ultraviolet light curing time. These scaffolds were evaluated using pulse-echo ultrasound techniques, resulting in measurements for the speed of sound, acoustic impedance, and attenuation coefficient of each hydrogel. Compression testing was also performed. The stiffness of the hydrogels was found to be between 5 and 235 kPa with an increase in the GelMA concentration or an increase in ultraviolet light curing time corresponding to a stiffer hydrogel. The impact of stiffness and preparation parameters on the acoustic properties of the hydrogels will be discussed. Further knowledge on the effect of crosslinking density on acoustic properties will inform the use of acoustic techniques in characterizing hydrogels.

9:15

1aBAb4. An *ex vivo* investigation of ultrasonic shear wave imaging for detecting liver cracks. Jingfei Liu (Mech. Eng., Texas Tech Univ., Box 41021, Lubbock, TX 79409, jingfei.liu@ttu.edu)

Liver crack is a type of liver trauma in which a capsular tear of different geometries occurs due to external impacts. Since liver crack is an important source of morbidity and mortality in emergency medicine, a timely and accurate detection of the crack location and geometry is highly demanded. In current emergency care, ultrasonography, although has a low accuracy, is mostly used for initial examination of liver trauma due to its immediate availability, high mobility, and nonionizing nature. To improve the diagnosis accuracy, ultrasonic shear wave imaging, a technique based on shear wave generation and tracking in liver tissue, was proposed. In this study, the feasibility and effectiveness of this method was investigated in an *ex vivo* scenario. A porcine liver with cracks of different geometries was tested. Shear waves were generated using acoustic radiation force impulse and recorded using ultrafast ultrasound imaging. To find the best way to display the cracks, different methods of signal processing based on time-of-flight, shear wave modulus, and accumulated shear wave path were applied to the shear wave displacement extracted. The results show that shear wave imaging is a more sensitive method than the conventional ultrasonography in detecting liver cracks.

9:30

1aBAb5. Histotripsy bubble dynamics in anisotropic hydrogels. Jacob C. Elliott (Graduate Program in Acoust., Penn State Univ., Res. West, State College, PA 16801, jce29@psu.edu), Julianna Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA), and Andrea Arguelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

Collagenous, anisotropic tissues such as tendon have demonstrated resistance to fractionation by histotripsy. B-mode imaging verifies the creation of bubbles, but oscillation and collapse of these bubbles does not cause fractionation. The objective of this work is to evaluate effects of anisotropy on bubble dynamics in tissue-mimicking hydrogels. Polyacrylamide, fibrin, and collagen hydrogels were fabricated. Axial sound speeds were measured in each direction to evaluate degree of anisotropy. Hydrogels were exposed to 1.5-MHz focused ultrasound with 10-ms pulses repeated at 1-Hz with $p+=89$ MPa, $p-=26$ MPa. Cavitation activity was monitored with high-speed photography and passive cavitation imaging with a Philips/ATL L7-4 transducer and Vantage® ultrasound system. Preliminary results show violent cavitation activity and rapid fractionation in polyacrylamide, collagen, and fibrin hydrogels which have low degrees of anisotropy (<1.2); such behavior is unlike what is observed in collagenous tissues. To make a gel more similar to collagenous tissues, fibrin gels were dehydrated and a >90% reduction of water content resulted in a 55% reduction in peak cavitation emission energy and a 260% increase in anisotropy. This suggests an influence from anisotropy, and therefore varying axial elastic moduli, on cavitation activity that will be explored further. [Work supported by NIH R21EB027886.]

9:45

1aBAb6. Atomization of a collagenase-treated tendon. Molly Smallcomb (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, molly.smallcomb@gmail.com) and Julianna Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Microfountains and atomization have been observed at tissue-air interfaces with focused ultrasound (fUS) and cause surface erosion. Highly collagenous tissues, such as tendon, have been resistant to erosion from atomization. Here, we evaluate whether tendon erosion is achievable after pre-treatment with collagenase, an enzyme that breaks down collagen. Fresh bovine tendons were cut lengthwise and submerged in 0.1% w/v collagenase solution for up to 24 h. Then, the tendon was partially submerged in water and the tendon-air interface insonified with 1.5 MHz fUS pulses of 10–20 ms at 1 Hz and predicted peak pressures of $p_+ = 106$ MPa ($p_- = 28$ MPa) for up to 5 min. The microfountain and atomization was monitored with high-speed photography. Results showed microfountains formed in collagenase-treated tendons with maximum heights reaching 4 mm; no fountains were observed in unaltered tendons. Additionally, surface erosion was observed upon gross examination in the collagenase-treated tendons. For 2-min treatments, 10 ms pulses produced ~24 mm diameter holes while 20 ms pulses produced ~34 mm holes but showed some thermal injury. No tissue erosion was observed in the unaltered tendons. These results suggest erosion may be achievable in collagen-degraded tendons such as tendinosis. [Work supported by NSF-GRFP-DGE1255832 and NIH-NIBIB-EB027886.]

10:00–10:15 Break

10:15

1aBAb7. Three-dimensional echo decorrelation imaging of percutaneous microwave ablation in liver tumors. Elmira Ghahramani Z., Peter D. Grimm, Bahar Saremi (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Kyuran A. Choe, Seetharam Chadalavada, Abouelmagd Makramalla, Ross L. Ristagno, Lulu Zhang (Radiology, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

In a pilot clinical study, patients underwent standard-of-care percutaneous microwave ablation (MWA) of liver tumors with monitoring by three-dimensional (3D) ultrasound echo decorrelation imaging. An MWA needle (Amica) was inserted into each tumor, guided by x-ray computed tomography (CT) and ultrasound imaging. A matrix ultrasound array (4Z1c array with Acuson SC2000 scanner, Siemens) was positioned on patients' skin near the MWA needle insertion point. Paired sequential complex in-phase/quadrature (IQ) echo volumes (inter-frame time 5–20 ms) were acquired throughout ablation at 10–20 s intervals, then processed to form 3D echo decorrelation images. Respiratory motion was estimated and compensated using a theoretical model that separates motion- and heat-induced echo decorrelation. Volumetric B-mode images constructed from the same IQ data were rigidly registered to follow-up contrast-enhanced magnetic resonance (MR) image volumes, recorded within 4 days after MWA. Specified registration transformations were then applied to cumulative 3D echo decorrelation maps, which were compared voxel-wise to ablation zones segmented from MR images. Receiver operator characteristic (ROC) curve analysis indicated effective prediction of local ablation by motion-compensated echo decorrelation imaging. Comparison with subsequent follow-up MR images tested the ability of 3D echo decorrelation imaging to predict any incomplete treatment or recurrence of treated tumors.

10:30

1aBAb8. Investigation of transcranial ultrasound delivery via Lamb wave mode conversion leveraging a transducer pair. Matteo Mazzotti (P. M. Rady Dept. of Mech. Eng., CU Boulder, 1111 Eng. Dr., CUB Mech. Eng., UCB 427, Boulder, CO 80309, matteo.mazzotti@colorado.edu), Eetu Kohtanen, Alper Erturk (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Massimo Ruzzene (P. M. Rady Dept. of Mech. Eng., CU Boulder, Boulder, CO)

Cranial Lamb waves have been recently proposed for the non-invasive identification of the mechanical properties of the cranial bone, the evaluation of its acoustic radiation characteristics, and the estimation of the elastic and transmission properties of cranial sutures. Few works have focused on the so-called Lamb mode conversion phenomenon for transcranial ultrasound delivery, in which an attempt was made to control the focal point within the brain for incident acoustic beams beyond the critical angle. In our study, we further investigate this problem by overcoming some of the limitations of previous works, which are mainly related to the use of simplified geometries and material properties. To this end, we experimentally and numerically investigate leaky Lamb waves in immersed cranial bones by adopting a two-step strategy. In the first step, the dispersion properties of leaky Lamb waves are retrieved in the 0–500 kHz range along the mechanical properties of the cranial bone layers. The material data are then used to determine, via numerical simulations, the range of frequencies in which Lamb mode conversion can be effectively exploited for transcranial focusing purposes. Finally, we experimentally validate the numerical findings by means of an immersed setup that utilizes a pair of ultrasonic transducers.

10:45

1aBAb9. Human skull profile and speed of sound estimation using pulse-echo ultrasound signals with deep learning. Churan He (Univ. of Illinois at Urbana-Champaign, Urbana, IL), Yun Jing (Acoust., Penn State Univ., State College, PA), and Aiguo Han (Univ. of Illinois at Urbana-Champaign, 306 North Wright St., Urbana, IL 61801, han51@illinois.edu)

Transcranial ultrasound brain imaging in human adults remains challenging, largely because human skulls cause severe phase aberration. Prior research has shown that phase aberration can be most accurately corrected if the skull profile (i.e., thickness distribution) and speed of sound are known *a priori*. We propose a deep-learning-based method to estimate the skull profile and sound speed from pulse-echo ultrasound signals. To demonstrate the feasibility of the method, *k*-wave simulations were performed to generate ultrasound radiofrequency (RF) signals backscattered from the skulls, using density and speed of sound maps derived from diagnostic CT scans of 5 *ex vivo* human skulls. An unfocused single-element transducer was simulated for ultrasound transmission and receiving at 0.75 MHz with 4-cycle tone bursts. A one-dimensional convolutional neural network (1D-CNN) model was designed and trained to predict the skull thickness and sound speed from RF signals. In an independent test set, the model yielded a mean absolute error of 0.3 mm for skull thickness prediction and 31 m/s for sound speed prediction. The result demonstrates that the deep learning method is capable of accurately estimating skull thickness and speed of sound, providing a potentially powerful tool for skull phase aberration correction in transcranial ultrasound brain imaging.

11:00

1aBAb10. Quantitative contrast-enhanced ultrasound for the evaluation and early detection of hepatocellular carcinoma. Connor D. Krolak (Bioengineering, Univ. of Washington, 1400 NE Campus Parkway, Seattle, WA 98195, krolakc@uw.edu), Alicia Clark (Bioengineering, Univ. of Washington, Seattle, WA), Manjiri Dighe (Radiology, Univ. of Washington Medical Ctr., Seattle, WA), and Michalakis A. Averkiou (Bioengineering, Univ. of Washington, Seattle, WA)

Hepatocellular carcinoma (HCC) is the most common primary liver malignancy with high mortality mainly due to pre-existing liver disease and delay in diagnosis, limiting treatment options. Currently, the state-of-the-art diagnostic metric for HCC is LiRADS (Liver Imaging Reporting and Data System), a qualitative image assessment based on the bolus transit characteristics of microbubbles that stratifies liver tumors suspected for HCC. However, with contrast-enhanced ultrasound, liver tumor perfusion can be quantifiably characterized to identify abnormal arterial flow after an initial portal venous and hepatic arterial flow reduction in the liver, which are typical features of HCCs. The image intensity in a region of interest (ROI) in the HCC is used to form a time-intensity curve (TIC) of the bolus transit (wash-in/washout). A lognormal curve is fitted to the TIC and parameters related to blood flow in the tumor are extracted. Respiratory gating and motion compensation are applied to the video loops before forming the TIC to remove noise and improve the accuracy of the HCC perfusion measurements. Parametric images of the rise time and peak intensity provide an HCC classification algorithm. Utilizing this approach, we provide an improvement to the LiRADS method for an earlier and more accurate diagnosis of HCC.

11:15

1aBAb11. Phase in bubble echoes due to nonlinear response improves contrast in ultrasound images. Ting-Yu Lai (Bioengineering, Univ. of Washington, 1705 NE Pacific St., Seattle, WA 98195, tlai9@uw.edu), Sara Keller (Bioengineering, Univ. of Washington, Seattle, WA), and Michalakis A. Averkiou (Mech. Eng., Univ. of Cyprus, Seattle, WA)

The image quality of contrast-enhanced ultrasound (CEUS) is limited by the tissue harmonics generated from the nonlinear propagation of sound waves, even at low amplitudes. Amplitude modulation (AM) is typically implemented by firing a full aperture and two complimentary half apertures. A phase difference ($\Delta\Phi_{AM}$) between the echoes from the full and half pulse has been observed which is lower for tissue than bubble echoes. This phase difference may be used as a segmentation criterion for tissues and bubbles to improve contrast to tissue ratio (CTR) of CEUS images. This technique may be further expanded by considering also the phase difference ($\Delta\Phi_{half}$) between the complementary half amplitude pulses of the AM scheme ($\Delta\Phi_{half}$). This additional phase difference results from the spatial differences of the complementary half amplitude fields. We considered the change in $\Delta\Phi_{half}$ when using various aperture patterns for AM. We observed that $\Delta\Phi_{half}$ increases as the complementary fields become spatially dissimilar. For certain aperture patterns, we found that the phase difference between the echoes of the complementary apertures was greater than that between the full and half pulse ($\Delta\Phi_{half}$i$>\Delta\Phi_{AM}$) and that it further contributed to the image CTR.

11:30

1aBAb12. Novel use of a lung ultrasound sensor for detection of lung interstitial syndrome. Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington1013 NE 40th St., CIMU Portage Bay Bldg., Seattle, WA98105, tdk7@uw.edu), Adam Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Gilles P. Thomas, Jeff Thiel, Alex T. Peek, Bryan W. Cunitz, Michael R. Bailey (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Kyle Steinbock, Layla Anderson, Ross Kessler, and Adeyinka Adedipe (Emergency Medicine Dept., Univ. of Washington, Seattle, WA)

The number and distribution of lung ultrasound (LUS) imaging artifacts—B-lines—is correlated with the presence of lung interstitial syndrome such as viral infection, acute respiratory distress syndrome, and pulmonary edema. The detection and quantification of B-lines requires specific training and is machine and operator dependent. In select clinical scenarios continuous hands-free monitoring of lung function is preferred, e.g., COVID-19 infection. The goals of this work were to: (1) identify RF signal features associated with B-lines and (2) develop a single-element, wearable, automated non-imaging lung ultrasound sensor (LUSS) for continuous monitoring. Verasonics Ultrasound Engine with a phased array probe (4.5 MHz) was used to perform standard 10-zone LUS in ten patients with confirmed pulmonary edema, and in five healthy volunteers. The RF data corresponding to each B-mode image and a series of 35 plane wave acquisitions were collected for off-line Doppler, decorrelation, and spectral analyses. B-line thickness and number were associated with peaks of Doppler power at the pleural line. Ten $10 \times 5\,\mathrm{mm}^2$ 5 MHz LUSS, powered by custom-built multiplexed transmit-receive circuit, were fabricated and tested with the same Doppler sequence and signal processing in a lung-mimicking sponge phantom containing variable volumes of water. [Work supported by NIH R01EB023910.]

11:45

1aBAb13. *In vivo* assessment of pulmonary edema in rats using the backscatter coefficient and envelope statistics. Theresa H. Lye (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., Fl. 9, New York, NY 10038, tlye@riversideresearch.org), Roshan Roshankhah (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Marie Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Jonathan Mamou (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

Ultrasound could be used as a convenient, non-ionizing method for monitoring pulmonary edema. In this study, the ability of backscatter coefficient (BSC) and envelope statistical quantitative ultrasound (QUS) parameters to evaluate edema in rat lungs was investigated. Ultrasound data were acquired from six healthy rats and six with pulmonary edema using a Verasonics Vantage ultrasound scanner. After ultrasound data acquisition, a piece of the lung was excised and immediately weighed (wet weight), before being dried in an oven and weighed again (dry weight). The wet and dry weight were used to compute the wet-to-dry (W/D) weight ratio, used as the gold standard for pulmonary edema severity. BSC and envelope statistical parameters were computed in overlapping time gates over the lung ultrasound data. The average and standard deviation of each parameter, along with the slope and intercept of a linear function fit to each parameter over acoustic propagation time, were obtained. Each QUS parameter was correlated with the W/D ratio. The strongest correlation was achieved by the standard deviation of spectral intercept and by the average homodyned K structure parameter, both with a 0.78 correlation coefficient magnitude. Therefore, BSC and envelope statistical parameters have the potential to improve monitoring of pulmonary edema.

Session 1aED

Education in Acoustics: Returning to Teaching Acoustics In-Person in the Post-COVID Era

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802

Andrew C. Morrison, Cochair
Natural Science, Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431

Chair's Introduction—9:00

Contributed Papers

9:05

1aED1. The successes and failures of flipping an introductory musical acoustics course in response to the COVID pandemic. Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

The advent of COVID-19 brought about many curricular changes throughout the academic world. One of the buzz words that quickly emerged was the idea of a flipped classroom. Just as quickly as we were all zooming, suddenly everyone was flipping their classrooms. A quick check into what this meant revealed itself to have its roots in the ideas of Physics Education Research. Sound & Music is an introductory course in musical acoustics that uses pedagogies based on these methods and has been taught continuously since 1996. The numerous hands-on activities that have been developed for this course were delivered in a specific sequence with two cumulative projects occurring at the end of the semester which focus on instrument design and room acoustics. When COVID occurred, these particular activities posed a big problem. Additionally, having to go remote hampered the format and delivery of the course. All activities and topic ordering had to be rewritten and reimagined. What was needed in the reimaging of the course was a flipping of the topics being taught without compromising the educational experience. Student involvement in the design and implementation of the changes over three semesters was a strong aspect of its overall success.

9:20

1aED2. Reflecting on universal design for learning in hybrid and in-person acoustics education. Katherine Simeon (Northeastern Univ., Boston, MA 02115, k.simeon@northeastern.edu) and Allison Hilger (Univ. of Colorado Boulder, Boulder, CO)

When transitioning to remote (Spring 2020) and hybrid (2020-21 academic year) learning during the COVID-19 pandemic, educators worked quickly to make lectures, coursework, and assessments accessible remotely. As educators return to in-person instruction, accessibility has become an even more essential and evolving part of effective course design (Kim, 2021). The Universal Design for Learning (UDL) Framework aims to maximize learning for all students through flexibility and choice (CAST, 2018). However, to broaden UDL's impact, these strategies should be transparent to students (Super *et al.*, 2020). This presentation will discuss the use of exam wrappers for students to reflect on UDL practices and their own learning in an undergraduate acoustics class. Exam wrappers are a reflective exercise where students analyze their study strategies after a summative assessment (Lovett, 2013). We will present how UDL practices were implemented in a Speech Science course. Then, we will review lessons learned from using a custom exam wrapper where students reflect on their study habits after exams that assess their application of anatomy and physiology to speech acoustic output. The exam wrapper also aims to obtain formative feedback on UDL practices. This work can inform strategies for UDL implementation and student reflection in acoustics education.

9:35

1aED3. Acoustics education accelerated by interactive simulators and research imaging system experiments. Thomas L. Szabo (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu) and Peter Kaczkowski (Verasonics, Inc., Kirkland, WA)

Learning acoustics traditionally involves abstract principles, equations and a limited number of specific exercises. We have developed an eight module introductory course in which students can engage with the ultrasound imaging curriculum experientially by interacting with real time GUI-based physics simulators. With only an introduction to the concepts, students with minimal mathematical background can quickly understand how the principles work by varying different simulator parameters over a wide range of examples and observing the output changes. Advanced students who know the simulator equations can learn the subtle interplay of different variables on outcomes on a deeper level. These simulators accelerate learning even more when the equations involve complicated integrals or equations which have to be evaluated numerically. Display options offer different ways of visualizing the acoustic and image data. Laboratories using a Verasonics ultrasound research imaging system provide hands-on learning of signal processing, transducer operation, and image processing.

9:50

1aED4. Strategies for teaching and extracurricular activities during COVID-19 transitions. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

As the COVID-19 pandemic moved through waves of severity, it became necessary to have contingency plans for teaching classes and alternative plans ready for related co-curricular and extra-curricular activities. This presentation will detail how our department handled the COVID-19 pandemic as we transitioned from fully online courses to a hybrid learning model. Also detailed is how the pandemic has affected our co-curricular and extra-curricular activities, many of which have connections to our department's undergraduate research program in acoustics. Strategies employed in the recent semester include online lecture components with on-campus laboratories, at-home portions of lab activities, creative uses of classroom spaces on campus, hybrid meetings, and hybrid colloquium presentions.

10:05–10:20 Break

10:20

1aED5. Penn State acoustic wave (PAW) kits: Development of "at-home" hands-on activities for first-year graduate level acoustics courses. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu) and Julianna Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Faculty in the Graduate Program in Acoustics at Penn State have often used in-class physical demonstrations to illustrate acoustic and vibration phenomena when teaching first year graduate students. During the Fall 2020 semester, when physical demonstrations were more difficult due to courses being moved online during COVID-19, we began including open-ended extra credit hands-on problems asking student to use household items to explore fundamental concepts. We observed that students who completed these extra credit experiments showed greater understanding of the material on homework assignments and exams than those who did not. Upon returning to in-person instruction for our first-year graduate courses, ACS 501 (Elements of Acoustics and Vibration) and ACS 502 (Elements of Waves in Fluids), we began developing a set of PAW (PSU Acoustic Wave) Kits that will allow all of our students—both resident and online—the opportunity to enhance their critical thinking in acoustics through inquiry-based learning. These PAW Kits will contain items not commonly found "at home" (masses, springs, dashpots, membranes, accelerometers, short pipes, lapel microphones, etc.), with the goal of at least one inquiry-based activity per major topic. This paper will describe some of the hands-on activities we are developing and testing.

10:35

1aED6. Return to in-person teaching and mentorship after COVID-19: hands-on photoacoustic demonstration apparatus. Leah E. Burge (Phys., U.S. Naval Acad., Annapolis, MD 21402, leahburge02@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

In March 2020, the U.S. Naval Academy implemented policies for teaching midshipmen off the "yard" (campus) to finish out their spring semester online. Strategies and regulations were put in place so that after spring break midshipmen would go home to families, relatives or friends and establish a learning environment space for on-line synchronous or asynchronous classes. An online summer school served as a link into the fall of 2020— which was a mixture of online and very small in-class size environments. (Every other week half of a class was online.) During the spring of 2021, classes were hybrid. However, Instructors alternated between adjacent classrooms teaching online—but in-person in one of the rooms. During COVID, mentorship student LEB worked off the yard to develop apparatus for online acoustic demonstrations and future research projects. LEB designed and built a homemade pulsed 620 nm LED photoacoustic apparatus for imaging experiments in water. Components include an LED current pulse driver, two aspherical focusing lenses, a cuvette filled with green dye or India ink, an ultrasonic 1 MHz immersion transducer and 4 low noise amplifiers with 87 dB total gain. Photoacoustic signals were averaged (10 000 trials) producing a strong 20 mV seven cycle pulse.

10:50

1aED7. Individual undergraduate research during remote learning. Kimberly A. Riegel (Phys., Queensborough Community College, 652 Timpson St., Pelham, NY 10803, kriegel@qcc.cuny.edu)

Due to the COVID-19 pandemic Queensborough Community College converted all learning to remote instruction. The college has remained fully remote from March 2020 until the present. During this time, the Queensborough ACousticS group (QuACS) continued to conduct individual undergraduate research. The author oversaw the NSF REU in a remote setting in the summer of 2020 and in a hybrid modality in the summer of 2021 including all professional development and logistical activities. For the first time, the program was able to accept students from community colleges as far away as California. This paper will highlight some suggestions about how to conduct remote research effectively. In addition, some observations about the difficulties of overseeing a remote undergraduate research program will be provided. Using the lessons learned from this remote experience, how to carry these lessons forward to enhance a traditional in person research experience will be discussed.

11:05–11:50
Panel Discussion

Session 1aPA

Physical Acoustics, Biomedical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Acoustofluidics I

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

Kedar Chitale, Cochair
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J. Mark Meacham, Cochair
Mechanical Engineering & Materials Science, Washington University in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm 203K, St. Louis, MO 63130

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

Chair's Introduction—7:55

Invited Papers

8:00

1aPA1. Fruitful femtofluidics using fast (acousto)fluidics. Naiqing Zhang (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA), Amihai Horesh, Ofer Manor (Chemical Eng., Technion - Israel Inst. of Technol., Haifa, Israel), and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, 9500 Gilman Dr. MC0411, MADLab SME344K, La Jolla, CA 92093, jfriend@ucsd.edu)

Acoustic streaming relies upon viscous attenuation of a propagating acoustic wave in a fluid to generate momentum flux and fluid flow. It is a nonlinear second-order phenomenon that generally produces rapid ($\sim$10 cm/s) fluid flow but only against weak ($\sim$10 kPa) pressure gradients. Here we describe a new form of acoustic streaming—acoustogeometric streaming—that is still nonlinear but relies on the interaction between the acoustic wave and the bounds of the fluid passage. If the boundaries oscillate with an amplitude that is significant in comparison to the width of the fluid passage, similarly rapid fluid flows may be generated even against very large ($>$1 MPa) pressure gradients. This is used to propel fluids at 6 mm/s in 20–150 μm wide, 100 nm tall nanoslit geometries and to transport, split, combine, and mix $\sim$100 fl droplets of fluid. In this presentation, we will also show what happens when the nanoslit height is reduced below 10 nm, and demonstrate specific examples with some simple analysis that explains the various phenomena, closed by some explanation of the broader potential benefit of this approach.

8:30

1aPA2. Theoretical and physical insights of acoustic radiation forces beyond Gor'kov potential. 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)

Acoustic radiation forces exerted on small objects by standing waves can be simply described by the Gor'kov potential that provides a basis for designs of acoustofluidic devices for trapping and separation of particles. However, when considering forces exerted on large objects by general sound beams, theoretical and physical insights into the forces must rely on momentum conservation. Nevertheless, the relative complexity of analysis of the force starting from stress tensor or spatial integral of acoustic radiation pressure creates a barrier on the insights in the frame of momentum conservation. Despite the early availability of momentum conservation frame for acoustic radiation pressure on interface between two media, the insights into acoustic radiation forces on large objects beyond Gor'kov potential are still limited. These insights are now enhanced by our recent theory of momentum conservation frame for three-dimensional acoustic radiations forces. The theory has simplified analytical results for acoustic radiation forces into compact and physically meaningful formulas. These formulas provide an understanding for forces and torques exerted by sound beams on both large objects and interface between two media in various scenarios, which will be summarized and reviewed.

9:00

1aPA3. Understanding and exploiting acoustic field-microswimmer interactions in acoustofluidic devices. J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm 203K, Saint Louis, MO 63130, meachamjm@wustl.edu), Minji Kim, Mingyang Cui (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, Saint Louis, MO), and Rune Barnkob (Heinz-Nixdorf-Chair of Biomedical Electronics, Tech. Univ. of Munich, Munich, Bavaria, Germany)

Acoustic microfluidic devices offer exquisite control for manipulation of objects sized from tens of nanometers to tens of microns. Our work focuses on applying various acoustofluidic solutions to understand microswimmer behavior and the mechanisms underlying the beating of propulsive cilia/flagella. Ultrasonic trapping in bulk acoustic wave (BAW) and substrate acoustic wave (SAW) fields enables observation of swimming without constraining rotational degrees of freedom and without damaging the cells. Here, we summarize recent studies using biciliate *C. reinhardtii* algae cells, including three-dimensional body motion, synchronization/asynchronization of *cis* and *trans* cilia waveforms, and thermal and fluid-property effects. Acoustofluidic approaches provide unprecedented flexibility as a tool to investigate cell motility. Conversely, acoustic field-microswimmer interactions can also be exploited by applying the *C. reinhardtii* cells as dynamic probes to assess acoustofluidic device performance. We have previously reported use of these cells to identify optimal resonant frequencies of operation and to qualitatively map the strength of the acoustic field in BAW devices. Here, we extend the method to quantify the acoustic energy density of a straight microchannel operating at the first half-wavelength resonance, achieving agreement within 1 % to a standard method based on passive particle tracking. New applications for SAW devices are also presented.

9:30

1aPA4. Theory for attenuation and dispersion of acoustic propagation through a muddy medium. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

Mud as a medium typically consists of clay particles, small silt particles (primarily quartz), and water. The clay particles lock together in what is called a card-house configuration, and the silt particles are held in suspension by the card house. The low frequency velocity of sound waves appears to be well-predicted by the Mallock-Wood theory that gives weighted averages of bulk compressibility and material density. At low frequencies, the attenuation is caused by the viscous dag of the water on the suspended silt particles, and the attenuation varies as the frequency squared. The deviation of the phase velocity from the low frequency limit at low frequencies is less well-understood but appears to vary as the frequency to the three-halfs. At higher frequencies, the dominant mechanism appears to be a relaxation mechanism associated with the card house. Each clay particle carries a net electrical charge and the resulting repulsion forces the clay particles to be spatially separated. However, the particles tend to stick together edge to face, and the van der Waals attraction between the clay particles tends to dominate over the electrostatic repulsion. However, the passage of a sound wave temporarily breaks some van der Waals bonds and they subsequently recombine as the acoustic pressure ebbs. The present paper seeks to quantify the effects of the relaxation processes. It is demonstrated that the latter leads to an attenuation that, at higher frequencies, varies nearly linearly with frequency. [Work supported by ONR.]

10:00–10:15 Break

Contributed Papers

10:15

1aPA5. Modulation of acoustofluidic parameters to assess effect on molecular loading in human T cells. Connor Centner (Bioengineering, Univ. of Louisville, 580 S Preston St., Louisville, KY 40214, connor.centner@louisville.edu), John Moore, Mary Baxter (Bioengineering, Univ. of Louisville, Louisville, KY), Mariana Vinseiro Figueira (Chemical Eng., Ohio State Univ., Columbus, OH), Zachary Long (Bioengineering, Univ. of Louisville, Louisville, KY), Clinton Belott, Michael Menze (Biology, Univ. of Louisville, Louisville, KY), Paula Bates (Medicine, Univ. of Louisville, Louisville, KY), Kavitha Yaddanapudi (Surgery, Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (Bioengineering, Univ. of Louisville, Louisville, KY)

T-cell therapies are rapidly emerging for treatment of cancer and other diseases but are limited by inefficient non-viral delivery methods. Acoustofluidic devices are in development to enhance non-viral delivery to cells. The effect of acoustofluidic parameters, such as channel geometry, on molecular loading in human T cells was assessed using 3D-printed acoustofluidic devices. Devices with rectilinear channels (1- and 2-mm diameters) were compared directly with concentric spiral channel geometries. Intracellular delivery of a fluorescent dye (calcein, 100 μg/ml) was evaluated in Jurkat T cells using flow cytometry after ultrasound treatment with cationic microbubbles (2.5% v/v). B-mode ultrasound pulses (2.5 MHz, 3.8 MPa output pressure) were generated by a P4-1 transducer on a Verasonics Vantage ultrasound system. Cell viability was assessed using propidum iodine staining (10 μg/ml). Intracellular molecular delivery was significantly enhanced with acoustofluidic treatment in each channel geometry, but treatment with the 1-mm concentric spiral geometry further enhanced delivery after acoustofluidic treatment compared to both 1- and 2-mm rectilinear channels (ANOVA $p < 0.001$, $n = 6$/group). These results indicate that 3D-printed acoustofluidic devices enhance molecular delivery to T cells, and channel geometry modulates intracellular loading efficiency. This approach may offer advantages to improve manufacturing of T cell therapies.

10:30

1aPA6. Acoustic radiation force and torque on objects with continuous and discrete inhomogeneity. Thomas S. Jerome (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, tsjerome@utexas.edu) and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

At the Fall ASA meeting in 2019, the Born approximation was extended to acoustic radiation force and torque on objects of arbitrary shape with spatially varying density and compressibility [*Proc. Meet. Acoust.* **39**, 045007 (2020)]. This model is applied here to a standing plane wave incident on objects either with continuously varying material properties or composed of connected homogeneous regions with different material properties. The approximation for inhomogeneous objects is subject to the same restrictions on incident field structure, scatterer size, and material contrast as for homogeneous objects. Results are presented for cylinders with a variety of strongly asymmetric inhomogeneity distributions. Closed-form expressions are obtained for a sphere with material properties that vary linearly with distance from its center and for a sphere surrounded by concentric spherical layers. Results for the latter case are shown to be in close agreement with existing theory for the radiation force on a nucleated cell modeled as a multilayered compressible sphere [Wang *et al.*, *J. Appl. Phys.* **122**, 094902 (2017)]. The Born approximation thus proves convenient for determining

radiation force and torque on soft objects with both arbitrary shape and inhomogeneity. [TSJ was supported by the ARL:UT McKinney Fellowship in Acoustics.]

10:45

1aPA7. Transferred energy and acoustic spin density in fluid spherical core-shell scattering scenarios. Jose R. Alcaras (Phys., Univ. of Sao Paulo, Av. Bandeirantes, 3900, Ribeirao Preto, Sao Paulo 14040900, Brazil, jose.alcaras@usp.br) and Alexandre S. Martinez (Phys., Univ. of Sao Paulo, Ribeirao Preto, Sao Paulo, Brazil)

Sound wave scattering, along with radiation force and torque, are constantly in vogue topics for several distinct applications, such as small object manipulations and Acoustofluidics. Specially for scattering scenarios, far-field approximations are often considered for dealing with the sound waves far beyond the scattering objects. However, a more in-depth investigation of near and internal fields during a scattering process may be able to unravel dynamical and mechanical properties of the scattering objects. In this study, we propose a time-averaged study of the transferred energy and acoustic spin density in a fluid spherical core-shell. We derive the expressions for the internal waves in terms of spherical harmonics and obtaining Mie-like coefficients. Using those expressions, we analytically compute the time-averaged acoustic energy transferred by a plane wave. Furthermore, we use very recent results with regards to acoustic spin to bring a discussion about spin density transfer, properly computing the quantity for this scenario. This is a follow-up study for understanding acoustical properties of compressible fluids with different geometries, searching for electromagnetic parallels, such as Fano resonance regimes, that may be applicable for building or characterizing acoustic metamaterials.

11:00

1aPA8. Direct observation of capillary waves due to high frequency ultrasound and their relation to atomized droplets. William Connacher (Mech. and Aerosp. Eng., Univ. of California, San Diego, 4787 Cather Ave., San Diego, CA 92122, wconnach@eng.ucsd.edu), Shuai Zhang (Mech. and Aerosp. Eng., Univ. of California, San Diego, San Diego, CA), Jeremy Orosco (Mech. and Aerosp. Eng., Univ. of California San Diego, Del Mar, CA), and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

Ultrasonic atomization has been studied and used in applications for decades, but a mechanistic understanding of droplet production does not exist at a quantitative level. Early work seemed to indicate that droplet size follows from the driving frequency according to Kelvin's equation. The underlying assumptions in this model do not hold at larger frequencies, and the model relies on a fitting parameter that has not been consistent across experimental studies. Additionally, modern experiments often indicate multi-modal droplet size distributions further complicating the physics. It is difficult to experimentally measure relevant quantities like capillary wavelength because of the time and length scales involved and the fact that the phenomenon occurs at a free liquid surface. We use a high speed digital holographic microscope, the only one of its kind in the world, to image the liquid surface—in full 3D and real time—fast enough to resolve capillary waves generated by 5–50 MHz ultrasound. We extract from this data the spectrum of capillary wavelengths and then compare these to droplet size distributions measured with laser diffraction particle sizing. Both thickness mode and surface acoustic wave lithium niobate transducers are included as well as a range of frequencies in each case. It is shown that droplet size does depend directly on capillary wavelength, but that capillary wavelength is not related to the driving frequency in a simple way.

11:15

1aPA9. Effect of basset history drag on the accurate determination of streaming flow with PIV measurement. Shuai Zhang (UCSD, San Diego, CA, shz134@ucsd.edu), Jeremy Orosco, and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

In acoustofluidics, the conflation of acoustic streaming and acoustic pressure effects on suspended particles, especially for particle image velocimetry (PIV) with micron-sized particles, leads to inaccurate streaming measurements. Typical multiphase streaming models fail to account for the Basset history drag, which emerges due to a mismatch between the momenta of the particle and the local flow. The history drag is proportional to a fractional-order derivative of the mismatch. Since this force integrates over time, its absence leads to significant modeling errors while also augmenting errors from other sources (e.g., particle-fluid density mismatch and acoustic radiation forcing). We investigate the effects of particle size selection on accurate streaming flow determination from PIV microacoustofluidic system measurements. We achieve this through experimental observation of particle behaviors under varied ultrasonic forcing frequencies (30–500 MHz) and particle sizes. We pair the empirical study with an analytical fractional-order differential model of a spherical particle under the influence of radiation forces. Rigorous comparative analyses between the model and observations are undertaken to more completely characterize the magnitude of various effects leading to spurious PIV data. We expect the results of this investigation to clearly delineate optimal PIV particle size selection for the continued study of microacoustofluidic systems.

11:30

1aPA10. From macroscale to microscale characterization of droplet breakup regimes using a micromachined ultrasonic droplet generator. Li Shan (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, St. Louis, MO), Kyle Krippner (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm 203K, Saint Louis, MO 63130, kkrippner@wustl.edu), and J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, Saint Louis, MO)

Controllable, uniform droplet generation is critical to many industrial processes and biomedical applications (e.g., materials synthesis, cell handling, and aerosol drug delivery). We have developed an ultrasonic droplet generator that can enable dynamic control over the droplet size distribution in the 2–75 μm range. The system uses a bulk piezoelectric transducer to generate a standing acoustic pressure field that drives fluid transport through an array of microscopic orifices. Our previous work suggests that certain resonances lead to variability in the pressure gradient at the orifices, affecting the droplet size distribution of the resultant spray. Here, we present a parametric study relating ejection uniformity to orifice size, operating frequency, and drive amplitude. We introduce a new system configuration that allows high-resolution stroboscopic imaging of the liquid-gas interface evolution at each orifice. Although the entire micronozzle array remains acoustically active, we use orifice size to control the flow resistance, restricting ejection to a single preselected orifice. Observed ejection regimes resemble classical descriptions of jet breakup (Rayleigh, wind-induced, and atomization), depending on drive voltage. A regime map is created using 5, 10, and 15 μm orifices for 0.5–2.0 MHz actuation from threshold ejection amplitude to the point of chaotic wind-induced breakup.

11:45

1aPA11. Acoustic erythrocytometer for mechanical cell probing. Andreas Link (Biomedical Eng., Univ. of Glasgow, Oakfield Ave. 79-85, Rankine Bldg., 706c, Glasgow City G12 8LT, United Kingdom, a.link.1@research.gla.ac.uk), Raymond Sparrow, and Thomas Franke (Biomedical Eng., Univ. of Glasgow, Glasgow City, United Kingdom)

The mechanical properties of cells provide key insights into the type, differentiation and/or pathology of a cell. The mechanical analysis of a cell population, such as a blood sample, enables meaningful biological and medical interpretation for the diagnosis and monitoring of diseases. There are number of diseases which alters the mechanical properties of human red blood cells (RBCs). Here, an acoustic method to probe both the viscous and elastic mechanics of single RBCs by SAW in a microfluidic device is presented. The device operates by exciting a surface acoustic wave in a microfluidic channel creating a stationary acoustic wave field of nodes and antinodes. RBCs are attracted to the nodes and are deformed. Using a stepwise increasing and periodically oscillating acoustic field the static and dynamic deformation of individual red blood cells one by one was studied and the deformation by the Taylor deformation index D and relaxation times were quantified. The precision of the measurement allows to distinguish between individual cells in the suspension and provides a quantitative viscoelastic fingerprint of the blood sample with a resolution of a single cell. The method overcomes limitations of other techniques that provide averaged values and has the potential for high-throughput screening.

Session 1aSA

Structural Acoustics and Vibration, Engineering Acoustics, Signal Processing in Acoustics, Computational Acoustics, and Physical Acoustics: Novel Techniques for Nondestructive Evaluation in Structural Acoustics and Vibration I

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

Tyler J. Flynn, Cochair
JHU/APL, 11100 Johns Hopkins Rd., Laurel, MD 20723-6099

Invited Papers

8:00

1aSA1. Remote acoustic sensing of damage in vibrating plates. David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI) and Tyler J. Flynn (Appl. Phys. Lab, Johns Hopkins Univ., 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu)

The presence of damage in a structure often changes its vibrational characteristics. For example, resonant frequencies and mode shapes may be changed in a damaged structure relative to a healthy counterpart. For structures immersed in a fluid medium, these changes alter the radiated sound such that passive recordings can be examined to glean information about the damage. This approach toward structural health monitoring enjoys the benefit of being non-contacting, allowing sensors to be independent from the system of interest, but comes at the cost of introducing challenges associated with the acoustic environment and exogenous noise. In this presentation we describe a set of remote acoustic techniques for detection, localization, and classification of damage in base-excited clamped plates in a reverberant environment. This is accomplished by measuring audible broadband sound (0.2–6 kHz) from $12 \times 12 \times 1/16\,\mathrm{in}^3$ plates with and without damage in an ordinary laboratory using a remote microphone array. For detection and localization, damage-free plates serve as baselines compared against potentially damaged plates. For classification, a data-driven approach is employed using finite element simulations as training data. Results are presented for various damage types—including cuts, boundary failures, and delamination—in aluminum and fiber-composite plates. [Sponsored by NAVSEA and the NDSEG Fellowship.]

8:20

1aSA2. Detection and imaging of stress-corrosion cracking in stainless steel using the linear elastodynamics gradient imaging technique. Marcel Remillieux (Geophys., Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, mcr1@lanl.gov), Luke Beardslee (Geophys., Los Alamos National Lab., Los Alamos, NM), and Timothy J. Ulrich (Detonation Sci. & Technol., Los Alamos National Lab., Los Alamos, NM)

Linear Elastodynamics Gradient Imaging Technique (LEGIT) is a nondestructive testing technique recently developed to image cracks on the surface of samples. This technique is based on the interaction of an acoustic wave with a crack and the resulting discontinuity in the acoustic wave field such an interaction generates. LEGIT is used to image stress-corrosion cracks in two samples of 304 stainless steel. The first sample is relatively small ($50 \times 25 \times 1.4\ \mathrm{mm}^3$) with two mm-size cracks. For this sample, the acoustic source is glued to the sample. The second sample is much larger ($180 \times 76 \times 25.4\ \mathrm{mm}^3$), has a curvature, a weld, and multiple cracks running parallel to and across the weld. The imaging of the cracks for this more complex sample is achieved without contact by partial immersion using an underwater acoustic transducer. Imaging data obtained from LEGIT in the two experiments show excellent agreement with a vibro-thermography.

8:40

1aSA3. Nonlinear ultrasound to characterize dislocation-based damage in classical and rolling contact fatigue. Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801, kmatlack@illinois.edu)

Material damage in structural components is driven by nano- and micro-scale defects that evolve early in the component's life. In metals, the interaction of an ultrasonic wave with these nano- and micro-structural defects, such as dislocations, precipitates, and micro-cracks, generates a second harmonic wave that is proportional to the acoustic nonlinearity parameter, which is an absolute and measurable material parameter. These defects are known to cause measurable changes in the nonlinearity parameter, which changes as the nano- and micro-structural defects evolve in the material. This talk will discuss how nonlinear ultrasound (NLU) techniques that measure the nonlinearity parameter can be used as a nondestructive evaluation tool to characterize dislocation-based damage in metals, and thus characterize early-stage damage. Our group's recent work on using NLU to characterize different types of cyclic loading, classical

fatigue and rolling contact fatigue, will be discussed. Specifically, ultrasonic measurements are related directly to materials characterizations on similar length scales to better understand the relationship between the nonlinearity parameter and the dislocation-based damage evolution.

9:00

1aSA4. Enhanced resonant ultrasound spectroscopy for multi-material characterization. Paul Geimer (Los Alamos National Lab., P.O. Box 1663, M.S. D446, Los Alamos, NM 87545, pgeimer@lanl.gov), Luke Beardslee (Los Alamos National Lab, Los Alamos, NM), Marcel Remillieux (Geophys., Los Alamos National Lab., Los Alamos, NM), and Timothy J. Ulrich (Los Alamos National Lab., Los Alamos, NM)

Resonant ultrasound spectroscopy (RUS) is a non-destructive technique to describe the elastic tensor of a material through inversion of a sample's measured resonance frequencies. The original RUS approach is limited to simple geometries as it uses a semi-analytical method to simulate the vibrational response. We modified this approach by combining the finite-element method with genetic algorithm. The new approach is applicable to systems with composite materials, complex geometries, and misaligned material and geometry axes. We present recent work focused on the response of a graphite-TZM composite system. We find composite material properties only vary a few percent from results from single-material RUS. Results from varied geometries and materials suggest further optimizations to enhanced RUS.

9:20

1aSA5. Nonlinear resonant ultrasound spectroscopy (NRUS) for nondestructive evaluation. Parisa Shokouhi (Eng. Sci. and Mech., The Penn State Univ., 212 EES Bldg., University Park, PA 16802, pxs990@psu.edu) and Jacques Riviere (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Nonlinear Resonant Ultrasound Spectroscopy (NRUS) is a simple yet effective technique for nondestructive evaluation of incipient damage in a wide range of materials. NRUS combines the advantages of resonance-based testing and nonlinear ultrasonic or acoustic testing in that it can be used for testing complex-shaped objects and detection of microscopic defects. In this talk, I describe several applications of NRUS in: (1) monitoring of microcracking in concrete materials due to various damage processes; (2) monitoring the growth of microscopic fatigue cracks in aluminum; and (3) predicting the fatigue life of additively manufactured metallic alloys. In addition, I provide a snapshot of our ongoing research in developing non-contact NRUS test set up and using NRUS for in-process monitoring of additively manufactured parts. I close with describing the limitations of NRUS and possible directions for future research to realize NRUS potential in nondestructive evaluation (NDE) applications.

9:40

1aSA6. Ultrasonic time reversal on defects in carbon fibre reinforced polymer using chirp-coded ultrasound in complex media. Martin Lints, Andrus Salupere (Dept. of Cybernetics, School of Sci., Tallinn Univ. of Technol., Tallinn, Estonia), and Serge Dos Santos (Inserm U1253 iBrain, INSA Ctr. Val de Loire, 3 Rue de la Chocolaterie, Blois 41034, France, serge.dossantos@insa-cvl.fr)

Various nonlinear methods have been investigated in recent years offering improvements of over conventional B-mode in medical imaging and non-destructive testing (NDT) [1]. The most common include harmonic and overtone generation, inter-modulation product generation and resonant frequency shift, also known as Nonlinear Elastic Wave Spectroscopy (NEWS) methods. Furthermore, new optimized excitations such as Nonlinear Time Reversal (TR) based methods are needed and, thanks to the analysis of symmetry properties of cross-correlation generated functions, a wide range of innovation are classified within TR-NEWS methods [2]. This work proposes to unite the frequency-modulated constant-wave scatterer localization with TR-NEWS principles to enable the localization of scatterers distant from the transducers and the analysis of their nonclassical nonlinearity which often can be caused by microdamage or cracking. In this paper, using the found frequency of the defect, the delayed TR–NEWS can be leveraged to periodically excite the damaged region with the focused ultrasound. Resonance excitation results show potential for enhancing the wave motion more than a simple focused waves [3]. Thanks to the complexity of the acoustical propagation in a complex medium, the physical interpretation of the cross-correlation function used during the TR-NEWS analysis can be easily understood in terms of the acoustic energy focused in space and in time.

10:00–10:15 Break

10:15

1aSA7. Nondestructive evaluation of solid cargo inside cylindrical containers with linear and nonlinear resonance spectroscopy. Kevin Y. Lin (Hyperion Technol. Group, Inc., 545 Commerce St., Tupelo, MS 38804, klin@hyperiontg.com), Joel Mobley (Phys. and Astronomy, Univ. of MS, University, MS), Wayne Prather (National Ctr. for Physical Acoust., Oxford, MS), and Joseph Gladden (NCPA / Office of Res., Univ. of MS, University, MS)

Nondestructive evaluation (NDE) of cylindrical containers is of critical importance because of their ubiquity in industrial applications. However, there is a paucity of research on the NDE of solid cargo inside sealed enclosures with external sensors. This talk covers the application of linear and nonlinear resonance spectroscopy to the NDE of cargo inside cylindrical containers. The first study focused on a Transnuclear-32 (TN-32) cask used for the dry storage of spent nuclear fuel assemblies. The modes of the TN-32 were measured onsite and Finite Element Analysis was used to investigate modal changes due to loadings of fuel assemblies. Because of the difficulties introducing damage to fuel assemblies inside a TN-32, a 1:6 scaled cask containing 32 mock-up fuel assemblies was manufactured for laboratory studies. Using the linear approach, metrics were developed that could detect cargo damage down to the single assembly level. To resolve the signatures of damaged versus missing assemblies, nonlinear ultrasound resonance spectroscopy was utilized to investigate the contact nonlinearity between a smaller scale container and loose spheres which mimicked internal debris. Based on the resulting data, a phenomenological contact-loss model was developed which could estimate the total mass of the spheres regardless of the material and radius.

10:35

1aSA8. Unraveling the nonlinear elastic response of materials with dynamic acousto-elastic testing. Jacques Riviere (Penn State Univ., University Park, PA, jvr5626@psu.edu), Linying Gao, and Parisa Shokouhi (Penn State Univ., University Park, PA)

Compared to standard nonlinear ultrasonic techniques based on harmonic generation or resonance tests that provide an average nonlinearity level, Dynamic Acousto-Elastic Testing (DAET) is a pump-probe approach that allows one to track the nonlinear elastic response at each phase of the dynamic cycle, and to quantify additional features such as transient softening/weakening and hysteresis. These so-called nonlinear signatures can help us unravel the origins of the nonlinear response at the micro/mesoscopic scale. In this talk, we first review some of the DAET results obtained by our group and others in a wide range of materials from cracked metals to rocks and unconsolidated granular media. Next, we present some of our most recent results where we investigate the effect of relative humidity (RH) on the nonlinear elastic response of granular media. We use DAET on samples of glass beads under dry ($\sim$10%), ambient ($\sim$60%) and humid ($\sim$100%) conditions. We find that all extracted nonlinear parameters in humid samples increases by an order of magnitude compared to those in dry samples. This is consistent with the idea that water adsorption on the grains makes the contact junctions weaker. This has great relevance when trying to quantify the nonlinearity of porous materials.

10:55

1aSA9. Singular value decomposition of acousto-elastic matrix for modal analysis of mechanical structures. Chloe Palerm (Langevin Inst., ESPCI, 1 rue Jussieu, Paris 75005, France, chloe.palerm@espci.fr), Claire Prada (Langevin Inst., ESPCI, Paris, France), Benoit Gerardin (SafranTech, Paris, France), Arnaud Talon (Safran Helicopter Engine, Bordes, France), and Julien De Rosny (Langevin Inst., ESPCI, Paris, France)

All mechanical structures exhibit natural vibration eigenmodes, characterized by their natural frequencies, modal shapes and damping factors. For some industrial applications, the investigation of those parameters is important either to avoid resonance phenomena or to detect the apparition of flaws. However, the analysis of symmetric or periodic structures can be complex because several modes can overlap at a same frequency. Here, we propose to measure the acousto-elastic transmission matrix between an array of loudspeakers and the displacement on the structure under test recorded with a laser vibrometer. The singular value decomposition of the transmission matrix allows detecting, counting and separating overlapping modes. The method is validated on a thin aluminum square plate, a gear and a turbine wheel. The results are compared to conventional impact hammer excitation. Based on this analysis, a selective excitation of a given mode is made possible. Finally, the effect of flaws on SVD is studied on the structures.

Contributed Papers

11:15

1aSA10. Airborne acoustic characterisation of cultural heritage paintings: A preliminary study. Victor Takahashi (Greman UMR 7347, INSA Ctr. Val de Loire, Tours Univ., CNRS, 3 rue de la Chocolaterie, Blois 41000, France, victor.takahashi@etu.univ-tours.fr), Jerome Fortineau, Michaël Lemâtre, and Marc LETHIECQ (Greman UMR 7347, INSA Ctr. Val de Loire, Tours Univ., CNRS, Blois, France)

In order to assess the health of cultural heritage paintings, for example prior to their restoration, visual inspections as well as techniques such as X-ray imaging are commonly used. Non-contact acoustic methods could become another exploration modality thanks to their non-destructive character and the fact that they are expected to be more sensitive to certain defects, such as a delamination, than other techniques. A setup has been designed for such applications, based on airborne ultrasonic transducers operating in the several hundred kHz to over 1 MHz range, specific transmit/receive electronics and automated scanning stages allowing 2D through-transmission scans to be performed on areas up to 80×80 square centimeters. Measurements on well-known thin plates made of plastic, wood and gesso on wood were first performed to evaluate the possibilities of the technique, then simple paintings on canvas were tested and finally a real ancient painting was scanned. A method where the transmission amplitude is plotted versus the angle between the propagation axis and the normal to the painting surface is developed for local characterization of the elastic constants of the sample under test, thanks to comparison with simulated Lamb wave dispersion curves using a fitting procedure. These promising results open the way towards comparison between acoustic, radiological and other imaging results on paintings with severe defects that require restoration.

11:30

1aSA11. Vibrometric ranging for air-coupled ultrasonic testing. Benjamin Bühling (Dept. of Non-Destructive Testing, Bundesanstalt für Materialforschung und -prüfung (BAM), Unter den Eichen 87, Berlin 12205, Germany, benjamin.buehling@bam.de), Stefan Maack, and Christoph Strangfeld (Dept. of Non-Destructive Testing, Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany)

Ultrasonic testing is a well-established method to assess characteristics of a specimen non-destructively via the time of flight (ToF) of the acoustic signal. While traditional methods require dry or liquid coupling of transducer and specimen, air-coupled ultrasound (ACU) transducers allow contact-free measurements. The signal is passing through air before it interacts with the specimen and, finally, is received by a sensor. Knowledge of the acoustic travel time between transducer and specimen is imperative to calculate the correct ToF through the specimen. In this contribution, we propose the use of a laser Doppler vibrometer to obtain this delay time through air. Exploiting the opto-acoustic effect, an ACU signal is observable before and after reflection at the specimen surface. By auto-correlating the signal, the delay time through air can then be found non-invasively, calibration-free, resonance-free, and on-line. Furthermore, this method provides a reference signal for pulse compression approaches. The benefits of vibrometric ranging are demonstrated experimentally using a piezoelectric and a fluidic transducer.

11:45

1aSA12. Fully acoustic NDT for composite plate via LDR technique. Nudurupati S. Hanuman (Mech. Eng., National Inst. of Technol. Meghalaya, Bijni Complex, Laitumkhrah, Shillong 793003, India, snudurupatinsvnhanuman@nitm.ac.in) and Tanmoy Bose (Mech. Eng., National Inst. of Technol. Meghalaya, Shillong, Meghalaya, India)

The contact-based NDT methods occupy the industrial application to evaluate the material integrity. But, some difficulties are faced during inspection of the structures due to vast impedance mismatch at boundary and highly elevated temperatures. It is a big challenge to the modern NDT field. The noncontact NDT methods have overcome these challenges. Meanwhile, nonlinear techniques have been growing in advance to evaluate micro-scale damages. This study presents that air-coupled acoustic testing based on the local defect resonance (LDR) technique. In order, the numerical and experimental study has been carried out on a composite plate. The explicit dynamic analysis is carried out for numerical study using the ABAQUS software package. The dual piezo-speakers and microphones are used in an experimental case. The chirp signals are provided to the plate to stipulate the damage in both numerical and experimental cases. Corresponding sound pressure levels (SPL) are obtained at the damage region in numerical and experimental cases. The output data are processed by using MATLAB in an experimental case to get sound pressure levels (SPL).

MONDAY MORNING, 29 NOVEMBER 2021 COLUMBIA A, 9:00 A.M. TO 12:00 NOON

Session 1aSC

Speech Communication: Second-Language Speakers and Listeners (Poster Session)

Jenna T. Conklin, Chair

Carleton College, One North College Street, Goodsell Observatory, Northfield, MN 55057

All posters will be on display from 9:00 a.m. to 12:00 noon. Authors of odd numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and authors of even numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

1aSC1. Change in pitch and voice quality of L2 speakers of English as a function of similarity between their L1 and English. Hahn Koo (San Jose State Univ., One Washington Square, San Jose, CA 95192-0093, hahn.koo@sjsu.edu)

Bilingual speakers sometimes change their pitch and voice quality when they switch from one language to the other. For example, when speaking in L2 rather than L1, German learners of French pronounce vowels with less adduction of the vocal folds (Pützer *et al.*, 2016), and Korean learners of English with a lower pitch (Cheng, 2020). Here, I present a corpus study which suggests that the extent to which L2 learners of English change their pitch and voice quality may depend on how similar their L1 is to English. I extracted 68 211 vowels—51 857 in L1 and 16 354 in L2—from 1617 speakers with 21 different L1 backgrounds (including English) in the CSLU: 22 Languages Corpus and measured F0, harmonics-to-noise ratio (HNR), and H1i–H2 for each vowel. I then computed two cluster distances for each L1 and for each measure: (1) vowels from the native English speakers versus L1 vowels from the learners and (2) L1 vowels versus L2 vowels from the learners. I found strong correlations between (1) and (2): $r = 0.416$ for F0 ($p = 0.068$), $r = 0.531$ for HNR ($p = 0.016$), and $r = 0.374$ for H1–H2 ($p = 0.105$).

1aSC2. Acoustic influence of L1 Spanish on L2 English vowel production. Jenna T. Conklin (Carleton College, One North College St., Goodsell Observatory, Northfield, MN 55057, jconklin@carleton.edu), Olga Dmitrieva, Ye-Jee Jung (Purdue Univ., West Lafayette, IN), and Weiyi Zhai (McGill Univ., Northfield, MN)

The strength of influence of the L1 vowel inventory on L2 vowel acquisition is well-documented: learners who identify an L2 phoneme as belonging to an L1 category may fail to establish a distinct category for the new phoneme (Flege, 2002), while those who establish a new category for an L2 vowel may ultimately exhibit more native-like pronunciation (Bohn & Flege, 1992). In learning an L2 with many more vowel categories than the L1, as Spanish-English bilinguals must, speaker-specific category assimilation and discrimination are important contributors to the level of accentedness and intelligibility. This study examines the influence of L1 vowels on L2 production via formants and duration of stressed and unstressed L1 Spanish/L2 English vowels. Ten advanced Spanish-English bilinguals completed a shadowing task in both languages, producing five vowels from each language in various target words. Examined in the aggregate, most vowels remained spectrally distinct along both the F1 and F2 dimensions, with a few categories collapsed across one dimension or the other; unstressed vowels raised in both languages, but raised further in English than Spanish. However, interspeaker differences in the patterns of category assimilation necessitate an individual analysis, highlighting the high level of variation in acquisition across individuals.

1aSC3. Production and perception of English phonemes for Chinese college students. Chang Liu (Univ. of Texas at Austin, Austin, TX, changliu@utexas.edu), Yunzi Wan (Chengdu Inst. Sichuan Int. Studies Univ., Chegdu, Sichuan, China), Junbo Liu (Woodbridge High School, Irvine, CA), Sha Tao, and Wenjing Wang (Beijing Normal Univ., Beijing, China)

The production and perception of English vowels and consonants were measured for Chinese college students who started their English education from 9 to 13 years old. In the production experiment, 12 English vowels were recorded in the /hVd/ context and twenty one English consonants were recorded in /aCa/ context for each participant. In the perception experiment, participants were asked to identify English vowels in /hVd/ and English consonants in /aCa/ that were produced by a young female American English native speaker. In both production and perception, vowels were markedly more challenging than consonants with vowel production and perception scores around 60% and consonant production and perception approximately 90%. Moreover, there was a significant correlation between consonant production and perception, but not between vowel production and perception. These results indicate that Chinse college students' main difficulty in English phonetic production and perception are vowels rather than consonants at simple phonetic contexts like /VCV/ and /CVC/ structures. The second language learning models will be discussed to interpret these findings. [Work supported by the Open Project Fund from the National Key Laboratory of Cognitive Neuroscience and Learning at Beijing Normal University, China, and University of Texas at Austin Research Grant.]

1aSC4. Expansion, contraction, and drift during the development of L2 speech categories. John Matthews (Chuo Univ., 742-1 Higashi Nakano, Hachioji, Tokyo 192-0393, Japan, matthews@tamacc.chuo-u.ac.jp), Takako Kawasaki, and Kuniyoshi Tanaka (Hosei Univ., Tokyo, Japan)

This paper investigates the acquisition of novel segmental categories that are similar to native language segments among two groups of non-native speakers and a native-speaker control group. English and Japanese both contain voiceless alveolar and post-alveolar fricatives, but while the alveolars are effectively identical, the post-alveolars differ in their precise places of articulation. English includes a palato-alveolar while Japanese includes an alveolo-palatal. We elicited pronunciations of these segments in both English and Japanese by a group of intermediate L2 learners of English in Japan and a group of Japanese–English late bilinguals with experience living and studying abroad, plus a third group made up of English native-speakers as a control group. Measurements of Center of Gravity taken at the midpoint of each fricative and F2 taken at the appearance of formant structure immediately following the fricative revealed expansive category ranges among the L2 learners but contracted categories among the bilinguals, with evidence of a new, independent post-alveolar fricative added to their L2 segmental inventory. L2 learners were found to experience persistent transfer effects from phonological neutralization of this contrast in Japanese, whereas bilinguals were found to have overcome such effects.

1aSC5. Perceptual adaptation to an artificial accent: Phonetic category shift or expansion? Yevgeniy Melguy (Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94704, ymelguy@berkeley.edu) and Keith Johnson (Linguist., Univ. of California, Berkeley, Berkeley, CA)

Listeners can rapidly adapt to a novel accent. For example, following exposure to a speaker whose /f/ sound is ambiguous between [s] and [f], they categorize more sounds along an [s]-[f] phonetic continuum as /f/. We tested the adaptation mechanism underlying such changes in category structure — do listeners adjust their phonetic boundary for the target sound (category shift), or simply relax their categorization criteria (category expansion)? We trained listeners on a pronunciation containing ambiguous /θ/ [θ / s] and then tested them on categorizing either [θ]-[s] or [θ]-[f] phonetic continua. If listeners simply shift their /θ/ boundary, we predicted an increase in /θ/ responses for the [θ]-[s] continuum, but not [θ]-[f]. By contrast, if listeners expand the category, we predicted a categorization change in both continua, indicating a general broadening of the /θ/ category in phonetic space. Consistent with the boundary shift hypothesis, we found that listeners tested on the [θ]-[s] continua showed a significant increase in proportion of /θ/ responses versus controls [$\chi^2(1) = 4.80$, $p < 0.05$), while those tested on [θ]-[f] did not [$\chi^2(1) = 1.30$, $p = 0.25$). [Work supported by an NSF GRFP Fellowship to YVM.]

1aSC6. Perception and imitation of prevoicing across language backgrounds. Emily J. Clare (Dept. of Lang. Studies, Univ. of Toronto Mississauga, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, e.clare@mail.utoronto.ca) and Jessamyn Schertz (Univ. of Toronto Mississauga, Mississauga, ON, Canada)

The present study investigates speakers' ability to perceive and imitate prevoicing, testing speakers of languages differing its contrastive status: those where prevoicing serves as a primary cue to the laryngeal contrast (Indo-Aryan languages; "true voicing" languages, e.g., Tagalog), those where prevoicing never occurs ("aspirating" languages, e.g., Cantonese), and those where prevoicing occurs in free variation (other aspirating languages, e.g., English). After hearing pairs of Hindi words differing in presence/absence of prevoicing, participants were asked to imitate them, followed by an ABX discrimination task. Preliminary findings show above-chance discrimination accuracy as well as significant imitation of the difference in participants all language backgrounds, with place of articulation, consonantal, and vocalic cues all affecting the extent of prevoicing in production. As expected, speakers of languages where prevoicing is absent showed less overall prevoicing than other groups; however, there were not clear differences across language groups in discrimination ability. While faithful imitation was usually associated with accurate discrimination, good discrimination was also found on many tokens that were not faithfully imitated, indicating that other (production-based) factors are necessary to account for variability in imitative ability.

1aSC7. Perception of non-native quantity contrasts by Cantonese and Japanese speakers. Albert Lee (The Education Univ. of Hong Kong, Rm. 9.23 Run Shaw Tower, Pokfulam, Hong Kong, albertlee@hku.hk), Yasuaki Shinohara (Waseda Univ., Tokyo, Japan), and Tsz Ching Mut (The Education Univ. of Hong Kong, Tai Po, Hong Kong)

This paper reports on the findings of a pilot study which investigated Cantonese and Japanese speakers' ability to perceive non-native quantity contrasts. Our goal was to examine whether L1 phonology affects the perception of non-native sound contrasts at a discrete, categorical level or a gradient, "featural" level. We recruited native Cantonese and Japanese speakers for a series of AXB discrimination and identification tasks. Cantonese has partial quantity contrasts (i.e., where duration is not the primary acoustic cue) in a few vowel pairs, whereas Japanese has systematic two-way quantity contrasts in both vowels and consonants. The synthesized stimuli were Japanese (non-native for Cantonese speakers) and Estonian (non-native for both groups) nonce words contrasting in vowel and consonant quantity. For Estonian, quantity is a three-way distinction (short, long, overlong). The results showed that while Japanese speakers outperformed their Cantonese counterparts in discrimination, their identification accuracy for overlong Estonian vowels was anomalously low. These findings are discussed with reference to the "feature hypothesis" in L2 phonological acquisition.

1aSC8. Exploring the role of personality traits in second language speech perception. Amy Hutchinson (Purdue Univ., 640 Oval Dr., West Lafayette, IN 47907, hutchi25@purdue.edu), Sophia Rodriguez (Luther College, Decorah, IA), and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

It is well established that some individual factors (i.e., age, amount of input, motivation, etc.) play a considerable role in second language (L2) speech perceptual learning (Akahane-Yamada, 1995; Flege *et al.*, 1997; Flege & Liu, 2001; inter alia). However, other factors, like personality type, have received less attention. This study therefore investigates the role of personality in L2 speech perception by examining how factors assessed in the Big 5 Inventory (Extraversion, Conscientiousness, Agreeableness, Neuroticism, and Openness; John *et al.*, 1991) and the Autism Spectrum Quotient (social skill, attention-switching, attention to detail, communication, and imagination; Baron-Cohen *et al.*, 2001) affect French nasal vowel identification by intermediate/advanced American learners of French ($n = 32$). Preliminary results ($n = 25$) analyzed with a mixed effects

binomial logistic regression revealed that learners with higher scores in Extroversion and Neuroticism were significantly more likely to select the correct nasal vowel. The analysis also demonstrated that L2 learners who were more careful, attentive, and diligent (represented by high scores in Conscientiousness and/or Attention to Detail) were significantly less target-like in their perception. In addition to presenting results from the full data set, this presentation will also discuss implications for L2 learning and L2 speech perception more broadly.

1aSC9. The evolution of vowel perception in second language learners of Spanish. Daniel Bates (Modern Lang., Florida State Univ., 600 W. College Ave., Tallahassee, FL 32306, dkb17b@my.fsu.edu)

Given the importance of understanding speech perception in language acquisition research (Best, 1991) and building on the methodology of Garcia de las Bayonas (2004), this remotely conducted study explores the acquisition of correct Spanish vowel perception by English-speaking learners with special attention paid to the role of syllable stress. Participants include Spanish speakers (n = 4), English speakers (n = 7), and beginning Spanish students (n = 9). In a discrimination task, learners performed most accurately with Spanish vowels in stressed syllables (88.6%) with less accuracy discriminating those in unstressed syllables (82.2%). This is likely due to the tendency of American English to reduce many unstressed vowels. In an identification task, participants selected vowel sounds (from English and Spanish inventories) believed to belong in Spanish words. High accuracy was shown identifying Spanish vowels /i/, /o/, and /u/ but all groups struggled to correctly identify /e/ and /a/, likely because these vowels occupy similar acoustic spaces as English vowels /ɛ/ and /æ/. Despite imperfect perception, learners report high confidence in their responses, indicating under-developed vowel inventories in the target language. At the time of presentation, these results will include larger and more diverse participant groups, including students at different stages of experience with the target language.

1aSC10. Working memory, selective attention, and input variability differentially contribute to non-native phonetic learning. Xiaojuan Zhang (English Dept. & Lang. and Cognit. Neuroscience Lab, School of Foreign Studies, Xi'an Jiaotong Univ., No. 28, Xianning West Rd., Xi'an, Shaanxi 710049, China, xiaojuan.110577@stu.xjtu.edu.cn), Bing Cheng (Xi'an Jiaotong Univ., Xi'an, China), and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Our recent training study demonstrated that acoustic variability, independent of talker variability, promotes nonnative phonetic learning with an enhanced audiovisual adaptive protocol of high variability phonetic training (HVPT). In this follow-up study, we further investigated how cognitive factors may contribute to the learning process. Forty native Chinese adults were randomly assigned to a multiple-talker (MT) group and a single-talker (ST) group to learn the English /i/-/ɪ/ contrast. Pre- and two-post-tests (one week after training and three months after training) employed the natural word identification task. Participants were also tested on working memory and selective attention prior to training. Consistent with previous results, MT and ST training produced similar improvements in the immediate post-test. While significant retention was observed only in the ST group after three months, retention was also found in some participants in the MT group. More importantly, in the MT group, working memory capacity predicted the immediate improvements, while selective attention predicted the amount of retained training effects. No such associations were found in the ST group. These results added new evidence for the differential roles of working memory and selective attention in speech learning. Learners with greater working memory abilities and attentional resources are more likely to benefit from increased acoustic variability in HVPT.

1aSC11. Language dominance affects auditory translation priming in heritage speakers. Rachel Soo (Linguist., Univ. of BC, 100 St. George St., Toronto, ON M5S 3G3, Canada, soorache@gmail.com) and Philip J. Monahan (Univ. of Toronto, Toronto, ON, Canada)

Late L2 learners show translation priming from the first language to the second (L1–L2), while L2–L1 effects are inconsistent (Altarriba & Basnight-Brown, 2007). Typically, late L2 learners are both less dominant in the L2 and have a later L2 age of acquisition, making the relative contribution of language dominance and age of acquisition in L2–L1 priming unclear. We test Cantonese heritage and native speakers in an auditory translation priming paradigm. As heritage speakers first learn Cantonese (L1) but later become more proficient in English (L2), this profile potentially allows for the dissociation of dominance and age of acquisition in translation priming. If age of acquisition is the primary factor, more priming is expected to occur in the L1–L2 (Cantonese-English) direction; however, if dominance plays a stronger role, priming is expected to occur in the L2–L1 (English-Cantonese) direction. Preliminary results indicate that native speakers show L1–L2 but not L2–L1 priming, consistent with previous findings, while heritage speakers show priming in both directions, but stronger L2–L1 priming. The inter-condition difference is greater for native speakers. In short, age of acquisition plays a role in bilingual language processing (Silverburg & Samuel, 2004) and potentially drives auditory translation priming effects more strongly than language dominance.

1aSC12. Examining the success of three kinds of cross-language similarity in predicting English listeners' discrimination of Danish vowels. Ocke-Schwen Bohn (Dept. of English, Aarhus Univ., Aarhus, Denmark, ocke.bohn@cc.au.dk) and Camila L. Garibaldi (Horsens Municipality, Horsens, Denmark)

Several methods have been used in second language speech research to predict nonnative listeners' discrimination of foreign language speech sounds. The present study examined how well three of these methods predict the discrimination of the four tightly spaced Danish unrounded front vowels by native listeners of Standard Southern British English (SSBE), whose four front vowels are evenly spaced along the close-open dimension. Experiment 1 examined the *ecphoric similarity* of the Danish and English vowels in a perceptual assimilation task in which L1 SSBE listeners used native labels and goodness-of-fit ratings in identifying the Danish vowels. Experiment 2 examined the *perceptual similarity* of the English and Danish vowels in a graded discrimination task in which the same listeners rated the degree of dissimilarity of Danish-English vowel pairs. The third prediction of discriminability was derived from an *acoustic comparison* of the Danish and English vowels in the F1/F2 space. The same participants as in Experiments 1 and 2 then discriminated three Danish front vowel contrasts. None of the three methods was found to fully successfully predict discrimination of the Danish vowels by the SSBE listeners. The presentation will discuss reasons for the lack of predictive power of the three widely used methods.

1aSC13. The impact of masker language and talker intelligibility on early bilinguals listeners' speech perception. Jailyn M. Pena (Linguist., New York Univ., 21-15 35th St. 1A, Astoria, NY 11105, jmp987@nyu.edu) and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Most real-life speech perception occurs in noisy environments. Previous research shows that listeners are better at understanding target speech that is masked by competing speech (babble masker) in a different (mismatched) language compared to when the target speech and babble are in the same (matched) language, a phenomenon referred to as "release from masking." Here, we investigated release from masking for monolingual English and early Korean-English bilingual listeners. We also investigated whether talker intelligibility mediates release from masking. Listeners completed a sentence recognition task with both matched and mismatched babble. Critically, the early bilinguals heard English and Korean target speech. Results indicate that early bilinguals did not differ significantly from monolinguals when listening to English. Interestingly, when listening to Korean, early bilinguals also received a release from masking, despite the babble being their dominant language (English). Overall, these results indicate that early bilinguals may be able to tolerate similar levels of adverse listening conditions as monolinguals in their dominant language. Furthermore, they are able to benefit from a target-babble linguistic mismatch in both their dominant and nondominant languages. Finally, across both languages, low-intelligibility talkers conditioned larger release from masking effects than high-intelligibility talkers, something previous research has not accounted for.

1aSC14. Real-time visual acoustic feedback in computer-assisted pronunciation training. Ivana Rehman (English, Iowa State Univ., 4716 Hutchison St., Apt. 2, Ames, IA 50014, ilucic@iastate.edu)

Previous research on visualization of speech segments for pronunciation training suggests that such learning results in improved segmental production (e.g., Katrushina *et al*., 2015; Olson, 2014; Patten & Edmonds, 2015). However, investigation into real-time formant visualization for L2 vowel production training has been limited to either training a single or a pair of vowels (Carey, 2004; Sakai, 2016) and to examination of improvement on trained items only (Katrushina *et al*., 2015). This project investigates the effects of real-time formant visualization on production training for eight L2 vowels in trained and untrained environments as well as spontaneous speech. L2 learners (n = 11) participated in nine 30-minute training sessions, during which they used a formant visualization system to practice their vowel production. A control group (n = 8) was involved in audio-only vowel production training. Pre-test, post-test, and delayed post-test design was used, and pronunciation improvement was analyzed acoustically using Mahalanobis distance and mixed-effects modeling. The use of real-time visual acoustic feedback resulted in retained improvement in vowel quality in both trained and untrained items than audio-only training. Spontaneous speech was not improved. The findings suggest that this system could be used as an effective pedagogical tool for L2 learners.

1aSC15. Learning non-native speech stop voicing as a result of exposure to foreign film. Amy Hutchinson (Purdue Univ., 640 Oval Dr., West Lafayette, IN 47907, hutchi25@purdue.edu) and Olga Dmitrieva (Purdue Univ., West Lafayette, IN)

While access to authentic input from native speakers is critical when learning to produce speech in another language (Flege *et al*., 1997; Flege *et al*., 1995; MacKay *et al*., 2001; inter alia), this option is not always accessible to all learners. With that in mind, this study explores how another type of naturalistic exposure, foreign film, can contribute to the development of L2 speech by specifically examining non-native stop voicing. To explore this research question, French speech shadowing data from naïve, monolingual English participants (n = 59) was collected before and after watching a short film in French. Collected productions were analyzed acoustically (voice onset time [VOT]) and through a series of AXB-style native French listener perceptual judgements (n = 221). While the acoustic analysis of VOT did not reveal a significant effect of film on duration when compared to a control group (n = 15), native listeners perceived post-film exposure productions to be significantly more target-like than pre-film exposure productions. This finding suggests that although VOT was seemingly unaffected by foreign film exposure, participants adjusted alternative acoustic correlates of voicing and that these modifications were perceptible to native listeners. This ultimately provides evidence for the effectiveness of foreign film exposure in non-native speech learning.

1aSC16. Investigating the effect of second language learning on the acquisition of a third language rhythm pattern. Carissa A. Diantoro (Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, carissad@uoregon.edu), Melissa A. Redford, and Jeffrey Kallay (Linguist., Univ. of Oregon, Eugene, OR)

Language transfer helps in the acquisition of a second language (L2) when the target L2 shares similarities with the first language (L1). We ask: Does proficiency in an L2 facilitate the acquisition of a structurally similar L3? Or, does training in any L2 help with L3 acquisition? We investigated L2 learners' ability to reproduce the rhythm of a novel L3, Indonesian, as a function of their exposure to a rhythm-similar language (French) or a rhythm-divergent language (German). Twelve English-speaking students with no college-level language served as control group; six 2nd-year French students served as the rhythm-similar group; and 6 2nd-year German students as the rhythm-divergent group. All produced French, German, and Indonesian sentences in a repetition task with native speaker prompts. Listeners then rated the similarity between their sentences and that of a native speaker. Interval-based rhythm metrics were also computed for the sentences. L2 groups were better at their L2 language than the control group based on the ratings and rhythm measures. They were also perceived as producing better L3 sentences than the control group. Overall, the results suggest that language learning may facilitate subsequent language learning; the effect is not limited to language transfer. [Work supported by NIH.]

1aSC17. Prosodic and segmental information effects on perceived accentedness of native and non-native speech. Kurtis Foster (Linguist., Univ. of Oregon, Straub Hall, 1451 Onyx St., Eugene, OR 97403, kurtisf@uoregon.edu) and Melissa M. Baese-Berk (Linguist., Univ. of Oregon, Eugene, OR)

Studies on the effects of prosodic information on perception of accent have investigated the ability of listeners to distinguish between accent varieties using prosodic cues in the absence of segmental information. The present study examined how prosodic and segmental information impact perception of accentedness. Specifically, we examined how listeners perceive accentedness of low-pass filtered and unfiltered speech, as low-pass filtered speech isolates prosody while removing segmental information. Native speaker English participants were exposed to speech from 1 of 4 conditions: unfiltered English produced by native English talkers, low-pass filtered English produced by native English talkers, unfiltered English produced by native Mandarin talkers, or low-pass filtered English produced by native Mandarin talkers. Participants rated each sentence with Likert scale responses from 1 (native speaker of English with no non-native accent) to 9 (strong non-native accent). Unfiltered native English speech received the lowest accentedness ratings and unfiltered Mandarin the highest. Low-pass filtered speech from all talkers was rated as intermediate between the two unfiltered conditions regardless of talker. These findings suggest that when making assessments of speaker accentedness, listeners make finer distinctions through the use of segmental information than they do with prosodic information.

1aSC18. Limits of Cantonese advantage on English stress discrimination: Rising pitch accent pattern and vowel reduction. William Choi (Human Commun., Development, and Information Sci., The Univ. of Hong Kong, Rm. 765, Meng Wah Complex, Hong Kong 852, Hong Kong, willchoi@hku.hk)

Can non-natives outperform natives on speech discrimination? Surprisingly, Cantonese listeners discriminated English stress more accurately than English listeners did. To ascertain its generalizability, I further asked whether this Cantonese advantage on English stress discrimination was equally potent across pitch accent and vowel reduction contexts. Sixty Cantonese and English listeners completed four blocks of English stress discrimination task with varying pitch accent and vowel reduction contexts. In the absence of rising pitch accent pattern and vowel reduction, the Cantonese listeners outperformed the English listeners on discriminating English stress. However, the Cantonese advantage disappeared when either rising pitch accent pattern or vowel reduction was present. When both rising pitch accent pattern and vowel reduction were present, the Cantonese listeners even performed poorer than the English listeners did. The findings underscore two constraints of the Cantonese advantage on English stress discrimination—rising pitch accent pattern and vowel reduction.

Session 1pAA

Architectural Acoustics and Noise: Back to Basics: Skill Building Seminar II

Eric Reuter, Chair
Reuter Associates, LLC, 10 Vaughan Mall, Suite 201A, Portsmouth, NH 03801

Invited Paper

1:10

1pAA1. Environmental noise. Eric Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

This session will introduce participants to the principles of measurement, analysis, and control of environmental noise. Topics will include the selection and operation of sound level meters, statistical descriptors, ordinances and regulations, standards, propagation modeling, and sound barrier applications. No prior experience with environmental noise is assumed.

2:25–2:55
Panel Discussion

MONDAY AFTERNOON, 29 NOVEMBER 2021 602 (L)/603 (O), 1:00 P.M. TO 2:00 P.M.

Session 1pABa

Animal Bioacoustics: Population Monitoring

David K. Mellinger, Chair
Coop. Inst. for Marine Resources Studies, Oregon State University,
2030 SE Marine Science Dr., Newport, OR 97365

Contributed Papers

1:00

1pABa1. Recent ecological insights into the endangered Cook Inlet beluga via acoustic monitoring. Manuel Castellote (Univ. of Washington, 7600 Sand Point Way NE, Seattle, WA 98115, manuel.castellote@noaa.gov), Lori Polasek, Justin Olnes (Alaska Dept. Fish and Game, Juneau, AK), Christopher Garner (JBER, Dept. of Defense, Anchorage, AK), Brian Taras (Alaska Dept. Fish and Game, Fairbanks, AK), Arial Brewer (Univ. of Washington, Seattle, WA), Justin Jenniges (Alaska Dept. Fish and Game, Juneau, AK), and Tom Gage (Alaska Dept. Fish and Game, Anchorage, AK)

The endangered Cook Inlet beluga population is non-migratory and located in south-central Alaska. Estimated at <300 individuals, it is declining more rapidly than previously thought. The Recovery Plan highlights a paucity of information on basic ecology impeding proper management decisions. The Alaska Department of Fish and Game and the National Marine Fisheries Service are collaborating on a passive acoustics monitoring program to address crucial information gaps. Year-round seasonal distribution and feeding occurrence is described for key areas within the critical habitat. Acoustic detections are greater in the upper inlet during summer, peaking in known concentration areas. Foraging peaks coincide with the presence of various anadromous fish runs from spring to fall. Low levels of feeding activity in winter suggest a lack of feeding aggregations in these areas. Thirteen sources of anthropogenic noise occur within the critical habitat, many exceeding behavioral harassment levels. Using a case-controlled approach in which periods with and without anthropogenic noise were compared, a significant negative correlation between noise levels and vocalization

detections was found, However, masking by flow noise due to the strong currents in Cook Inlet might be affecting these results more than a true behavioral response. Further research remains ongoing to confirm/reject this hypothesis.

1:15

1pABa2. Exploring sperm whale population demographics in the northern Gulf of Mexico using passive acoustic monitoring. Renea Briner (School of Marine Sci. and Policy, Univ. of Delaware, REU Site: Dept. of Phys., University of Louisiana at Lafayette, UL Box 43680, Lafayette, LA 70504-3680, rbriner@udel.edu), Natalia Sidorovskaia, and Naomi Mathew (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA)

Sperm whales are a sexually dimorphic species widely distributed throughout the world's oceans, with social units of mature females and young thought to inhabit tropical and temperate waters, while individual mature males travel to higher latitudes in small bachelor groups or alone. Seasonal population demographics, particularly sex distributions, and the group structures of sperm whales have not been closely analyzed for the Gulf of Mexico. Utilizing the unique multi-pulse structure of sperm whale echolocation clicks, their total body length can be estimated, and, as a result of their extreme sexual dimorphism, their sex can be determined. Continuous acoustic recordings from June 2018 through June 2020 by the Environmental Acoustic Recording System (EARS) buoys, deployed off the coast of Louisiana, were analyzed using CABLE software [Beslin *et al.*, *JASA* (2018)]. The results suggest that the sperm whale population is dominated by females in both summer and winter seasons, although mature whales of both sexes were detected in both seasons. Additionally, for each year of study, respectively, the mean size and the total number of whales detected were greater in winter months. Further insights into group structures will be discussed in the presentation. [Research supported by NSF REU (Award No. 1659853) and BOEM.]

1:30

1pABa3. Acoustical characteristics of avian choruses in a western migratory bird refuge. Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmoats359@gmail.com), Mylan R. Cook, Claire L. Teusch (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (Dept. of Biology, California State Univ. Bakersfield, Bakersfield, CA), Kent L. Gee, and Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Migratory bird refuge soundscapes are seasonally dynamic due to changes in wildlife populations. Some of the most prominent acoustical events in a bird refuge are the morning and evening avian choruses, particularly during spring and early summer when breeding activity of birds is high. For this study, near-continuous spectral data were collected at the U.S. Federal Fish and Wildlife Services Bear River Migratory Bird Refuge. Although data fidelity can be compromised by wind and rain, changes in the avian chorus characteristics over time are observed. Likely correlations with avian migration and refuge land management are discussed.

1:45

1pABa4. Large whale surveys off Southern California using passive acoustic gliders. David K. Mellinger (Cooperative Inst. for Marine Ecosystem and Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Selene Fregosi (Cooperative Inst. for Marine Ecosystem and Resources Studies, Oregon State Univ., Newport, OR), and Kristen Ampela (HDR, Inc., San Diego, CA)

A passive acoustic survey off Southern California was conducted using gliders measure the distribution of several baleen whales and sperm whales. Two gliders were flown to estimate the occurrence of blue, fin, humpback, minke, and sperm whales in over the continental shelf, shelf slope, and deep waters offshore of the shelf slope up to 350 km from shore. One glider malfunctioned shortly after deployment and had to be recovered; the second followed its desired $\sim$1200 km trackline over 54 days. Whale vocalizations were detected using a mix of automated and manual tools. Fin whale sounds were detected every day, and humpback whale sounds were detected frequently but more sporadically, along the route, particularly over the continental shelf. Sperm whale sounds were detected episodically over deep ($>$2000 m) waters and briefly on the shelf. Blue whale sounds were detected on three days over the shelf, and minke whale sounds were not detected at all. [Funding from Navy NAVFAC.]

Session 1pABb

Animal Bioacoustics: Analysis and Classification of Animal Bioacoustics Signals

Marie J. Zahn, Chair
School of Aquatic and Fishery Sciences, University of Washington, 1122 NE Boat St., Seattle, WA 98105

Contributed Papers

2:20

1pABb1. Acoustic differentiation and classification of wild belugas and narwhals using echolocation clicks. Marie J. Zahn (School of Aquatic and Fishery Sci., Univ. of Washington, 1122 NE Boat St., Seattle, WA 98105, mzahn@uw.edu), Shannon Rankin (Southwest Fisheries Sci. Ctr., National Marine Fisheries Service, NOAA, La Jolla, CA), Jennifer L. McCullough (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI), Jens C. Koblitz (Max Planck Inst. of Animal Behavior, Konstanz, Germany), Frederick Archer (Southwest Fisheries Sci. Ctr., National Marine Fisheries Service, NOAA, La Jolla, CA), Marianne H. Rasmussen (The Univ. of Iceland's Res. Ctr. in Húsavík, Húsavík, Iceland), and Kristin L. Laidre (Polar Sci. Ctr., Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Belugas (*Delphinapterus leucas*) and narwhals (*Monodon monoceros*) are highly social toothed Arctic cetaceans with large vocal repertoires and similar acoustic profiles. Passive Acoustic Monitoring (PAM) that uses multiple hydrophones over large spatiotemporal scales has been the primary method to study their populations, particularly in response to rapid climate change and increasing underwater noise. This study marks the first acoustic comparison between wild belugas and narwhals from the same location and reveals that they can be acoustically differentiated and classified solely by echolocation clicks. Acoustic recordings were made in the pack ice of Baffin Bay, West Greenland during 2013. Multivariate analyses and Random Forests classification models were applied to eighty-one single-species acoustic events comprised of numerous echolocation clicks. Results demonstrate a significant difference between species' acoustic parameters where beluga echolocation was distinguished by higher frequency content. Classification model predictive performance was strong with an overall correct classification rate of 97.5% for the best model. The most important predictors for species assignment were defined by peaks and notches in frequency spectra. Our results provide strong support for the use of echolocation in PAM efforts to differentiate belugas and narwhals.

2:35

1pABb2. Envelope modulation as a communicative cue mediating male–male interactions in bullfrogs. Andrea M. Simmons (Brown Univ., 190 Thayer St., Box 1821, Providence, RI 02912-9067, Andrea_Simmons@brown.edu) and Laura Kloepper (Saint Mary's College, Notre Dame, IN)

Chorusing male bullfrogs (*Rana catesbeiana*) modify the envelope waveforms of the individual notes in their advertisement calls in a stereotyped pattern. Natural advertisement calls begin with an unmodulated first note, with subsequent notes typically exhibiting increasing numbers of modulations. We asked whether these envelope modulations serve a male-male communicative function. In a playback design, we presented to actively chorusing males a series of notes with varying numbers of envelope modulations. We quantified the numbers of vocal responses as well as the total acoustic energy in calls of all males in the chorus. Over 12 experiments at two chorus sites, neither numbers of responses nor total energy differed significantly between stimuli. These two measures were highly correlated, indicating that the energy measure captured chorus dynamics. Modifications of envelope waveform do not appear to convey information to other males, and instead may be a byproduct of vocal production mechanics.

2:50

1pABb3. Marine mammal phonations of Barkley Canyon: A publicly available annotated data set. Jasper Kanes (Sci. Services, Ocean Networks Canada, 2474 Arbutus Rd., Victoria, BC V8N1V8, Canada, ksjkanes@oceannetworks.ca)

Marine mammal phonations were manually annotated in a subset of hydrophone data collected by Ocean Networks Canada from May 2013 to January 2015 to produce a data set that could be used for algorithm development and marine mammal research. Data were collected near Barkley Canyon, a biologically productive submarine canyon approximately 60 km southwest of Vancouver Island that draws aggregations of euphausiids, hake, herring, and larger animals. The data set contains 10 905 annotated phonations from fin whales, blue whales, humpback whales, sperm whales, orcas, Pacific white-sided dolphins, Risso's dolphins, and other delphinids. All three regional orca ecotypes are represented within the data set. Humpback whale vocalizations were found in nearly $1/2$ of all files analyzed, and fin whales were conclusively identified in approximately $1/4$ of files though may be present in up to $1/2$ of files. Blue whale phonations were uncommon and only recorded in the fall and early winter. While sperm whales, Pacific white-sided dolphins, and Risso's dolphins were noted in only 3%–7% of files, they visited the site most days. Orcas were rare visitors to the area. This data set will be made publicly available for further use.

3:05

1pABb4. Evaluation of the Coastal Acoustic Buoy for offshore wind for real time mitigation of North Atlantic right whales. Kaitlin Palmer (SMRU Consulting, 55 Water St., Vancouver, BC V6B 1A, Canada, kjp@smruconsulting.com), Jesse Turner, Sam Tabbutt (SMRU Consulting, Friday Harbor, WA), Douglas Gillespie (SMRU, Univ. of St. Andrews, St. Andrews, Scotland, United Kingdom), Jessica Thompson (SMRU Consulting, Friday Harbor, WI), Paul King, and Jason Wood (SMRU Consulting, Friday Harbor, WA)

A recent Incidental Harassment Authorization requires a 10 km zone to be monitored with passive acoustics before and during pile driving. SMRU Consulting has built the Coastal Acoustic Buoy for Offshore Wind (CABOW) to address this need. Each CABOW consists of: a bottom lander with three hydrophones; data acquisition system; and onboard computer running PAMGuard. The CABOW detects potential calls and relays clips to a base station for classification, bearing estimation, and validation. Bench testing characterized detector performance and measured bearing accuracy as a function of SNR. Field trials used playbacks of real and simulated right whale upcalls to evaluate the detection probability with respect to the range and signal excess of the call; estimate bearing accuracy; and test the range and stability of the radio communications. Field trials indicated a maximum detection range of 4–12 km under high and very low noise levels, respectively. Finally, we used a simulation to compare true and false alarm rates of the CABOW to an equivalent single sensor system for monitoring a fixed

area. Simulations show that incorporating bearings to calling animals could provide a substantial improvement in false alarm rates relative to single sensors if properly placed with respect to potential noise sources.

3:20

1pABb5. Intra-individual variations within humpback whale song sessions: More signs of sonar. Eduardo Mercado (Univ. at Buffalo, SUNY, 4360 Gunnville Rd., Clarence, NY 14031, emiii@buffalo.edu)

Analyses of consecutive songs produced by singing humpback whales recorded off the coast of Hawaii revealed that singers constantly vary the acoustic qualities of their songs across “repetitions. Unlike the progressive changes in song structure that singing humpback whales make across years, intra-individual acoustic variations within song sessions appear to be largely stochastic. Four sequentially produced song components (themes) were each found to vary in unique ways. The most extensively used theme was highly variable in overall duration within and across song sessions, but varied relatively little in frequency content. In contrast, the remaining themes varied greatly in frequency content, but showed less variation in duration. Analyses of variations in the amount of time singers spent producing different themes suggest that the mechanisms that determine when singers transition between themes may be comparable to those that control how long terrestrial animals fixate while scanning visual scenes. The dynamic changes that individual singers make to songs within song sessions are counterproductive if songs serve mainly to provide conspecifics with indications of a singer’s fitness. Instead, within-session changes to acoustic features of songs may serve to enhance a singer’s capacity to echoically detect, localize, and track conspecifics from long distances.

3:35

1pABb6. A study of the vocal behavior of adult bald eagles during breeding and chick-rearing. Joann McGee (Auditory Neurobiology Lab., VA Loma Linda Healthcare System, Loma Linda, CA 92357, mcgeej@umn.edu), Peggy Nelson (Ctr for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Julia B. Ponder (The Raptor Ctr., Univ. of Minnesota, St. Paul, MN), Christopher Feist, Christopher Milliren (St. Anthony Falls Lab., Univ. of Minnesota, Minneapolis, MN), and Edward J. Walsh (Auditory Neurobiology Lab., VA Loma Linda Healthcare System, Loma Linda, CA)

Previous efforts to characterize the spectral and temporal properties of calls contained in the vocal repertoires of adult bald and golden eagles, *Haliaeetus leucocephalus* and *Aquila chrysaetos*, respectively, led to the generation of a catalog of call types with distinguishing acoustic properties (McGee *et al.*, 2019). Calls analyzed as part of that investigation represent vocal activity produced by captive, adult birds, some of which are long-standing residents of the Raptor Center at the University of Minnesota, as well as temporary residents undergoing rehabilitation from injury. Here, we compare the spectral and temporal properties of those calls with calls produced by free-ranging adult bald eagles recorded along with video during nest preparation, courtship, egg incubation and chick-rearing activities at nest sites monitored by the Raptor Resource Project. One goal of this project is to determine if the vocal repertoire of bald eagles studied in captivity is representative of that of free-ranging wild eagles. Another goal is to determine the extent to which spectrotemporal characteristics of calls recorded from captive and free-ranging birds overlap. Findings from this study will provide guidance in ongoing efforts to develop acoustic alerting signals designed to minimize the risk of eagle-wind turbine collisions.

Session 1pAO

Acoustical Oceanography and Signal Processing in Acoustics: Acoustical Oceanography Using Ocean Observatory Systems I

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

Andone C. Lavery, Cochair
AOPE, Woods Hole Oceanographic Institution, 98 Water Street, Woods Hole, MA 02543

Felix Schwock, Cochair
Electrical and Computer Engineering, University of Washington, 185 Stevens Way, Paul Allen Center – Room AE100R, Seattle, WA 98195

Chair's Introduction—1:00

Invited Papers

1:05

1pAO1. Combined acoustical and seismic studies of Axial Seamount. William S. Wilcock (School of Oceanogr., Univ. of Washington, Box 357940, Seattle, WA 98195, wilcock@uw.edu)

The NSF Ocean Observatories Initiatives Regional Cabled Array at Axial Seamount provides a remarkable facility for sustained studies of an active submarine volcano. A local seismic network supports a real-time earthquake catalog and hydrophones monitor sounds in the water column. In April 2015, Axial Seamount erupted shortly after the cabled observatory came on line. The seismic record of brittle deformation in the summit caldera during the eruption is complemented by acoustic observations that tracked a dike propagating along the north rift using T phases and recorded both impulsive and continuous sound sources on the seafloor at the sites of erupting lava. Seafloor pressure gauges are presently documenting the reinflation of the volcano towards its next eruption and these geodetic studies would be enhanced by deploying a transponder network to support horizontal ranging. The availability of real-time data to detect and localize future eruptions will enable effective autonomous and ship-based responses to study how hydrothermal systems and their chemosynthetic biological communities respond to eruptions. The seismic and acoustic networks could also support studies of fin and blue whales to understand whether their distribution is modulated by the enhanced productivity associated with the presence of a hydrothermally active seamount.

1:25

1pAO2. Resident autonomous underwater vehicles for persistent mobile ocean observing. Dana Manalang (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, manalang@uw.edu)

Whether near shore or in the open ocean, stationary ocean observing systems lack the ability to respond spatially to transient and difficult-to-predict phenomena, such as mid-ocean ridge eruptions, instabilities in methane hydrate deposits, turbidite flows, sea–ice interactions, mass wasting events, or changes in coastal currents. Detecting and characterizing these types of events and the processes driving them requires mobile systems available at the time of the event onset. Ideally, mobile autonomous underwater vehicles (AUVs) would permanently be stationed in or near areas where events of interest are likely to occur. Traditionally, due to their dependence on batteries, AUV mission durations are often measured in hours and require costly on-water support vessels and human intervention, making it impractical to sustain a mobile observing presence. However, advances in cabled power/comms and renewable marine energy technologies present significant opportunities for developing persistent mobile subsea systems able to recharge their batteries from fixed underwater infrastructure. In addition to power, these "resident" AUVs will require navigation and data transmission support. Acoustic systems will play key roles as sensing systems, navigation, vehicle homing, data transfer, and advanced robotic operations. This talk will present an overview of the systems and concepts relevant to subsea mobile ocean observing systems along with example applications.

1:45

1pAO3. Six years of low-frequency underwater noise off the Oregon coast from the OOI Regional Cabled Array System (RCA) and interpretive linkage with RCA Oceanographic Data. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Approximately 6 years of underwater noise data recorded from the Regional Cabled Array (RCA) network, at the site known as Hydrate Ridge, is studied for purposes of examining long term trends. The acoustic data originate from station HYS14 located on the seabed at a depth of approximately 800 m and 80 km offshore of Newport, OR. The sample rate is 200 Hz, setting the analysis limit to less than 100 Hz. Seasonal warming results in a highly repeatable annual change in the levels at shipping frequencies (50–70 Hz) of order 4 dB which is out-of-phase with concurrent observations of shipping density. Using data from the shallow-water profiler (RCA station SF03A) the influence seasonal warming on the acoustic observations are studied. It is shown how higher ocean surface temperatures cause mode excitation support to move deeper and away typical ship source depths resulting in less efficient transmission of underwater sound. Deviations in the seasonal trends are also observed and shown to be correlated to atypical climatic events such as the warm water blob event of 2015–2016, and suppressed shipping activity during economic downturns, such as observed during COVID-19.

Contributed Papers

2:05

1pAO4. An overview of ambient sound using OOI hydrophone network. John Ragland (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA, jhrag@uw.edu), Felix Schwock (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA), Matthew Munson, and Shima Abadi (Univ. of Washington, Seattle, WA)

The NSF-funded Ocean Observatories Initiative (OOI) is a large-scale project that provides a unique observation of the ocean, collecting acoustic, meteorological, and oceanic datasets. Specifically, for hydrophone data, there are five low frequency hydrophones (Fs = 200 Hz), and six broadband hydrophones (Fs = 64 kHz) dispersed around the north-east pacific. This acoustic dataset presents numerous potential areas of study in the field of ambient sound analysis. In this talk, we analysis 6 years of acoustic data to identify prominent features that are present in the OOI acoustic dataset. Some notable occurrences that are observed in the dataset include volcanic activity, rain and wind noise, fin whale vocalizations, and shipping events. For all low frequency hydrophones and four of the six broadband hydrophones, we will present long-term spectrograms, time-series trends for different spectral bands, different acoustic features that are present in the dataset, and different statistical metrics about the acoustic environment. We find that 6-year, acoustic, trends vary depending on the location of the hydrophone and the spectral band that is observed. Some locations and spectral bands see increases in spectral levels while others see decreases in spectral levels over the course of the 6 years. [Work supported by ONR.]

2:20

1pAO5. Soundscape around the Dongsha reef atoll at the northern South China Sea. Barry B. Ma (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, barry@apl.uw.edu) and Linus Y.-S. Chiu (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

The Dongsha reef atoll is on the direct path of trans-basin large internal solitary waves (ISWs) at the northern South China Sea. The ISWs bring nutrient-rich bottom water to the surface enriching marine life around the atoll. This is an important ecosystem for the ambient noise study to categorize the source of underwater sound and general soundscape. Four Passive Miniature Acoustic Recorders (PMARs) developed by the Applied Physics Laboratory, University of Washington were deployed in 2018 and 2019 within and near the outer bank of reef ring at the depth of ~10 m. The PMARs collected the ambient noise from 100 to 50 kHz about 2–3 months each year. There are clear diurnal signals in pre-dawn and dusk periods with a peak frequency at 10 kHz. The noise level is about ~5 dB higher than the background noise. The outer reef has a higher noise level (~8 dB) than the inner reef due to stronger biological activities. In general, biological noise is the dominant source in this reef ecosystem, yet during the typhoon Mangkhut the wind noise is much higher at the lower frequency (below 3 kHz) and the near-surface bubbles layer generated during the typhoon inhibited the sound transmission which causes the lower noise level above 5 kHz.

2:35–2:50 Break

2:50

1pAO6. Understanding Gulf of Mexico soundscapes. Naomi Mathew (Phys., Univ. of Louisiana219 Rosemary PL, Lafayette, LA70508, naomi.mathew1@louisiana.edu) and Natalia Sidorovskaia (Phys., Univ. of Louisiana, Lafayette, LA)

The contributors, defining baseline soundscapes in the Gulf of Mexico, are analyzed exploring the subtle changes in ambient noise due to the pandemic impact on the maritime and energy industry. The acoustic data were collected by five Environmental Acoustic Recording System buoys in the Mississippi Valley/Canyon region between May 2018 and June 2020, as a part of the long-term observational efforts. In April 2020, the 63-Hz and 125-Hz decidecade band levels were among the lowest in two years. On the contrary, noise levels in the 250-Hz and 500-Hz bands stayed high, probably indicating the presence of the fleet, transporting essential goods, and fishing industry. Marine mammal's acoustic presence exhibits strong seasonal patterns (with the maximum band levels during the winter months) and was not impacted by the COVID-19 closures. To interpret the trends, shipping traffic density, vessel category metadata, marine mammal detections, and weather data are correlated with noise dynamics. April 2020, the first COVID-19 closure month, provided a rare opportunity to measure baseline levels in bands usually dominated by the energy industry activities. These levels are important for future monitoring efforts and understanding overall GoM noise environment on the Wenz curve scale. [Research funded by BOEM via HDR.]

3:05

1pAO7. Minimal COVID-19 quieting measured in the deep, offshore waters of the U.S. Outer Continental Shelf. Jennifer Miksis-Olds (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu), S. B. Martin (JASCO Appl. Sci., Halifax, NS, Canada), Kim Lowell (Univ. of New Hampshire, Durham, NH), Christopher Verlinden, and Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA)

The Atlantic Deepwater Ecosystem Observatory Network (ADEON) provided an opportunistic dataset to examine potential COVID-19 effects in a deep, offshore region of the US southeastern Outer Continental Shelf (OCS) to determine if the quieting documented in coastal soundscapes extended to deep, offshore waters. A two-year time series of one-minute sound pressure level averages was calculated from 2019–2020 at the seven ADEON sites extending along the OCS from Virginia to Florida. Sites ranged in depth from approximately 200–900 m and in distance to the shelf break and shipping lanes. Sound pressure level metrics in decidecade, shipping-related bands were compared between years. Results indicate that the drastic drop in ocean sound levels experienced in coastal areas did not extend to all the ADEON sites. Sound level decreases were observed at sites closest to the shelf break and shipping lanes, while relatively no change was observed at other sites. These observations are consistent with AIS vessel

tracks within 100 km of each ADEON bottom lander that showed increased numbers in 2020 compared to 2019 at majority of the offshore sites. [Study concept, oversight, and funding provided by BOEM, in partnership with ONR and NOAA. Funding for ship time was by ONR, Code 32.]

3:20

1pAO8. Measuring the soundscape of the Eastern United States outer continental shelf using an array of autonomous observatories. S. B. Martin (Halifax, JASCO Appl. Sci., Halifax, NS, Canada, bruce.martin@jasco.com), Katie Kowarski, Colleen Wilson (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), Michael A. Ainslie (Germany, JASCO Appl. Sci., Hesse, Germany), Kevin D. Heaney (Univ. of New Hampshire, Fairfax Station, VA), and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

The Atlantic deep-water Ecosystem Observation Network project (https://adeon.unh.edu/) deployed seven autonomous long-term observatory landers in waters between 250 and 900 m deep along the east coast of the United States for three years (November 2017–November 2020). The objectives of the project included quantifying the variability of the soundscape along and across the outer continental shelf in time and frequency. An early objective of the project was publication of a soundscape standard, whose recommendations were subsequently applied to the data analysis. Here we show how monthly probability density functions of the 20, 80, 630, and 3150 Hz decidecade bands provided insight to the contribution of mammals, vessels, and wind to the measured soundscapes. At some stations the soundscape measurements were hampered by flow noise at low frequency. Clear relationships between stations by month and location changed with frequency band. Application of automated detectors and a standardized approach for validating their outputs provided information on the spatial and temporal presence of marine mammals. The application of standardized methods accelerates the analysis of large passive acoustic monitoring datasets and facilitates communication of results.

3:35

1pAO9. Modelling the global soundscape: Validating a study of the COVID-19 impact using in-shore and off-shore observatory data. Lanfranco Muzi (Ocean Networks Canada, Univ. of Victoria, 2474 Arbutus Rd., Ste. 100, Victoria, BC V8N 1V8, Canada, muzi@oceannetworks.ca), Christopher Verlinden, and Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA)

This study presents a data-model comparison of the soundscape in the Northeast Pacific Ocean in 2019 and 2020. Focussing on frequencies below 400 Hz, where the soundscape is dominated by shipping, marine mammals (at selected frequencies) and large storms, a parabolic-equation model computes the propagation loss from a grid of points to a four-dimensional, high-resolution grid of receivers. The modelling data are extracted from a dynamic model of the global shipping and wind noise soundscape that utilizes AIS ship-tracking data. A comparison of the modelling results between 2019 and 2020 revealed a marked difference, reasonably due to the COVID-19 global shutdown. These results are compared to measurements from the long-term time series collected by Ocean Networks Canada's in-shore and off-shore cabled observatories in the Salish Sea and Northeast Pacific Ocean.

Session 1pBAa

Biomedical Acoustics, Engineering Acoustics, and Physical Acoustics: Cavitation Nuclei: Bubbles, Droplets, and More II

Stuart Ibsen, Cochair
Biomedical Engineering, Oregon Health and Science University, 3181 S.W. Sam Jackson Park Rd., KCRB 5001.41, Portland, OR 97239

Michaelann Tartis, Cochair
Chemical Engineering, New Mexico Institute of Mining and Technology, 801 Leroy, Socorro, NM 87801

Invited Papers

1:00

1pBAa1. Membrane effects from ultrasound interactions with microbubbles—From drug delivery to mechanotransduction. Stuart Ibsen (Biomedical Eng., Oregon Health and Sci. Univ., 3181 S.W. Sam Jackson Park Rd., KCRB 5001.41, Portland, OR 97239, ibsen@ohsu.edu)

Ultrasound is an ideal modality to interact noninvasively with tissues deep within the body because of its significant penetration depth. However, noninvasive low intensity ultrasound on its own has difficulty creating mechanical forces necessary for localized activation of cells through mechanotransduction or rupture of non-echogenic drug delivery vehicles. Lipid-coated microbubbles are particles that provide a high-level of impedance contrast with the surrounding tissue and undergo significant size oscillations when exposed to ultrasound. This imparts mechanical forces on nearby lipid membranes which can be used for a variety of applications. Here, I will discuss the mechanical forces that are created by ultrasound/microbubble interactions that can be used to transduce mechanical forces across intact membranes for cellular activation, and to rupture membranes of drug delivery vehicles. I will also discuss the interaction of lipid debris left over from by microbubble destruction with cell membranes. The cell activation and vehicle release show promise for clinical applications including neuromodulation and cancer chemotherapy delivery.

1:20

1pBAa2. Acoustic vaporization of internalized superheated perfluorocarbon nanodroplets for imaging and macrophage function modulation. Lalit Chudal (Radiology, UT Southwestern Medical Ctr., Dallas, TX), Caroline de Gracia Lux (Radiology, UT Southwestern Medical Ctr., 5323 Harry Hines, Dallas, TX 75390-8514, Caroline.Lux@UTSouthwestern.edu), Jacques Lux, and Robert Mattrey (Radiology, UT Southwestern Medical Ctr., Dallas, TX)

Ultrasound (US) can detect and interact with its contrast agents, allowing perfluorocarbon-based nanodroplets (NDs) or microbubbles (MBs) to detect and treat tumors in deep tissues. Since NDs given IV passively accumulate in tumor macrophages, we aimed to determine whether phagocytosed perfluorobutane (PFB) NDs that boil at -2 °C, can be vaporized *in vivo* using a clinical scanner. Should that be possible, PFB NDs can then be used to deliver drugs or genes to modulate macrophage function. We formulated stable PFB NDs and validated their internalization into THP-1 macrophages using confocal microscopy. PFB ND loaded macrophages were then infused into nude rats and US imaging (low MI US imaging, 18H6 transducer) of the liver and spleen performed *in vivo* using a Siemens Sequoia clinical scanner before and after applying MB destruction pulses (at 1.4 MI, 10L4 transducer). PFB ND-loaded macrophages were not visible using low MI imaging, before applying MB destruction pulses. They immediately vaporized into US visible microbubbles when destruction pulses were applied over the liver and remained visible for many minutes after vaporization. Our current focus is to assess ND-loaded macrophage function before and after ND activation to determine the feasibility of modulating immune function. [Study supported by CPRIT RR150010. RFM is a CPRIT Established Investigator. Siemens ACUSON Sequoia Ultrasound System provided as a loan by Siemens Medical Solutions, USA.]

1:40

1pBAa3. Gas-stabilizing solid cavitation nuclei for systemic or transdermal ultrasound-enhanced drug and vaccine delivery and immunomodulation. Brian Lyons (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Johanna Hettinga, Joel Balkaran, Abigail Collins, Matilde Maardalen, Prateek Katti, Christophoros Mannaris, Luca Bau, Cameron Smith, Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), James Kwan (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Robert Carlisle (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)

Originally developed for diagnostic purposes and already approved for clinical use, lipid and protein-shelled microbubbles were a natural choice as initial nucleation agents for cavitation-based therapies entering the clinic. However, several emerging therapeutic ultrasound applications require nuclei that: (i) are significantly smaller in size, in order to overcome a particular biological barrier such as the leaky vasculature of tumours or the stratum corneum; (ii) offer greatly increased cavitation persistence, both during a single extended ultrasound pulse and in terms of extended circulation following intravenous administration; (iii) have better resilience to sudden ambient pressure changes in order to enable direct injection without nuclei destruction into tissue targets via a needle and syringe; and (iv) are made of materials or have an increased payload or surface area that can interact beneficially with the relevant tissue target during or following cavitation. Gas-stabilizing solid particles will be reviewed in this context, providing an overview of their known characteristics in terms of size distribution, associated acoustic emissions, cavitation thresholds, cavitation persistence and circulation. The relationship between this acoustic characterization and associated bioeffects including drug and vaccine delivery, and immunomodulation will subsequently be explored.

2:00

1pBAa4. Fluorocarbon-assisted nanoparticle-decorated microbubble design. Marie Pierre Krafft (CNRS, Univ. of Strasbourg, Inst. Charles Sadron, 23 rue du Loess, Strasbourg 67034, France, krafft@unistra.fr)

In addition to the use in cardiovascular diagnostic procedures, microbubbles (MBs) have potential in diagnosis of tumours, vascular and blood flow abnormalities, as well as for targeted, ultrasound-triggered drug delivery and as intravascular mechanical devices. All commercial soft, phospholipid-shelled MBs contain a fluorocarbon (*F*C) gas. Both experimental data and theoretical models confirm that the cohesion pressure in phospholipid monolayers is decreased by *F*Cs, supporting a strong shell film fluidizing effect. The co-surfactant activity of *F*C gases towards MB shell components or therapeutic cargos, including polymers, proteins, fluorinated drugs, biomarkers and tracers, will be discussed. A new approach allowed immobilization of fluorinated tracers in phospholipid-shelled MBs. Perfluorohexane allowed preparation of stable spherical hydrophobin HFBII-coated MBs, while only inadequate mixtures of aggregates and elongated MBs are obtained under air. Nanodiamonds, iron oxide and cerium oxide nanoparticles can be driven to the air/water interface when exposed to a *F*C vapor. This new phenomenon enables production of stable nanoparticle-shelled MBs without need for a surfactant. Such microbubbles have potential as multimodal contrast agents (MRI, optical, photoacoustics, PET/ultrasound), as image-guided delivery systems for ultrasound-mediated cancer treatment, and for cell labelling and imaging. (1) M. P. Krafft and J. G. Riess, *Adv. Colloid Interface Sci*. **294**, 102407 (2021).

Contributed Papers

2:20

1pBAa5. Sonobiopsy increases release of circulating tumor DNA for sensitive molecular diagnosis of glioblastoma. Christopher P. Pacia (Washington Univ. in St. Louis, 4511 Forest Park Ave., St. Louis, MO 63108, cpacia@wustl.edu), Jinyun Yuan, Yimei Yue, Lu Xu, Arash Nazeri, Rupen Desai, H. M. Gach (Washington Univ., School of Medicine, St. Louis, MO), Xiaowei Wang (Univ. of Illinois at Chicago, Chicago, IL), Michael Talcott, Aadel Chaudhuri, Gavin Dunn, Eric Leuthardt (Washington Univ., School of Medicine, St. Louis, MO), and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

Though surgical biopsies provide direct access to tissue for genomic characterization, they are invasive and pose significant clinical risks. Brain cancer management via blood-based liquid biopsies is a noninvasive alternative; however, the blood-brain barrier (BBB) restricts the quantities of circulating brain tumor-derived molecular biomarkers necessary for sensitive diagnosis. Focused ultrasound-enabled blood-based liquid biopsy (sonobiopsy) locally disrupts the BBB to release circulating tumor DNA (ctDNA). Magnetic resonance imaging was performed to guide FUS sonication and evaluate the extent of BBB disruption. Droplet digital PCR was used to quantify the levels of circulating tumor DNA. Histological staining and flow cytometry were used to assess off-target tissue damage and release of tumor cells, respectively. Compared to conventional blood-based liquid biopsies, sonobiopsy enriched the ctDNA and improved the detection sensitivity of EGFRvIII from 7.14% to 64.71% and TERT C228T from 14.29% to 45.83% in the mouse glioblastoma model. We developed a porcine GBM model to characterize the translational implications of sonobiopsy and demonstrated that sonobiopsy improved the sensitivity from 28.57% to 100% for EGFRvIII and from 42.86% to 71.43% for TERT C228T. Converging evidence from both glioblastoma models strongly supports the clinical translation of sonobiopsy for noninvasive, sensitive molecular characterization of brain cancer.

2:35

1pBAa6. Non-planar holographic lens for mild hyperthermia in a broad area. Mohamed A. Ghanem (Appl. Phys. Lab., Univ. of Washington, 1013 NE Boat St., Seattle, WA 98105, mghanem@uw.edu), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Lance H. De Koninck (Bioengineering, Univ. of Washington, Seattle, WA), Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Michalakis A. Averkiou (Bioengineering, Univ. of Washington, Seattle, WA)

High intensity focused ultrasound (HIFU) has been used as a non-invasive tool for thermal ablation treatments. Typically, HIFU can ablate a small region (2×10 mm^2) depending on the frequency and transducer dimensions. Cavitation-induced bubbles or injected microbubbles have been shown to enhance tissue heating during HIFU at lower acoustic intensities. The formed lesion is limited to the focal region, necessitating prolonged time of treatment to expose the entire target. To solve this problem, we designed a non-planar diffraction lens that mounts onto the surface of a 900 kHz focused transducer (64 mm diameter and 62 mm focus) to enlarge the focal region. The lens was designed with an iterative angular spectrum approach to achieve a lateral beamwidth of 8 mm while maintaining a similar axial beamwidth. The focal beam size (transverse $\times$ axial) increased from 2.5×15.8 mm^2 to 8×20 mm^2 which also resulted in a corresponding increase in thermal lesion. We plan to expand this work to matrix phased arrays for application to mild hyperthermia for enhanced drug delivery.

Session 1pBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics II

Diane Dalecki, Cochair
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John M. Cormack, Cochair
Center for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Institute, Department of Medicine, University of Pittsburgh Medical Center, Pittsburgh, PA 15261

Contributed Papers

1:00

1pBAb1. A rapid method for pressure-mapping ultrasound fields using a moving hydrophone. David Giraud (Knight Cardiovascular Inst., Oregon Health & Sci. Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239, giraud@ohsu.edu) and Babak Nazer (Knight Cardiovascular Inst., Oregon Health & Sci. Univ., Portland, OR)

Medical ultrasound transducers are commonly characterized in well-controlled water tanks using miniature pressure-calibrated hydrophones. Spatial beam dimensions of the ultrasound field can be quantified by positioning hydrophones in a 2D plane where acoustic waveforms are acquired at closely spaced locations. However, even when using automated computer-controlled positioning, this technique is time consuming if the hydrophone is repositioned and allowed to come to rest at each location where pressure measurements are desired. Here we present a novel technique where acoustic waveforms are acquired while the hydrophone is in motion. The transducer under test is pulsed at a rate which, depending on the hydrophone speed, determines the spatial resolution of the pressure map. Applying this technique results in substantial time savings and allows larger beam areas to be more accurately mapped. Limitations, such as translation stage motor noise and pulse length, are discussed and examples are presented which demonstrate the ability to rapidly characterize a variety of ultrasound transducer types.

1:15

1pBAb2. Comparison of attenuation compensation methods in the estimation of the backscatter coefficient of breast cancer. Laura Castañeda-Martinez (Inst. de Fisica, Universidad Nacional Autonoma de Mexico, Circuito de la Investigacion S/N, Mexico City 04510, Mexico, lcastaneda@ciencias.unam.mx), Hayley Whitson (Dept. of Medical Phys., Univ. of Wisconsin, Madison, WI), Noushin Jafarpisheh (Dept. of Elec. and Comput. Eng., Concordia Univ., Montreal, QC, Canada), Jorge Castillo-Lopez, Hector Galvan-Espinosa, Lesvia Aguilar-Cortazar, Patricia Perez-Badillo, Yolanda Villaseñor-Navarro (Área de Imagen, Instituto Nacional de Canerología, Mexico City, Mexico City, Mexico), Timothy J. Hall (Dept. of Medical Phys., Univ. of Wisconsin, Madison, WI), Hassan Rivaz (Dept. of Elec. and Comput. Eng., Concordia Univ., Montreal, QC, Canada), and I. Rosado-Mendez (Dept. of Medical Phys., Univ. of Wisconsin, Mexico City, Mexico)

Goal: This study compares two strategies to compensate for attenuation in the estimation of the backscatter coefficient σ of breast cancer Methods: Attenuation compensation was performed using local attenuation estimates from a constrained Spectral-Log-Difference (CSLD) method, and average estimates from a dynamic programming (DP)-based regularized method. First, bias and linearity of σ estimates from both strategies were evaluated in a Gammex 410SCG phantom with inclusions with known σ imaged at 8 MHz on a Verasonics Vantage 128 scanner. Then, the strategies were compared in terms of the contrast-to-noise ratio (CNR) between adipose tissue and invasive ductal carcinoma. σ estimates were obtained from 10 patients with biopsy-confirmed IDCs imaged *in vivo* with a 12L4 transducer on a Siemens Acuson S2000 as part of an IRB-approved protocol at the National Institute of Cancerology in Mexico City. DP σ estimates provided smaller bias and better linearity compared to estimates of CSLD. In the *in vivo* study, DP σ estimates produced negative IDC-vs-adipose CNR that was significantly different from zero ($p = 0.002$) and agreed with the hypoechoic appearance of IDC. [Work supported by CONACYT Ciencia de Frontera 1311307, ASA international student support, and UNAM PAPIIT IA102320. We thank Siemens Mexico for scanner loan.]

1:30

1pBAb3. Image-guided focused ultrasound-mediated drug delivery and comparison between 2-D and 3-D therapeutic strategies. Ryan Margolis (Bioengineering, Univ. of Texas at Dallas, 800 W Campbell Rd. BSB, Richardson, TX 75080, ryan.margolis@utdallas.edu), Lokesh Basavarajappa, Junjie Li, and Kenneth Hoyt (Bioengineering, Univ. of Texas at Dallas, Richardson, TX)

The objective of this research was to test a new US image-guided FUS (USgFUS) system for stimulating microvascular permeabilization and enhancing intratumoral drug delivery in volume space. 3-D USgFUS therapy functionality was implemented on a programmable US system with a dual transducer for interleaved imaging and 3-D therapy. 3-D therapy consisted of focal zones distributed over a user-defined volume whereas 2-D treatment had the same focal zones but about a single plane. Using breast cancer-bearing mice ($N = 4$), US therapy was conducted with a mechanical index of 0.5, pulse repetition frequency of 10 Hz, and duty cycle of 10%. Therapy was performed immediately after a 100 μl bolus injection of microbubbles and IR-780 dye. *In-vivo* optical imaging was performed at 0, 24, and 48 h. Tumors were then excised for *ex-vivo* analysis. 3-D USgFUS treatment increased dye extravasation within the tumor space at 24 (245.5%) and 48 h (297.5%) when compared to 2-D US therapy ($p = 0.01$). *Ex vivo* optical images revealed a similar trend with 3-D US treatment producing a higher fluorescent signal than the 2-D approach. Preliminary results suggest that 3-D USgFUS-mediated drug delivery is an exciting prospect for treatment of the entire cancer burden.

1:45

1pBAb4. Focused ultrasound-enhanced delivery of intranasally administered anti-programmed cell death-ligand 1 antibody to an intracranial murine glioma model. Dezhuang Ye (Biomedical Eng., Washington Univ. in St. Louis, 1155 Claytonia Terrace #2N2N, St. Louis, MO 63117, dezhuang.ye@wustl.edu), Jinyun Yuan, Yimei Yue (Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO), Joshua B Rubin (Dept. of Pediatrics and Neurosci., Washington Univ. School of Medicine, St. Louis, MO), and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

Immune checkpoint inhibitors have great potential for treating the gliomas; however, their therapeutic efficacy has been partially limited by their inability to cross the blood–brain barrier (BBB). The objective of this study was to evaluate the capability of focused-ultrasound-mediated intranasal brain drug delivery (FUSIN) in achieving the locally enhanced delivery of anti-programmed cell death-ligand 1 antibody (aPD-L1) to the brain. aPD-L1 was labeled with a near-infrared fluorescence dye (IRDye 800CW) and administered to mice through the nasal route to the brain, followed by focused ultrasound sonication in the presence of systemically injected microbubbles. FUSIN enhanced the accumulation of aPD-L1 at the FUS-targeted brainstem by an average of 4.03- and 3.74-fold compared with intranasal (IN) administration alone in the non-tumor mice and glioma mice, respectively. Immunohistochemistry staining found that aPD-L1 was mainly located within the perivascular spaces after IN delivery, while FUSIN further enhanced the penetration depth and delivery efficiency of aPD-L1 to the brain parenchyma. The delivered aPD-L1 was found to be colocalized with the tumor cells after FUSIN delivery to the brainstem glioma. These findings suggest that FUSIN is a promising technique to enhance the delivery of immune checkpoint inhibitors to gliomas.

2:00

1pBAb5. Predicting spontaneous preterm birth risk is improved when quantitative ultrasound data are included with prior clinical data. Barbara L. McFarlin (Human Development Nursing Sci., Univ. of Illinois at Chicago, Chicago, IL), Shashi Roshan (Dept. of Statistics, Univ. of Illinois at Urbana-Champaign, Champaign, IL), Aiguo Han (Univ. of Illinois at Urbana-Champaign, Urbana, IL), Douglas G. Simpson (Dept. of Statistics, Univ. of Illinois at Urbana-Champaign, Champaign, IL), and William D. O'Brien (Elec. and Comput. Eng., Univ. of Illinois, 306 North Wright St., Urbana, IL 61801, wdo@uiuc.edu)

Predicting women at risk for spontaneous preterm birth (sPTB) has been medically challenging due to lack of signs and symptoms of preterm labor until intervention is too late. Hypothesis: advanced statistical modeling predicts sPTB risk from quantitative ultrasound (QUS) plus prior data (prior birth history through first clinical cervical length) better than that of prior data alone. Study population included 250 full-term births (FTBs) and 25 sPTBs. QUS scans (Siemens S2000 & MC9-4) were performed using a standard cervical length approach by registered diagnostic medical sonographers. Two cervical QUS scans were conducted at 20 ± 2 and 24 ± 2 wk gestation. Multiple QUS features were processed from calibrated raw radiofrequency backscattered ultrasonic signals. Two statistical models designed to determine sPTB risk were compared: (1) QUS plus prior data and (2) prior data alone. Test ROC AUC compared both models. Using statistical methods, QUS plus prior data identified women at risk for sPTB with better AUC (0.68; std error 0.01; 95% CI, 0.66–0.70) than that of prior data alone (0.63; std error 0.01; 95% CI, 0.61–0.65). Even with only 25 sPTBs, data suggest that there is value added for predicting sPTB when QUS data are included with prior data. [R01HD089935.]

2:15–2:30 Break

2:30

1pBAb6. Reversible increase in the EKG of rats using transthoracic pulsed ultrasound. Olivia Coiado (Carle Illinois College of Medicine, 1914 Max Run Dr., Champaign, IL 61822-3449, oliviacoiado@hotmail.com), Rahul Yerrabelli, Alex Lucas, Anton Christensen (Carle Illinois College of Medicine, Champaign, IL), Marcin Wozniak, and William D. O'Brien (UIUC, Urbana, IL)

This study showed the ability to increase inotropy in a rat heart while maintaining hemodynamic stability. A total of fourteen rats were used in this experiment. They were divided into four group including a control group (no ultrasound exposure). 3-month-old female rats were transthoracically to 3.5 MHz ultrasonic pulses of 2.0-MPa peak rarefactional pressure amplitude, $\sim$0.5% duty cycle and an increase pulse repetition frequency (PRF) sequence (4–5–6, 5–6–7, and 6–7–8 Hz). All rats underwent the same anesthesia, preparation, measurement, and positioning with only experimental rats being exposed to the turned-on transthoracic ultrasound. Rats were positioned in dorsal recruitment position and cardiac parameters were monitored using EKG and echocardiography. An increase in the heart rate was observed followed by each PRF sequence with increased heart rates. A mean of $550 + -50$, 563 ± 45, and 712 ± 61 bpm at the conclusion of ultrasound for 4–5–6 Hz, 5–6–7 Hz, and 6–7–8 Hz sequences versus the control baseline of 248 ± 19 bpm was achieved. Other cardiac parameters were normal or had a compensatory decrease by 3- and 15-min post-ultrasound compared to control. EKG data of the experimental group of rats showed areas of rapid peaks outside of sinus rhythm during ultrasound pulsing while the control group-maintained sinus rhythm throughout the experiment.

2:45

1pBAb7. Validation of the spatial resolution of super-resolution ultrasound imaging using vessel mimicking microfluidic channels. Qiyang Chen (Univ. of Pittsburgh, 3550 Terrace St., 624, Pittsburgh, PA 15261, qic41@pitt.edu) and Kang Kim (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Super-resolution ultrasound (SRU) imaging is an emerging technology that can identify microvessels with unprecedented spatial resolution beyond the acoustic diffraction limit. Our group introduced a deconvolution-based SRU imaging technology and successfully applied it on mouse and human kidneys for assessing the renal microvascular changes during the progress of kidney diseases. However, a thorough validation on the most achievable spatial resolution has not yet been systematically performed due to the limitation with *in vivo* imaging including imperfect co-registration between SRU image and ground truth such as histology. In this study, we evaluated the spatial resolution of our SRU imaging algorithm using a custom-designed PDMS microfluidic channels as the ground truth. In the reconstructed SRU image, the two microchannels separated by wall-to-wall distance of 30 μm were successfully identified, which assured the expected spatial resolution of SRU using 15 MHz transducer. The validation using custom-designed microfluidic channels as the ground truth in the scale of tens of microns was performed for the first time to the best of our knowledge for SRU imaging. Such channels of various diameters and separation distances can be an effective tool for the systematic evaluation of the imaging capabilities of different SRU algorithms.

3:00

1pBAb8. Acoustic patterning of endothelial cells *in vivo* to produce microvascular networks. Diane Dalecki (Biomedical Eng., Univ. of Rochester, 210 Goergen Hall, P.O. Box 270168, Rochester, NY 14610, dalecki@bme.rochester.edu), Eric S. Comeau, Melinda A. Vander Horst, Carol H. Raeman (Biomedical Eng., Univ. of Rochester, Rochester, NY), and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

Previous work from our lab developed an ultrasound standing wave field (USWF) technology to noninvasively pattern endothelial cells within collagen hydrogels to fabricate vascularized engineered tissue constructs. Here, we translate this technology towards acoustic patterning and microvascular network formation *in vivo*. To enable acoustic patterning *in vivo*, we developed a two-transducer ultrasound system with intersecting beams to produce an USWF at the intersection of the crossed beams. Acoustic radiation forces associated with the USWF patterned cells into planar bands located at nodal planes. Distance between planar bands was controlled through design of acoustic frequency and angle between ultrasound beams. To demonstrate acoustic patterning *in vivo*, solutions of collagen and human umbilical vein endothelial cells (HUVEC) were injected subcutaneously into flanks of mice and then exposed to an USWF. High-frequency ultrasound imaging was employed to visualize initial acoustic patterning in tissues. Ten days after USWF-mediated cell patterning, tissues were excised, and vessels derived from USWF-patterned cells were identified using an antibody specific to human von Willebrand factor (hVWF). Microvessels expressing hVWF were present in USWF-exposed tissues and not in sham-exposed controls. Furthermore, blood vessels expressing hVWF contained red blood cells, indicating anastomosis of ultrasound-fabricated microvessels with host vasculature and tissue perfusion.

3:15

1pBAb9. Evaluation of biological proxies for surgical instruments and their accuracy for acoustic decontamination techniques. Craig N. Dolder (Sloan Water Technol., Sloan Water Technol., 1 Venture Rd., Southampton, Hampshire SO16 7NP, United Kingdom, dolder@utexas.edu), Maryam Malakoutikhah (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), Thomas J. Secker (School of Biological Sci., Univ. of Southampton, Southampton, Hampshire, United Kingdom), and Timothy G. Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

Surgical instruments go through rigorous protocols involving decontamination and sterilization. Of particular concern is Variant Creutzfeldt-Jakob Disease, because (unlike viruses and bacteria) the prions responsible for this are not destroyed by the high temperatures, cleaning chemistries, and radiation used. This paper is part of a study on the effectiveness of ultrasonically-activated streams to remove dangerous prions from surgical instruments to prevent infections from spreading between patients. Cleaning techniques are often evaluated against standard surgical instrument substitutes in a test such as animal bioassays and amplification assays. These test uses thin wires (0.016 mm diameter), which are pre-contaminated, put through the cleaning process, and then sent for analysis to determine how much contamination remains. Despite being an industry standard with proven effectiveness as a proxy for cleaning surgical instruments using heat/chemicals/radiation, thin wires are shown to be a poor proxy when ultrasonically cleaned. The acoustics of thin wires and surgical instruments differ significantly and therefore the performance of acoustic cleaning methods can fail this test while being ultimately effective on surgical instruments. We present measurements on and in the vicinity of thin wires and steel plates to illustrate the differences between these two targets and discuss the reasons for this disparity.

3:30

1pBAb10. Holographic lenses for the spatial patterning of particles toward tissue engineering *in vivo*. Mohamed A. Ghanem (Appl. Phys. Lab., Univ. of Washington, 1013 NE Boat St., Seattle, WA 98105, mghanem@uw.edu), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), Diane Dalecki (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY), and Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

For tissue engineering, cells are aligned along defined structures that allow for cellular agglomeration and regeneration. Acoustic radiation force has been used for the contactless generation of patterns. Particles that are much smaller than the wavelength are agglomerated along high pressure regions and the forces are predicted using Gor'kov potential. Ultrasound standing waves have been used *in vitro* to create parallel planes where the cells agglomerate allowing for vascular tissue growth between planes. Here, we design and build a single-sided source with a holographic lens made using Fourier spectrum approach to generate parallel lines of high pressure amplitude over a limited range, few wavelengths, for cellular patterning. A few sources were fabricated at 1.5 and 2 MHz using geometrical and holographic lenses. The phase generation from holographic lenses was improved using global phase unwrapping scheme. Acoustic holography measurement was performed for all sources to verify the pressure output. The cellular patterning was tested on polyethylene microspheres (75-90 μm in diameter) that were placed in a cuvette with water and exposed to ultrasound. The microspheres were aligned along the high pressure region as predicted. This work has the potential to use single-sided transducers for cellular patterning toward tissue engineering *in vivo*.

Session 1pED

Education in Acoustics: General Topics in Education in Acoustics

Daniel A. Russell, Chair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802

Contributed Papers

1:00

1pED1. A step by step educational guide to designing and testing an underwater acoustic modem. Rami Rashid (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., 323 Dr Martin Luther King Jr. Blvd, Newark, NJ 07102, raa62@njit.edu), John DellaMorte, James DellaMorte (delResearch, LLC, Sandwich, MA), Erjian Zhang, and Ali Abdi (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ)

We examine the design and systematic implementation of a receiver system for underwater acoustic communication. The system includes, but is not limited to, passband filter, down converter, matched filter, and detection logic. Our final implementation target is a frame-based system to minimize memory use and system latency. The matched filter uses the overlap-save method in frequency domain to reduce computational cost. Initially, the system is designed and tested for performance using the MATLAB platform in a non-real time vector-based implementation. The code is then restructured to a MATLAB frame-based model. The code contains several functions; each function performs a specific task and maintains its own histories as needed. Once the performance of the frame-based model is measured to be identical to the vector-based model, we systematically move each function to C++. Each C++ function is individually substituted into the frame-based model using MEX files and verified for performance. Once all C++ functions are verified for performance, the entire C++ implementation is uploaded to the Popoto Modem™ real-time hardware and performs real-time transmission and reception in both air and underwater environments, with known and tested performance characteristics. [Work supported in part by NSF, Grant IIP-1500123, the INTERN program.]

1:15

1pED2. Design of virtual reality for teaching ultrasound in medical and engineering education. Jan Lee (Carle Illinois College of Medicine, Champaign, IL) and Olivia Coiado (Carle Illinois College of Medicine, 1914 Max Run Dr., Champaign, IL 61822-3449, oliviacoiado@hotmail.com)

This works presents the design of a virtual reality (VR) platform that generates 3D models from ultrasound imaging used to teach medical and engineering students about anatomical structures. The VR platform will allow students to learn about the theory of wave propagation in tissues by providing 3D visualizations of abstract physics concepts and organ systems in an immersive and engaging environment. This phase of the project has been focused on learning to develop VR environments and how to translate medical imaging formats into a feasible tool. The specific focus has been on converting DICOM (Digital Imaging and Communications in Medicine) files into elements that can be managed in the VR environment (Unity 3D Development). This includes converting imaging data sets into 3D meshes that can be displayed and manipulated. The nearly infinite control over virtual environments can present alternative visualizations of ultrasound that may be accessible to the learner. The use of acquired ultrasound data within the virtual space will help learners transfer skills into point-of-care ultrasound (POCUS). The VR environment will help students to explore ultrasound without the limitations of a real patient and guide them to the physics principles to better understand ultrasound propagation in the human body.

1:30

1pED3. Cultural policy infrastructure and the representation of Nigeria's music in education. Andrea Calilhanna (MARCS Inst. for Brain, Behaviour and Development, Western Sydney Univ., Sydney, New South Wales 2126, Australia, A.Calilhanna@westernsydney.edu.au), Adebowale Adeogun (Music, Univ. of Nigeria, Nsukka, Nigeria), and Stephen G. Onwubiko (Brigham Young Univ., Enugu, Nigeria)

Serious systemic and unethical consequences of notation-based music theory, which focuses on tonality, include its contribution to excluding the discourse of most of the world's music traditions because most of them are not notated. New projects seeking inclusion require the analysis of the meter for accurately documenting and discoursing music. The implications of the results of this study will likely involve significant music education policy and cultural infrastructure policy adjustments and change (Calilhanna, 2017, 2018, 2019, 2020, 2021; Calilhanna *et al*., 2019). Some of these changes will involve key competencies; other policies will require cultural heritage acknowledgement, further consideration to research-based evidence for transferable skills, and literacy and numeracy. A scientific approach to analysing similarities and differences of music traditions delves into finer details with suitable analytical instruments and processes. Yet most of the cultural contributions, the musical heritage of people worldwide, for example, the traditional music of post-colonial countries such as the Igbo in Nigeria, has been silenced, is at risk of extinction, remains unaccounted for or is inaccurately explained. Cohn's (2020) psychoacoustic approach to the meter is discussed through visualising the beat-class in a three-step approach with the ski-hill, circle, and linear, cyclic graphs.

1:45

1pED4. Teaching speech acoustics concepts in an undergraduate speech science course: Are distributed or massed practice lab assignments more effective? Kathleen Siren (Speech-Language-Hearing Sci., Loyola Univ. Maryland, 4501 N. Charles St., Baltimore, MD 21210, ksiren@loyola.edu)

Educational research holds that individuals learn concepts more effectively when practice applying these concepts is distributed rather than massed. However, few studies have investigated distributed versus massed practice for student learning of a set of related concepts. This study investigated student learning and retention of speech acoustics concepts in an undergraduate speech science course. In one class design, students completed several short speech acoustics labs spaced throughout the semester to practice applying learned concepts as the semester progressed. In another class design, students completed one extensive speech acoustics lab at the end of the semester to apply concepts learned throughout the semester. Students in all courses completed a comprehensive examination at the end of the semester. An analysis of summative assessment data indicates that distributed practice may be more effective in facilitating student learning and recall of individual concepts. However, massed practice may promote broader conceptual learning of related concepts, possibly by facilitating discovery of connections across concepts.

Session 1pNS

Noise, Architectural Acoustics, ASA Committee on Standards, and Structural Acoustics and Vibration: Noise Issues in Modern Construction

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Chair's Introduction—1:20

Invited Papers

1:25

1pNS1. Innovative solutions for acoustic disturbances occurring in slender buildings. Patrick Murray (SoundSense, New York, NY), Bonnie Schnitta (SoundSense, 34 Industrial Rd., Unit 6, Wainscott, NY 11975, bonnie@soundsense.com), Collin G. Champagne, and Jeremy R. Newman (SoundSense, Morristown, NJ)

Many tall, slender residential building have risen from the streets of Manhattan and other cities over the past decade. Numerous problems have been recently published about acoustic disturbances present in these slender buildings due to deflection, principally from wind. These disturbances occur, not due to any structural weakness of the building, but due to excessive rigidity and lack of resiliency of the interior architectural elements, such as wall studs, tracks and gypsum board. These components must be designed and installed for resiliency in order to respond to the slender building movement, which is essential in order to ensure the building's structural integrity. This movement consists of deflection and torquing of the slabs relative to each other. These acoustic disturbances cause occupants to lose sleep and cause them to feel unsafe, as though structural failures are occurring. Additionally, interior finishes become warped and lose aesthetic quality. These issues are detrimental to the value of units within these slender buildings. This paper presents case studies of innovative methodologies to solve these acoustic disturbances present in slender buildings. These include methodologies which ensure resiliency of the interior architectural elements. These case study examples have been successfully implemented during the base-unit design and also retrofitted after construction.

1:45

1pNS2. Acoustic isolation of cabinets on walls with resilient channels. Michael Raley (PAC Int., 2000 4th Ave., Canby, OR 97013, mraley@pac-intl.com)

Single-stud walls (wood or steel) with resilient channel are commonly used to provide acoustic isolation for apartments and hotel rooms. In apartments, kitchen cabinets are often mounted on demising walls and/or corridor walls where resilient channel is also being used. In hotels, heavy items such as headboards, nightstands, and desks are often mounted to the demising walls. It is often impossible to entirely avoid mounting items on the side of the wall with resilient channels. Commonly, contractors will attach plywood directly to the studs between the resilient channel to fill the void created by the resilient channel, and the cabinets or other heavy items are screwed through the gypsum board and into the plywood, completely short-circuiting the isolation provided by the resilient channel. This presentation provides laboratory test results showing the detrimental effects of this mounting method. The presentation also introduces a new method of mounting cabinets (and other heavy items) that maintains the acoustic isolation of the resilient channel. Laboratory test results for the new mounting method are included and they show that the overall acoustic performance (STC rating) of the wall is maintained when cabinets are mounted.

2:05

1pNS3. Use of LIIC and HIIC metrics to evaluate the efficacy of ceiling isolators. Michael Raley (PAC Int., 2000 4th Ave., Canby, OR 97013, mraley@pac-intl.com)

The LIIC and HIIC dual-rating method has been developed to better evaluate the impact noise isolation of floor-ceiling assemblies. LoVerde *et al.* have argued that impact noise is characterized by two distinct frequency domains that can vary independently based on changes to an assembly. The LIIC rating addresses low-frequency impact noise and is calculated using the impact sound pressure levels in the 50, 63, and 80Hz 1/3-octave bands. The HIIC rating addresses high-frequency impact noise and is calculated using the impact sound pressure levels in the 400 to 3150 Hz 1/3-octave bands, inclusive. This presentation looks at the ability of the dual-rating method to compare the efficacy of a range of ceiling isolators from generic resilient channel to spring isolators using a set of laboratory tests

conducted at the same laboratory on the same base floor/ceiling assembly. The results show that the dual-rating method can provide useful information for design decisions even when the test data is from a lab with a receiving room volume that is less-than-ideal for measurements at the low frequencies required for the LIIC metric.

2:25

1pNS4. Better than concrete. Light-weight fitness floating floor solutions for reducing impacts. Adam P. Wells (None, 1177 Ave. of the Americas, Fl. 7, New York, NY 10036, adam.prescott.wells@icloud.com)

Structure-borne noise and vibration from weight drops and fitness activities are an ever-increasing source of disturbance within multi-residential, hospitality, and multi-tenant commercial buildings. High-mass floating floors supported on low-resonant frequency isolators have been used with good effect to reduce impact noise and vibration, but commonly require expensive supplemental supports or structural stiffening to be implemented properly. Common light-touch fitness solutions feature continuous or interlocking resilient surface treatment. This approach is often ineffectual at high-energy impact fitness areas and the pursuit of soft surface solutions to achieve greater dissipation of impact energy can create an unstable surface for weightlifting. An alternate approach to reducing structure-borne noise exists which focuses on dissipating the impact energy through constrained layer damping in conjunction with low-resonant frequency isolators. Research and development testing shows that an engineered all-dry lightweight fitness floating floor is capable of achieving equal or greater impact insulation performance than a high-mass concrete floating floor and significantly better performance than light-touch resilient surface treatment. A review of the testing and development process and underlying principles will be reviewed to illuminate the mechanisms at work in the next generation of fitness floating floors.

Session 1pPA

Physical Acoustics, Biomedical Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Acoustofluidics II

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Max Denis, Cochair
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Chair's Introduction—1:20

Contributed Papers

1:25

1pPA1. Physical and acoustic response of *Pinus palustris* to fire. Kara M. Yedinak (Bldg. and Fire Sci., U.S. Forest Service, Forest Products Lab., 1 Gifford Pinchot Dr., Madison, WI 53726, kara.yedinak@usda.gov), J. K. Hiers (Tall Timbers Res. Station, Tallahassee, FL), Ian Grob (Missoula Technol. and Development Ctr., Missoula, MT), James P. Riser (Wisconsin Dept. of Natural Resources, Madison, WI), and Scott Pokswinski (Tall Timbers Res. Station, Tallahassee, FL)

Crackling sounds from burning vegetation provide information to fire-experienced land managers regarding the combustion process. These acoustic impulse events can inform practitioners of a variety of situations, from the type of vegetation involved and relative "dryness," to alerting wildland fire fighters to increased burning along tree boles and in canopies. More recently, acoustic sampling has supported these observations by quantifying species and drought-stress specific differences in the acoustics impulse events in both the laboratory and field. Our team builds on this work by quantifying the response of long leaf pine (*Pinus palustris*) in grass-stage to low-intensity fire. We planted live fuel beds in Northern Florida with long leaf pine seedlings. We then used prepared dead long leaf pine needles to carry low-intensity fire past the live plants. Crackling sounds from both the live and dead needles were recorded. These acoustic impulse events were then analyzed to identify any unique characters associated with the live and dead needles. We pair these results with single leaf-cluster scale laboratory experiments in which we documented the mechanical and acoustic response along the axial extent of the leaves. We present the results of long leaf pine's acoustic and mechanical response to low-intensity fire.

1:40

1pPA2. Microstreaming induced by two interacting acoustic bubbles. Claude Inserra (LabTAU, 151 Cours Albert Thomas, Lyon 69008, France, claude.inserra@inserm.fr), Gabriel Regnault (Ecole Centrale Lyon, Lyon, France), Cyril Mauger, Alexander Doinikov (INSA Lyon, Lyon, France), and Philippe Blanc-Benon (CNRS, Ecully, France)

Microbubbles submitted to a sufficiently strong ultrasound field can show nonspherical oscillations called surface modes. These oscillations induce a slow mean flow named microstreaming in the vicinity of the bubble interface. We present experimental and theoretical investigations of the microstreaming induced by a bubble pair as an elementary modelling of a dense bubble cloud. Two bubbles are trapped in a dual-frequency acoustic chamber that allows studying their interface dynamics. The secondary radiation force they experience and make them attract under increasing acoustic forcing is quantified through the relative location of the bubble within the acoustic field. Each bubble can exhibit several nonspherical modes that are either axisymmetric or not. The typical flow signatures induced by these nonspherical modes are identified. When the bubble shape oscillations are axisymmetric, the extension of microstreaming pattern does not allow fluid interaction in the region between the two bubbles. In case of asymmetric oscillations, large-scale microstreaming is obtained. These experimental findings are discussed by means of a theoretical modelling of the fluid flow induced by two nonspherically-oscillating microbubbles.

1:55

1pPA3. Towards 3D selective manipulation of cells and microparticles with spiraling holographic interdigitated transducers. Michael Baudoin (IEMN, Université de Lille, Ave. Poincaré, Villeneuve d'Ascq 59652, France, michael.baudoin@univ-lille.fr) and Zhixiong Gong (IEMN, CNRS, Lille, France)

In previous papers [Baudoin *et al.*, *Sci. Adv.* **5**, eaav1967 (2019); Baudoin *et al.*, *Nat. Commun.* **11**, 4244 (2020)], our team demonstrated the possibility to trap and manipulate microparticles and cells selectively, in 2D and in a standard microfluidic environment with some spiraling interdigitated transducers (S-IDTs). These S-IDTs were designed as the phase hologram of specific wave structures called focused acoustical vortices. Here we explore theoretically with an angular spectrum code the 3D trapping capabilities of these transducers. We demonstrate: (i) that the field synthesized by S-IDTs creates a 3D radiation trap for microparticles and cells and (ii) that the trap can be moved axially by simply tuning the driving frequency. This work opens perspectives for 3D manipulation of cells with single beams acoustical tweezers.

2:10–2:25 Break

2:25

1pPA4. A numerical model for acoustophoretic separation based on elastic moduli differences of cells/microparticles. Alara Karaman (Mech. Eng. Dept., Middle East Tech. Univ., Ankara, Turkey), Hande Nur Açıkgöz, Barbaros Çetin (Mech. Eng. Dept., Bilkent Univ., Ankara, Turkey), and Mehmet Bülent Özer (Mech. Eng. Dept., Middle East Tech. Univ., Orta Doğu Teknik Üniversitesi, Makina Mühendisliği Bölümü, Ankara 06800, Turkey, ozerb@metu.edu.tr)

Acoustophoresis is known to be a promising technique in cell and particle separation. A more comprehensive study that includes the elastic model of the human cell is needed for more realistic simulations of acoustophoretic cell manipulation and separation. Here, we implemented a finite-element based approach that consists of the simple elastic modeling of a human cell to calculate the acoustic radiation force acting on the cells. Using this numerical model, new separation modes based on the difference in elastic properties of particles and cells are found and simulated. The proposed numerical simulation does not utilize the widely used Gor'kov's analytical acoustophoretic force formulation since the proposed new separation mode is observed at frequencies where acoustic wavelengths are comparable to cell dimensions. Unlike most studies in the literature, the proposed numerical model accounts for acoustic and hydrodynamic particle-particle interactions for the simulation of the cell trajectories. We believe this proposed simulation method and the proposed mode of acoustophoretic separation will be beneficial for separating similar-sized cells/particles with different elastic moduli using acoustophoresis.

2:40

1pPA5. Investigation of the effect of chip material on acoustofluidic microparticle manipulation. Hande Nur Açıkgöz (Mech. Eng. Dept., Bilkent Univ., Ankara, Turkey), Alara Karaman (Mech. Eng. Dept., Middle East Tech. Univ., Ankara, Turkey), Mehmet Akif Şahin (Mech. Eng. Dept., Bilkent Univ., Ankara, Turkey), Ömer Çaylan, Göknur Cambaz Büke (Dept. of Mater. Sci. and Nanotechnology Eng., TOBB Univ. of Economics and Technol., Ankara, Turkey), Ender Yıldırım (Mech. Eng. Dept., Middle East Tech. Univ., Ankara, Turkey), Barbaros Çetin (Mech. Eng. Dept., Bilkent Univ., Ankara, Turkey), and Mehmet Bülent Özer (Mech. Eng. Dept., Middle East Tech. Univ., Orta Doğu Teknik Üniversitesi, Makina Mühendisliği Bölümü, Ankara, Ankara 06800, Turkey, ozerb@metu.edu.tr)

Ultrasonic particle manipulation in microchannels is a rapidly growing research field with significant potential applications in biological studies. Several factors are essential in achieving effective manipulation, such as obtaining resonance conditions in the actuator, chip, and/or the microchannel. The choice of chip material is an important design parameter since it has significant implications on acoustic performance, thermal performance, stability, ease of manufacturing and the overall cost of the acoustofluidic separation unit. Even though different materials are employed in acoustofluidic studies, most studies implement silicon as the chip material due to its superior acoustic properties. On the other hand, silicon has a complicated manufacturing procedure that requires cleanroom facilities. Therefore, from the manufacturing perspective, it is beneficial to use materials that enable mechanical machining. In this study, for the first time in the literature, we have compared the performance of acoustofluidic chips manufactured from silicon, glass, PMMA (polymethyl methacrylate) and PDMS (polydimethylsiloxane) materials in a comparison study. Furthermore, we propose using a new composite material, FR4, which seems to possess favorable acoustic properties of acoustically hard materials (such as silicon and glass) and the ease of manufacturing of acoustically soft materials such as polymers.

2:55

1pPA6. On sound dispersion and attenuation in simple and multi-fluids due to thermal and viscous effects. Azeddine Zaidni (MSDA, Université Mohammed VI Polytechnique, Ksar Hart, Goulmima, Errachidia 52250, Morocco, azeddine.zaidni@um6p.ma) and Saad Benjelloun (MSDA, Université Mohammed VI Polytechnique, Casablanca, Morocco)

We present a new formula for dispersion relation for sound waves in viscous and heat conducting (mono)fluids. In particular, this dispersion (i.e. variation of speed of sound with frequency) is shown to be of second order of magnitude, with respect to Knudsen numbers, as in the Stokes case, corresponding to non-conductive fluid (Infinite Prandtl number). This formula completes the classical attenuation relation called Stokes–Kirchhoff. In the meanwhile, we represent in a simplified manner the Kirchhoff approach to derive this attenuation, starting from the 3D compressible Navier-stokes system and emphasis the hypothesis that are essential to its validity. In a second part of the presentation we deal with the case of mixture of fluids (multi-fluids) and present how viscosity and thermal conductivity have effects on the dispersion and attenuation of sound. In addition, we show how the approach may be used to discriminate models in multi-phase modeling that do not reproduce physically meaningful sound propagation speeds.

3:10

1pPA7. Using bulk acoustic waves for filtering microplastic on polluted water. Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), Berliana N. Sari, Laila S. Urrohma, Dian Permana, and Arkilaus B. Felle (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Microplastic is known as one of the polluted objects on seawater which can contaminate marine biota. In this report, we designed microplastic filters using Bulk acoustic waves. Two speakers were added to the design to create acoustic waves, and three channels were created on the tube where the polluted water flows. The force produced by the wave can separate the microplastic from the polluted water by creating node pressure on the tube, so it presses microplastic particles towards the center of the channel, whereas the clean water on the other two channels. Using this method, the filter can filter all microplastics and have an efficiency of up to 99% for nylon and 95% microplastics.

Session 1pPP

Psychological and Physiological Acoustics, Education in Acoustics, Biomedical Acoustics, and Speech Communication: Acoustics Outreach to Student Scientists in Clinical and Physiological Research

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Chair's Introduction—1:30

Invited Papers

1:35

1pPP1. Testing the plausibility of harmonic templates based on temporal fine-structure. Yi-Hsuan LI (KU Leuven, Campus Gasthuisberg, O&N 2 Herestraat 49 - bus 102, Leuven 3000, Belgium, yihsuan.li@kuleuven.be) and Philip X. Joris (KU Leuven, Leuven, Belgium)

The existence of harmonic templates in the central nervous system is a basic tenet of place models of pitch. Shamma & Klein (2000) propose that coincidences in temporal fine-structure across auditory nerve fibers in response to broadband stimuli suffice to set up harmonic templates for all fundamental frequencies in the phase-locking range. However, their computational model uses cochlear filters with unphysiologically steep slopes, calling into question the plausibility of the proposed scheme. We obtained responses from chinchilla auditory nerve fibers and "high-sync" neurons to pure tones, low-pass noise, and broadband noise, including "warped" noise (Heinz, 2006; Cedolin & Delgutte, 2010). Coincidences were computed using shuffled auto- or crosscorrelograms. We find that the maximum number of coincidences in response to tones of different frequencies, simulating the output of fibers with infinite sharp filters, indeed shows the template structure modeled by Shamma and Klein. This was not the case for responses to low-pass or broadband noise. However, if coincidences were only counted at zero delay, higher numbers of coincidences were indeed found for harmonically-related characteristic frequencies, even in response to noise. We conclude that steep-sloped cochlear filtering is not a necessary condition to obtain harmonic templates based on fine-structure.

1:55

1pPP2. Encoding of fundamental frequency and fine structure of speech sounds: A comparison between adults and newborns. Sonia Arenillas-Alcón (Brainlab - Cognit. Neurosci. Res. Group, Univ. of Barcelona, Gran Via de les Corts Catalanes, 585, Barcelona, Catalonia 08007, Spain, sonia.arenillas@ub.edu), Teresa Ribas-Prats, Carles Escera (Brainlab - Cognit. Neurosci. Res. Group, Univ. of Barcelona, Barcelona, Catalonia, Spain), M. Dolores Gómez-Roig (BCNatal - Barcelona Ctr. for Maternal Fetal and Neonatal Medicine, SJD Barcelona Children's Hospital, Barcelona, Catalonia, Spain), and Jordi Costa-Faidella (Brainlab - Cognit. Neurosci. Res. Group, Univ. of Barcelona, Barcelona, Catalonia, Spain)

Encoding changes of voice pitch and formant structure plays an important role in speech comprehension since birth. Thus, proper functional assessment in perinatal hospital settings is of essence, but time constraints render the evaluation unfeasible. Here we present a novel two-vowel, rising-pitch-ending stimulus (/oa/; 250 ms) to analyze pitch tracking and low-frequency formant encoding accuracy simultaneously in approximately 30s, extracting measurements from the Frequency-Following Response (FFR) auditory evoked potential recorded with clinical hardware developed for audiology assessments. Thirty-four healthy newborns (mean gestational age = 40 week) and 18 normal-hearing adults were tested. Results showed significant differences across groups in neural lag and F0 spectral amplitude, but not in F0 signal-to-noise ratio, stimulus-to-response correlation or pitch measures, indicating that newborns and adults are able to track F0 changes accurately. FFR's fine-structure analyses revealed that adults encode formant structure of vowels significantly better than newborns, with largest differences at high-frequency ranges. Here, we demonstrate the feasibility to assess speech envelope and fine-structure encoding within clinical evaluation times in a hospital setting. Our results indicate that voice pitch encoding is adult-like at birth, while formant representation is maturing at birth, suggesting the possibility to use this new stimulus as a measure of auditory system development in newborns.

2:15

1pPP3. Top-down attention modulates neural responses in neurotypical, but not ADHD, young adults. Jasmine Kwasa (Neurosci. Inst., Carnegie Mellon Univ., 5000 Forbes Ave., Apt. 22, Pittsburgh, PA 15217, jkwasa@andrew.cmu.edu), Abigail Noyce, and Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA)

Individuals differ in their ability to selectively attend to goal-relevant auditory stimuli. People with Attention-Deficit/Hyperactivity Disorder (ADHD) in particular tend to show cognitive deficits associated with inhibition and attention. We hypothesized that people with ADHD would exhibit poorer performance and weaker neural signatures of attention when undertaking a challenging auditory task that required strong top-down attentional control. Neurotypical ($N = 20$) and ADHD ($N = 25$) young adults with normal hearing listened to one of three concurrent, spatially separated speech streams and reported the order of the syllables presented while we recorded electroencephalography (EEG). We tested both the ability to sustain attentional focus on a single "target" stream and the ability to monitor the target but flexibly switch attention to an unpredictable "interrupter" stream from another direction if and when it appeared. In both groups, the interrupter evoked larger neural responses when it was to be attended compared to when it was irrelevant, including for the P3a "reorienting" response previously described as involuntary. This attentional modulation was weaker in ADHD listeners, even though their behavioral performance was no lower. Additionally, individual performance correlated with the degree of top-down modulation of neural responses. These results demonstrate that listeners differ in their ability to modulate neural representations of sound based on task goals.

Contributed Paper

2:35

1pPP4. Acoustic bandwidth effects on envelope following responses to simulated bimodal hearing. Can Xu (Speech, Lang., and Hearing Sci., Univ. of Texas at Austin, Speech, Lang., and Hearing Sci., 2504A Whitis Ave. (A1100), Austin, TX 78712-0114, canxu@utexas.edu), Fan-Yin Cheng, Sarah Medina, and Spencer Smith (Speech, Lang., and Hearing Sci., Univ. of Texas at Austin, Austin, TX)

Bimodal hearing, which combines a cochlear implant (CI) with a contralateral hearing aid, provides significant speech recognition benefits in quiet and noise. These benefits have also been observed in normal-hearing listeners using vocoder-based CI simulation combined with low-pass filtered speech, even with acoustic bandwidths as narrow as 125–250 Hz. However, it is challenging to measure the optimal acoustic amplification with difficult-to-test populations, such as young children and adults with disabilities. The frequency following response (FFR) offers a potential solution to this problem, as it objectively quantifies subcortical phase-locking to speech features. Recently, FFR fundamental frequency amplitude in the non-implanted ear was found to be well-correlated with bimodal benefit in CI patients. The present study aimed to parametrically examine acoustic bandwidth effects (125, 250, 500, and 750 Hz) on speech evoked FFRs using simulated bimodal stimuli. We hypothesized that FFR_{env} amplitudes would increase as bandwidth increases up to 750 Hz and the minimal acoustic bandwidth needed to derive FFR bimodal benefit is less than 250 Hz.

Invited Papers

2:50

1pPP5. Sentence recognition in individuals with history of multiple concussions while listening to accented speech in noise. Madison Buntrock (Hearing and Speech, Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, mbuntroc@terpmail.umd.edu)

It has been well established that listening to spoken language is a complex process that becomes even more difficult in noise, a common experience in everyday life. Previous research suggests that speech perception may be affected by the type of background noise. There has been less research with the initial aim to evaluate the role of task load and susceptibility to distraction affecting the difficulty of speech processing. Furthermore, recent work suggests a protracted recovery from a concussion. However, there is much needed to research the nature of these persisting deficits, specifically regarding language. Research suggests there are language deficits following a concussion, but the full extent is unknown. We evaluated the differences between a group with a history of concussion and a control group. Using a sit-down transcription task, participants were asked to type what they heard the talker say. The female talker was either speaking in a native accent or a foreign accent. These spoken sentences were either presented in quiet or in a multi-talker babble. We evaluated accuracy and response time. Our hypotheses were twofold, there would be a significant difference in reaction time between groups and interaction between background noise and accent of the talker.

3:10

1pPP6. A machine-learning approach to studying categorical versus continuous perception. Sara D. Beach (Div. of Medical Sci., Harvard Univ., 43 Vassar St., 46-4033, Cambridge, MA 02139, beach@g.harvard.edu)

Robust and efficient speech perception relies on the interpretation of acoustically-variable phoneme realizations. Some neuroimaging evidence supports a categorization account, in which speech-evoked patterns of brain activity are transformed into categorical representations. Other evidence supports a continuous account, in which subphonemic detail is maintained over time. However, it is not well understood if or how these patterns of brain activity undergo a continuous-to-categorical transformation, nor how task demands may modulate this process. My approach involves applying biologically- and psychologically-plausible linear classifiers to neural activity as measured by magnetoencephalography (MEG). Data came from adult participants who heard isolated, randomized tokens from a /ba/-/da/ speech continuum. In the passive task, their attention was directed elsewhere. In the active task, they categorized each token as "ba" or "da." I find that classifiers successfully decode "ba" versus "da" perception from the MEG data. But do they perform like humans, with excellent sensitivity to between-category differences and poor sensitivity to within-category differences, or do they respect stimulus identity? And do passive versus active task demands affect these patterns of classifier performance? Given the wealth of information in the MEG signal, we can test these hypotheses over time and space, as perception unfolds across the brain.

Contributed Paper

3:30

1pPP7. A naturalistic approach to studying temporal processing during music performance. Elise Piazza (Univ. of Rochester, 10 Meliora Hall, Rochester, NY 14534, epiazza@ur.rochester.edu), Riesa Cassano (Univ. of Rochester, Rochester, NY), Marius Cătălin Iordan, Jamal Williams (Princeton Univ., Princeton, NJ), Sarah Izen (Univ. of Rochester, Rochester, NY), and Uri Hasson (Princeton Univ., Princeton, NJ)

In recent years, the push to embrace naturalistic stimuli over artificial designs has enriched what we know about the neural underpinnings of human attention, memory, and communication in real life. Previous work using natural stories scrambled (at the word, sentence, and paragraph level) has revealed a hierarchy of brain regions that organize natural acoustic input at these different timescales. While this approach has advanced our understanding of language processing, many fewer studies to date have explored the neural underpinnings of music perception, let alone music production, in naturalistic settings. In our novel paradigm, we asked expert pianists to play musical pieces, scrambled at different timescales (measure, phrase, section) on a non-ferromagnetic piano keyboard inside the fMRI scanner. This dataset provides unprecedented access to expert musicians' brains starting from their first exposure to a novel piece and continuing over the course of learning to play it. We found distinct patterns of tuning to musical timescales across several clusters of brain regions (e.g., sensory/motor, parietal, and frontal/memory). We also found that musical predictability impacts functional connectivity between auditory, motor, and higher-order regions during performance. Finally, we applied several machine learning analyses to understand how the brain dynamically represents acoustic and musical features.

MONDAY AFTERNOON, 29 NOVEMBER 2021 501 (L)/504 (O), 1:15 P.M. TO 3:45 P.M.

Session 1pSA

Structural Acoustics and Vibration, Engineering Acoustics, Signal Processing in Acoustics, Computational Acoustics, and Physical Acoustics: Novel Techniques for Nondestructive Evaluation in Structural Acoustics and Vibration II

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

Tyler J. Flynn, Cochair
JHU/APL, 11100 Johns Hopkins Rd., Laurel, MD 20723-6099

Invited Papers

1:15

1pSA1. Imaging solids using full waveform inversion. Sezgin Kucukcoban, Heedong Goh (Civil, Architectural and Environ. Eng., The Univ. of Texas at Austin, Austin, TX), and Loukas Kallivokas (Civil, Architectural and Environ. Eng., The Univ. of Texas at Austin, 301 East Dean Keeton St., Stop C1748, Austin, TX 78759, loukas@mail.utexas.edu)

Wave-driven approaches have long been dominant in the nondestructive evaluation of both engineered and natural systems. In this presentation, we discuss recent progress in the full-waveform inversion (FWI) approach for the material characterization and condition assessment of solids probed by elastic waves. In FWI, characterization and localization of defects is automatically revealed once the distribution of the spatially-varying material properties is completed. In this work, we seek to image the Lamé parameters of arbitrarily heterogeneous solids, when probed by elastic waves in the time domain. Accordingly, we use the apparatus of PDE-constrained optimization and seek the Lamé parameter distributions that minimize the misfit between measured and computed responses, subject to the governing PDEs. As is commonly the case, the resulting inverted profiles of the second Lamé parameter (μ) are of better quality than those of the first (λ). To improve the resolution of both Lamé parameters, we discuss the use of three robustifying schemes, namely, source-frequency continuation, regularization factor continuation, and a search direction-biasing scheme. We demonstrate with numerical experiments the effect the schemes have on the inversion process and conclude with an application of the robustified full-waveform method to a challenging adaptation of the benchmark Marmousi2 model.

1:35

1pSA2. Ultrasonic inspection of lithium-ion batteries to detect cell safety and prevent thermal runaway. Tyler M. McGee (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Barrett J. Neath (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Samuel B. Matthews, Erik J. Archibald, Ofodike A. Ezekoye (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

Accurately monitoring the safety of lithium-ion batteries to prevent a thermal runaway (TR) event is of utmost importance especially in high-power applications such as electric vehicles. The fire caused by a lithium-ion battery is particularly difficult to extinguish because of its continuous emittance of highly flammable gas and its extremely high temperature. Current battery management systems (BMS) only monitor cell safety by measuring voltage, current, and temperature on the module level. Because the BMS operates on the module-level as opposed to the cell-level, these metrics can fail to predict a TR event, which may begin with individual cell failures. We report on the viability of early detection of cell damage by simultaneously monitoring voltage, current, temperature, mechanical clamping force, and ultrasonic wave transmission through individual battery cells under various electrical, mechanical, and thermal loading conditions. Time- and frequency-domain features of the acoustic signals transmitted along three propagation paths and excitation frequencies between 0.1 and 1 MHz are monitored and correlated with electrical, temperature, and force sensor data under mechanically clamped overcharge and thermal abuse scenarios. Early indicators of damage in the ultrasonic signals are presented and discussed with respect to propagation paths and known damage mechanisms in lithium-ion batteries.

1:55

1pSA3. Nondestructive evaluation of adhesively bonded plate-like structures through local wavenumber estimation. Jakub Spytek (Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland), Lukasz J. Pieczonka (Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Al. A. Mickiewicza 30, Krakow 30059, Poland, lukasz.pieczonka@agh.edu.pl), and Lukasz Ambrozinski (Robotics and Mechatronics, AGH Univ. of Sci. and Technol., Krakow, Poland)

Nondestructive testing (NDT) of adhesively bonded plate-like structures using traditional ultrasonic (US) testing modalities can be challenging and time-consuming. Faster inspection can be achieved with the use of full-field techniques based on guided waves. In particular, the local wavenumber estimation (LWE) of Lamb waves has been proven effective for damage detection in various plate-like structures. However, the effectiveness of LWE depends on multiple parameters, such as the excitation frequency, the choice of a Lamb wave mode, or the inspection side, among others. This is especially significant in the case of multilayer plates, where different types of damage result in a nontrivial behavior of Lamb waves. In this work, we present a framework for the evaluation of disbonds in adhesively bonded multilayer plates through local wavenumber estimation. We demonstrate theoretically and experimentally that the proposed methodology allows obtaining clear images of defects and determining their through-thickness locations in multilayer plate-like structures.

2:15–2:30 Break

Contributed Papers

2:30

1pSA4. Passive structural health monitoring using an acoustic digital twin. Simone Sternini (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), Alexis Bottero (Scripps Inst. of Oceanogr., UC San Diego, Marine Physical Lab. 8820 Shellback Way, La Jolla, CA, abottero@ucsd.edu), Jit Sarkar, and William Kuperman (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

This work presents simulation results on passive structural health monitoring of a vibrating elastic structure for defect localization. We consider an object that consists of a shell with vibration sensors surrounding a complex, vibrating structure such that they are coupled at a finite number of points. The non-defective model of the external shell is called an acoustic digital twin (ADT). We use a combination of the concept of an acoustic digital twin (ADT) together with an adjoint-based high resolution array processing approach to detect a defect. The "dataset" in the presence of a defect was obtained from simulations and was "measured" on a linear array placed on the surface of a candidate ADT. A set of sensors placed at the connecting points provides the time histories of the excitation forces, which are used as inputs to the ADT in order to obtain the fields at the surface in the unperturbed case (no defect). The residual field (difference between defect and non-defect case) was used in the Matched Field Processing (MFP) algorithm in order to successfully locate the defect on the surface of the object.

2:45

1pSA5. Nonlinear ultrasonic testing for evaluation of fracture toughness in 4130 steel. Colin Williams (Eng. Sci. and Mech., The Penn State Univ., 212 EES Bldg., University Park, PA 16802, clw5756@psu.edu), Cody Borigo (Guidedwave, Bellefonte, PA), Jacques Riviere, Cliff Lissenden, and Parisa Shokouhi (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

An estimation of plane strain fracture toughness [endif]–> is essential to assess the operational safety of fracture-critical systems; loss of [endif]–>can lead to catastrophic failure. However, it is not possible to quantify [endif]–>*in situ*; the laboratory testing to determine [endif]–>is destructive. Here, we investigate nonlinear ultrasonic testing as a nondestructive method to evaluate [endif]–>. This is based on a hypothesized correlation between the nonlinear ultrasonic parameters and [endif]–>characteristics of a material given their mutual dependence on microstructure. Using Second Harmonic Generation (SHG), surface and bulk waves are used to measure the (relative) nonlinearity parameter for tempered 4130 steel samples. Coupons made of the same materials are also tested for their [endif]–>characteristics using Charpy V-Notch testing, providing a direct comparison between destructive and nondestructive measures. Our results show correlations between ultrasonic parameters and [endif]–>characteristics of 4130 steel samples, but different relations are observed for bulk and surface waves. The observed differences are possibly due to sample heterogeneity and different wave structures of the two wave modes and should be further investigated. We demonstrate the potential of nonlinear ultrasonic testing for [endif]–>estimation. The ultimate goal is developing a nondestructive method to assess [endif]–> *in situ*.

3:00

1pSA6. High frequency ultrasonic scattering from cracks in orthotropic silicon wafers. Lauren Katch (The Penn State Univ., 212 Earth and Eng. Sci. Bldg., State College, PA 16802, Luk50@psu.edu) and Andrea Arguelles (The Penn State Univ., State College, PA)

Silicon wafers, commonly used in the photovoltaic industry, are susceptible to cracking during manufacturing and processing that may impact their performance. These microscale cracks are difficult to detect ultrasonically due to the anisotropic nature of silicon and the high frequencies necessary for inspection. For instance, phased array (PA) approaches commonly used to detect and size cracks are unsuitable for this application due to the frequency constraints of commercial PA systems. Therefore, crack scattering in anisotropic media has mostly been studied from a theoretical perspective and little work has been done from an experimental lens. In this presentation, we experimentally evaluate ultrasonic scattering from cracks in sub-millimeter silicon wafers using single-element 100 MHz probes. We use an oblique-incidence, pulse-echo configuration to spatially map shear scattering from cracks at different orientations in orthotropic silicon samples. Our results illustrate the strong scattering dependence on crystal orientation and elucidate the complexities in the response as a function of the planes of symmetry. These results will enable more in-depth sensitivity analysis for crack detection and characterization in anisotropic samples.

3:15

1pSA7. Sound and vibration measurements for pipe wall condition assessment of concrete pressure pipes. Werner Richarz (ICONAC, 55 Administration Rd., Unit 37, Concord, ON L4K 4G9, Canada, werner@iconac.co), Harrison F. Richarz (ICONAC, Concord, ON, Canada), John Sun, Fatemeh Karimi, and Taiwo Ricketts (KenWave Solutions, Concord, ON, Canada)

The condition of buried watermains and similar pipelines can be assessed by non-intrusive means. To this end certain features of sound propagation in the water column within the main are measured. Most applications use the Korteweg-Lamb correction to estimate the average wall thickness for homogeneous pipes. For complex composite structures such as concrete pressure pipes, condition assessment is presently based on a great deal of empiricism. This can now be replaced with proper physical models based on measurements conducted on a 25 m long test pipe. Using controlled sound sources, hydrophones and surface mounted accelerometers, features of the acoustic waves inside the water column and the associated wall vibrations were deduced from two-point space-time correlations. The data permits the determination of the complex-valued wavenumbers as well as the dynamic coupling of the liquid column and the pipe wall from which certain properties of the pipe wall can be characterized to ascertain if the pipe is fit for purpose.

3:30

1pSA8. Discussion on harmonic scattering of A0 wave from a local nonlinear damage—A numerical study. Pravinkumar R. Ghodake (Mech. Eng., Indian Inst. of Technol., Bombay, IIT Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

In nonlinear ultrasonics, the interaction of waves with uniformly distributed damages in metallic plates is very well studied through different theoretical, computational, and experimental studies. But in practice, highly local damages such as micro-cracks, and local plasticity are introduced in plates. This motivates to get more detailed insight into the harmonic scattering of A0 waves from early-stage local damages. To investigate the harmonic scattering from local damage in plates, numerical studies (FEM) are carried out. Local damage is modeled as the Murnaghan model for aluminum at the center of the spatial domain by avoiding linear impedance mismatch between linear and nonlinear regions. Harmonic scattering due to monochromatic A0 wave input is captured successfully. Higher harmonics of backscattered and forward scattered displacements noted at the plate surface are observed. The separation of second harmonics from primary mode due to group velocity mismatch is not observed as the local damage sizes are relatively short (5i–20 mm). One-way and two-way two-wave mixing with group velocity mismatch studies shows sum, difference, and higher harmonics along with mode conversion of both the backscattered and forward scattered waves. Higher harmonics of the harmonically scattered waves are highly sensitive to damage size and intensity of damages.

Session 1pSC

Speech Communication: Speech Acoustics and Production (Poster Session)

Rajka Smiljanic, Chair
Univ. of Texas at Austin, 305 E. 23rd St., B5100, Austin, TX 78712

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. Voice quality differences in Dunan: Links between gemination and fortis-lenis contrasts. Raksit T. Lau-Preechathammarach (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720-2650, raksit@berkeley.edu)

The fortis-lenis contrast is traditionally defined by "articulatory force" and correlates with a host of measures, including closure duration, f0, amplitude, VOT, open quotient, and spectral tilt (Kohler & Dommelen, 1987; Cho *et al*., 2002). These measures are also features of voice quality (Keating *et al*., 2011; Brunelle *et al*., 2020) and gemination contrasts (Idemaru & Guion, 2008; Kraehenmann, 2011), and gemination has been suggested to be a type of fortis-lenis distinction (Ladd & Schmid, 2018). The current study explores acoustic correlates of initial /T K/ vs. /t k/, a contrast variably termed as initial gemination or fortis/lenis, in Dunan, a highly endangered Southern Ryukyuan language (Bentley, 2008; Yamada *et al*., 2015). Audio from fieldwork sessions with an elderly speaker were aligned with the Montreal Forced Aligner (McAuliffe *et al*., 2017) and measures of /T K t k/ and the following vowel were taken. The contrast is cued not only by closure duration and VOT, but also by f0, amplitude, H1*-H2*, H1*-A3*, and CPP differences that persist over the following vowel. This naturalistic data contributes to evidence for a constellation of acoustic features shared between fortis-lenis, gemination, and voice quality contrasts that may point to commonalities in laryngeal configuration.

1pSC2. The temporal unfolding of register reveals distinct mechanisms at play: A case study of Western Khmer. Sireemas Maspong (Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, sm2627@cornell.edu)

Register distinctions are a type of suprasegmental contrasts manifested in several acoustic dimensions, e.g., pitch, vowel quality, and voice quality. However, several aspects of register production remain unclear (Brunelle & Kirby, 2016). In this paper, we focus on whether a target laryngeal configuration may be responsible for the multiplicity of acoustic cues to register. We approach this question through a production study of Western Khmer, an Austronesian language spoken in Chanthaburi, Thailand. Data were collected from 19 speakers producing words with modal or breathy register in a carrier sentence. The measurements we took are trajectories of F0, F1, H1*-H2*, and H1*-A1*. Our results show that all cues are significant in distinguishing the two registers, confirming static measurement results in the previous literature (Wayland, 2001). However, a dynamic study of these register cues reveals a dissociation in their temporal alignment. The F1 transitions that mark changes in vowel quality are not aligned with the transitions of H1*-H2* and H1*-A1* that mark changes in voice quality. These findings suggest that separate asynchronous mechanisms may be responsible for the voice quality and vowel quality components of register production.

1pSC3. Stop voicing contrast in Lio. Veronica Miatto (Linguist., Stony Brook Univ., Stony Brook, NY 11790, veronica.miatto@stonybrook.edu) and Grace B. Wivell (Linguist., Stony Brook Univ., Stony Brook, NY)

Typologically, two-way or three-way voicing contrasts in stops can be reliably defined along the VOT continuum, that is, by measuring the gap between the start of voicing and the stop burst, which can be negative or positive. In the literature, three main categories are therefore derived: prevoiced, unaspirated, and aspirated stops (Keating, 1984; Lisker and Abramson, 1964; and Cho *et al*., 2019). For Lio, an understudied Austronesian language of Central Flores (Eastern Indonesia), it has been claimed that the stop voicing contrast is realized with prevoicing for voiced stops and aspiration for voiceless stops (Elias, 2018). However, it is unclear how long the aspiration is, and whether Lio voiceless stops can be characterized as unaspirated or aspirated; furthermore, this claim was not based on an acoustic analysis, and relied heavily on one speaker. This study offers a descriptive analysis on voicing contrasts in Lio stops. The acoustic analysis reveals that the contrast is one of 'true voicing', with the closure phase of voiced stops presenting voicing (prevoiced) and short-lag VOT voiceless stops (unaspirated). This study will offer a valuable contribution to voicing contrasts typology, especially compared to Cho *et al*. (2019), and serves as a building block for future Lio research.

1pSC4. Acoustic features of Hadza clicks. Didier Demolin (Lab. of Phonet. and Phonol., CNRS-UMR 7018, Sorbonne Nouvelle, Paris, France, didier.demolin@sorbonne-nouvelle.fr), Andrew Harvey, Richard Griscom (Linguist., Universiteit Leiden, Leiden, The Netherlands), and Alain Ghio (LPL, Aix-Marseille Univ. & CNRS UMR7309, Aix-en-Provence, France)

Hadza (Khoesan) has several clicks in its sound inventory [1,2]. There are 4 types of clicks in the language: bilabial, dental, alveolar and lateral [⊙, ǀ, !, ǁ] that can be accompanied, by aspirated, glottal and nasal features [⊙̰, ǀˀ, ǀʰ, ᵑǀ, ᵑǀˀ, !ˀ, !ʰ, ᵑ!, ᵑ!ˀ, ǁ, ǁʰ, ᵑǁʰ, ᵑǁˀ]. Our study was made with 9 native speakers (4 women and 5 men) using various instrumental techniques (Acoustic, aerodynamic, EGG and Video). Clicks have two components: an attack and an extinction transient. Bilabial clicks have a noisy release, sometimes whisteled. Aspiration is short for dentals and alveolars, long for laterals. Glottals have a long lag after the burst release. The 4 types of clicks can be produced with the nasal feature. Acoustically, clicks are described with 2 features [grave versus acute] and [abrupt versus noisy] [3]. The [grave] feature has a peak in the FFT spectrum around 1200Hz and the [acute] around 3000 Hz. For some clicks, the cavity posterior to the constriction is excited. Short resonances occur between 1 and 2 kHz following the posterior release, they result from the lowering of the tongue dorsum and from the reduction in the volume of the pharyngeal cavity. [1] K. Miller, *Hadza Grammar Notes* (Ms Riezlern, 2008). [2] B. Sands, "Phonetics and phonology: Hadza," in *The Khoesan languages*, edited by R. Vössen (Routledge, London, 2013). [3] A. Traill, s"The perception of clicks in !xoo," *J. African Lang. Linguist.* **15**, 161–174 (1994).

1pSC5. Acoustic features of palatalized consonants in Nasa Yuwe. Elodie Vanzeveren (Lab. of Phonet. and Phonology, CNRS-UMR 7018, Sorbonne Nouvelle, Paris, France), Esteban Diaz, Tulio Rojas (Linguist., Univ. of Cauca Popayan, Popayan, Colombia), and Didier Demolin (Lab. of Phonet. and Phonology, CNRS-UMR 7018, Sorbonne Nouvelle, Paris, France, didier.demolin@sorbonne-nouvelle.fr)

Nasa Yuwe a Colombian language spoken in the Cauca presents complex features of palatalization [1,2]. Stops and affricates can be palatalized, aspirated and palatalized-aspirated at all places of articulation while prenasalized consonants and fricatives can only be palatalized. Data for this study were recorded from an acoustic recording session and by combining, in a second phase, synchronized acoustic, EGG and oral airflow recordings from 6 speakers (3 female and 3 male). Results show that all palatalized stops are characterized by a short frication noise at the end and a F3/F4 pole at the start of the following vowel. Aspirated stops have a greater frication noise duration and an open glottis. Palatalized-aspirated stops and affricates are produced with frication noise, a F3/F4 pole at the start of the following vowel and an open glottis. Palatalized-aspirated consonants have two phases: a short and intense frication noise accounting for the palatalization, followed by a lower intensity aspiration noise. Both phases can be distinguished by their FFT spectra and oral airflow volume velocity. Prenasalized consonants behave as palatalized stops but without the aspiration feature. Palatalized fricatives have a resonance during the frication, reflecting a tongue movement towards the palatal place. [1] E. Dias, *El Habla Nasa (paez) de Munchique: Nuevos Acercamientos a Su Sociolingüistica, Fonologia y Morfosintaxis* (Université Lumière Lyon 2, 2019). [2] T. Rojas, *La Lengua Paez* (Ministerio de Cultura, Bogota, 1998).

1pSC6. Effects of uvular consonants on vowel quality in Lushootseed. Ted K. Kye (Dept. of Linguist., Univ. of Washington, Guggenheim Hall, 3940-2425 Benton Ln., Seattle, WA 98195, tkye29@uw.edu)

Although the influence of uvular consonants on vowel quality has been studied in Interior Salish languages (Bessell, 1992; 1998; Flemming *et al.*, 2008), it has not been studied in Coast Salish languages. This study investigates the effect of uvular consonants on vowel quality in Lushootseed, a Coast Salish language spoken in the Puget Sound region of the Pacific Northwest, which has uvular stops and fricatives. Archival recordings (Burke Museum, 2015) of a native speaker were examined. A study was performed to analyze the effects of uvular consonants on formants of adjacent vowels. F1 and F2 were measured for vowels adjacent to uvular consonants that were positioned pre- and post-vocalically. These measures were compared with vowels adjacent to non-uvular consonants. The results reveal that there was an increase in F1 and a decrease in F2 for all vowels. The effects were greater for F1 than F2. These results are in contrast with those of Bessell (1992; 1998), who observed two effects: (i) only non-low vowels were affected, and (ii) F2 showed a greater effect than F1. These findings provide further implications on acoustic phonetic research for Coast Salish languages.

1pSC7. Nomlaki vowel quality: An acoustical and archival examination. Anna Björklund (Linguist., UC Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, aebjorklund@berkeley.edu)

This study examines vowel quality and duration in Nomlaki, a Wintuan language of Northern California that survives via archival recordings. Phonemic vowel length is reconstructed in Proto-Wintuan (Shepherd, 2005) and is inherited in Nomlaki's sister languages (Pitkin, 1984; Lawyer, 2015), producing a vowel system of five long/short pairs. Despite this, current orthography used in Nomlaki tribal language revitalization distinguishes Nomlaki vowel pairs via tense/lax quality. This study uses data from archival recordings to document how F2 primarily distinguishes these historically long/short vowel pairs. Data consisted of 1574 tokens spoken by two male speakers recorded in 1953 and 1975. Each token contained F1, F2, F3, and duration information; 234 tokens were labelled according to the official tribal spelling when spelling was known. A Linear Discriminant Analysis (LDA) model was fit with 5-fold cross-validation on the labelled set, after which each of the other tokens was assigned a vowel label per the LDA model. The model was 64% accurate. 86.7% of the variance was accounted for by the first dimension, 8.2% by the second, and 4.8% by the third. These dimensions were primarily weighted by F2, F1, and duration, respectively. The author has no known conflicts of interest.

1pSC8. Speaking rate and language-specific voice onset time effects on burst amplitude: Cross-linguistic observations and implications for sound change. Chandan Narayan (York Univ., 4700 Keele St., Toronto, ON M3J1P3, Canada, chandann@yorku.ca)

The relationship between speaking rate and burst amplitude in voiceless plosives was investigated in two languages with differing oro-laryngeal timing implementations of phonological voicing, North American English and Indian Tamil. Burst amplitude (reflecting both intraoral pressure and flow geometry of the oral channel) was hypothesized to decrease in CV syllables with increasing speaking rate, which imposes temporal constraints on both intraoral pressure buildup behind the oral occlusion as well as respiratory air flow. Increased speaking rate led to decreased burst amplitude (relative to vowel amplitude) in both languages, with the magnitude of the effect being considerably weaker in Tamil, which has short-lag implementation of voice onset time. Bilabials in both languages were affected disproportionately relative to other places of articulation. Additionally, burst amplitudes were lower overall in Tamil, reflecting lower intraoral pressure which promotes faster vocalic onset. Results are discussed in terms of language-internal and extra-linguistic phonetic phenomena potentially serving as perceptual triggers for historical sound change.

1pSC9. Vowel duration in the Buckeye corpus. Sean A. Fulop (Linguist., California State Univ. Fresno, 5245 N Backer Ave., Linguist. PB92, Fresno, CA 93740-0001, sfulop@csufresno.edu) and Hannah J. Scott (Linguist., California State Univ. Fresno, Corvallis, OR)

Phonological accounts of American English [W. Labov *et al.*, *The Atlas of North American English* (Walter de Gruyter, Berlin, 2006)] rely on historical and phonotactic analysis to categorize the vowels as either long (two morae) or short (one mora). The vowels of *beat, bait, boot, boat, bought, bite, bout, boy* are traditionally identified as long, while *bit, bet, bat, book, but, bot* are supposedly short. It is a reasonable question whether these vowels are indeed longer and shorter, respectively, in phonetic duration. A study of "General American" [H. A. Rositzke, "Vowel-length in general American speech," *Language* **15**(2), 99–109 (1939)] examined the durations of the vowels of five subjects in scripted utterances, without controlling for dialect. The findings were mostly in accord with the above categories, with the exceptions that [æ, ɑ] *(bat, bot)* were both phonetically long. The present study is an effort to replicate Rositzke's on 341 425 vowel tokens in the Buckeye Corpus of conversational English (40 speakers from Columbus, Ohio). The results essentially confirm that the vowels [æ, ɑ] *(bat, bot)* are phonetically as long as the traditional "long" vowels. This situation calls the traditional analysis of American English vowel length into question.

1pSC10. Beyond midpoints: Vowel dynamics of the low-back-merger shift. Joey Stanley (Linguist., Brigham Young Univ., 4064 JFSB, Provo, UT 84602, joey_stanley@byu.edu)

The Low-Back-Merger Shift—the lowering/retraction of /æ/, /ɛ/, and /ɪ/—is a now-well-studied phenomenon of American English. However, it is primarily described by the relative *position* of the vowels using single-point acoustic measurements. This study seeks to fill an apparent gap in the literature by describing change in these vowels' formant *trajectories*. Data come from 45 h of sociolinguistic interviews conducted in southwest Washington (ages 18–86). A generalized additive mixed-effect model (Wood, 2007) fit to pairs of formant measurements extracted at 11 timepoints per vowel token confirms that the lax vowels are shifting and uncovers changes in their formant dynamics in apparent time. For instance, the bulk of /æ/ lowering and retraction is a result of its *onset* shifting rather than the *offset*. For all three vowels, the amount of change in F2 over the vowel's duration decreased over time, resulting in qualitative changes in the shape of the vowel trajectory (when visualized in the F1-F2 space). Furthermore, though the men lagged behind the women in relative *position* in the vowel space, both genders changed the *shape* of their trajectories at the same time. This study shows that there is more to vowel shifts than changes in their midpoints.

1pSC11. Sample size matters when calculating Pillai scores. Joey Stanley (Linguist., Brigham Young Univ., 4064 JFSB, Provo, UT 84602, joey_stanley@byu.edu) and Betsy Sneller (Linguist., Lang., and Cultures, Michigan State Univ., East Lansing, MI)

Pillai scores, an output statistic from a MANOVA model, are used to measure acoustic overlap between two merging, splitting, or shifting vowels (Hay *et al*., 2006; Nycz & Hall-Lew, 2013, inter alia). They range from 0 (complete overlap) to 1 (complete separation). In this paper, we demonstrate that sample size matters in two important ways when calculating Pillai. First, relatively small datasets (<60 per sample) tend to underreport mergers. In simulated data, we draw two samples (5–100 tokens in each) from the same normal distribution to represent two vowels known to be merged. We find that a sample size of at least 60 tokens is required for at least 95% of the iterations to return a Pillai score less than the commonly used threshold of 0.1. s, Pillai scores are sensitive to the difference in sample sizes across two samples, with large differences resulting in Pillai scores that overreport mergers for samples drawn from two distinct distributions. If Pillai scores are used to measure vowel overlap, we recommend (1) using the same number of tokens across both samples and (2) using p-values from the MANOVA model to accompany them to assess whether the difference between two distributions is significant.

1pSC12. Modal coupling of acoustic wave propagation and mechanical wall vibration in the time-varying vocal tract. Gordon Ramsay (Dept. of Pediatrics, Emory Univ. School of Medicine, Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, Georgia, gordon.ramsay@emory.edu)

Coupling between acoustic wave propagation and mechanical vibration of wall tissue along the vocal tract has long been considered to be an important loss mechanism in speech, affecting the frequencies and bandwidths of the lower formants. However, the detailed mechanism of energy transfer between fluid and structure as the shape of the vocal tract changes over time has not been fully elucidated. To examine this question, a time-domain finite-volume simulation of quasi-one-dimensional fluid flow and mechanical wall vibration was derived from underlying conservation laws. By calculating the eigenvalues and left and right eigenfunctions of the resulting implicit matrix recursion at each point in time, the sound field can be decomposed into modal vibrations describing coupled motion of air and tissue. Standing waves in the fluid within the vocal tract create sound waves that propagate in the wall tissue. Conversely, standing waves in the wall tissue excite waves in the fluid that propagate as sound. The coupling between airborne and mechanical modes of vibration and the transfer of energy between the two systems can easily be understood by examining the time-varying amplitude and phase of the spatial eigenfunctions. Illustrations are provided for simulations of VCV transitions.

1pSC13. Effects of vowel devoicing on the closure duration of japanese singleton and geminate stops. Kimiko Yamakawa (Faculty of Contemporary Culture, Shokei Univ., 2-6-78 Kuhonji, Chuo-ku, Kumamoto 8628878, Japan, jin@shokei-gakuen.ac.jp) and Shigeaki Amano (Faculty of Human Informatics, Aichi Shukutoku Univ., Nagakute, Aichi, Japan)

Closure duration is an acoustic variable known to distinguish Japanese singleton and geminate stops. It has been established that the closure duration varies with speaking rate. However, it has not been determined whether the closure duration is affected by devoicing of vowels preceding or following the stops. To determine the effects of such vowel devoicing, a production experiment was conducted with 20 Japanese native speakers. They pronounced four minimal pairs of onomatopoeic words contrasting singleton and geminate stops at various speaking rates. Devoiced vowels were found to affect closure duration preceding the stops, such that the closure with a devoiced preceding vowel is longer than that with a voiced preceding vowel, although the effect is weak for singleton stops. However, devoiced vowels do not affect the duration when they follow stops. These results suggest that vowel devoicing interacts with an acoustic variable of stop consonants, especially in the case of geminate stop.

1pSC14. Coarticulation across communicative contexts: An acoustic analysis of the LUCID corpus using spectral and temporal measures. Zhe-chen Guo (Linguist., The Univ. of Texas at Austin, 307 E 31st St. Apt. 105, Austin, TX 78705, y9024131@gmail.com) and Rajka Smiljanic (Univ. of Texas at Austin, Austin, TX)

Coarticulation reflects a balance between talker efficiency and speech comprehension. This study examines coarticulation across different communicative contexts in the LUCID corpus. The talkers completed an interactive spot-the-difference task with a listener when there were no communication barriers (NB), when their speech was vocoded (VOC), when it was masked by multi-talker babble (BABBLE), and when the listener was an English language learner (L2). They also read sentences without an interlocutor conversationally (READ-CO) and clearly as if talking to someone with hearing loss (READ-CL). The spectral distance and the relative duration of coarticulatory transition between the adjacent segments were computed. Both measures showed that speech produced in the presence of a communication barrier (BABBLE, VOC, L2, and READ-CL), be it real or imagined, is more resistant to coarticulation than speech produced in the barrier-free conditions (NB and READ-CO). Spectral distance was more sensitive to differences among all conditions involving a real listener (BABBLE, VOC, L2, and NB) while relative transition duration measure better captured differences between read and spontaneous speech for the barrier-present conditions (BABBLE, VOC, L2, and READ-CL). The results suggest that talkers reduce coarticulation in response to the particular communication challenges which may aid speech perception.

1pSC15. Phonetic convergence in temporal organization during shadowed speech. Roberto Gonzalez (Linguist., The Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, gonzalez.594@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

The goal of this study was to examine phonetic convergence in various measures of temporal organization during shadowed speech across different American English dialects. Participants from the Northern and Midland American English dialect regions, as well as "Mobile" participants who had lived in multiple region, read 72 sentences to establish a baseline for temporal organization, and then repeated the same 72 sentences after Northern, Midland, and Southern model talkers. Measures of temporal organization (i.e., %V, ΔC, ΔV, rPVI-C, and nPVI-V) were calculated for the read sentences, shadowed sentences, and model talker sentences. Statistical analysis of the differences in distance between the model talker sentences and the shadowers' read and shadowed sentences, respectively, revealed significant convergence by all three shadowing groups toward the model dialects for ΔV, and significant divergence by Mobile talkers away from the model talkers for nPVI-V. Though the result of divergence by Mobile talkers was unexpected, both results provide evidence that support previous studies, which claim that social perception is a large contributing factor in convergence and divergence. These results are also consistent with previous findings demonstrating variation across dialects in temporal organization and in addition, provide evidence for variation across dialects in convergence in temporal organization.

1pSC16. Acoustic cues to non-native-directed speech. Kevin Lilley (Linguist., The Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, lilley.35@osu.edu), Rebecca Scarborough (Linguist., Univ. of Colorado, Boulder, Boulder, CO), and Georgia Zellou (Linguist., Univ. of California, Davis, Davis, CA)

Facing linguistic barriers to communication, talkers spontaneously adjust their speech, facilitating perception by their listener. These adjustments characteristically involve hyperarticulation, such as reduced speech rate, increased vowel dispersion, and increased segmental duration. Listener-based hyperarticulation could result from listener modeling, in which the talker monitors their listener's comprehension. Hyperarticulation could also result from rich phonological representations, in which listener-based variation is directly represented. This study examined predictions of monitoring and representational accounts of listener-based hyperarticulation using a perception task. Participants completed a forced-choice identification task which tasked them with identifying the background of an interlocutor from recorded speech. That speech was originally produced to imagined and real and to native and non-native interlocutors. If listener-based variation originates with listener modeling, accuracy should be high and similar for each listener background. If that variation originates from rich representations, accuracy should be low, variable, and different across listener backgrounds. The results revealed poor accuracy, even for speech directed toward imagined non-natives, the most hyperarticulate speech. Participants' choices were influenced most by speech rate, second by vowel dispersion, but not by vowel duration. We discuss the implications of these results for cognitive models of listener-based phonetic variation.

1pSC17. Gender and speech fluency in Mandarin and English news interviews. Zhihan Hu (Shanghai Univ., No. 99 Shangda Rd., Baoshan District, Shanghai 200444, China, hzh1011@shu.edu.cn) and Cong Zhang (Radboud Univ., Nijmegen, The Netherlands)

Speech fluency affects the effectiveness of speech communications. Previous research has been mostly looking at the fluency performances in L2 and in atypical population. This paper investigates the fluency pattern in news anchor's natural interview speech, which represents the upper limit of native language fluency, and examines the effect of gender on fluency. We collected a gender-balanced corpus of audio clips of publicly accessible news programs by six native Mandarin speaking news anchors and six native American English news anchors. The total length of interview used in the analysis was 20 min for each language. This study examined six temporal and non-temporal measurements of fluency: temporal measures included speech rate, mean length of run (MLR), mean length of utterance (MLU), and mean length of pauses (MLP); non-temporal disfluency measures included syllable count and disfluency rate. The results showed that male Mandarin speakers had significantly lower MLR, lower MLU, and higher disfluency rates than female Mandarin speakers; no significant gender difference was found in MLP. The English data did not show any gender difference in any measurement. The results indicated that female Mandarin speakers spoke more fluently than male Mandarin speakers, while male English speakers and female speakers were equally fluent.

1pSC18. Acoustic correlates of gender identity in American English speech. Chen Zhou (Linguist. Program, City Univ. of New York - Graduate Ctr., 365 Fifth Ave., New York, NY 10016, czhou1@gradcenter.cuny.edu) and Jason Bishop (Linguist., City Univ. of New York, New York, NY)

A recent study with native German speakers finds that acoustic properties of speech associated with biological sex may also vary within-sex as a function of gender identity, defined gradiently along continuous scales of self-reported masculinity/femininity [M. Weirich and A. P. Simpson, "Gender identity is indexed and perceived in speech," *PLoS One* 13(12), e0209226]. In particular, Weirich and Simpson find that men who rate themselves as more feminine produce speech with somewhat more female-like characteristics, such as higher mean f0 and larger vowel space area (VSA) relative to men who rate themselves as less feminine. In the present study, we investigate whether such gradient measures of gender identity have a comparable relationship to acoustics in American English. Recordings of read speech produced by 74 native speakers from metropolitan New York City (38 females, 36 males) were analyzed with respect to VSA and spectral properties of the sibilant fricative /s/. Gradiently defined gender identity was, as in Weirich and Simpson (2018), estimated for all speakers using two established measures of self-reported femininity. We report findings related to VSA and /s/ for both male and female speakers, and discuss the implications for the role of sex and gender in phonetics research.

1pSC19. Variation in back vowel fronting and diphthongization in Louisiana English. Irina Shport (Dept. of English, Louisiana State Univ., 260 Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu)

Fronting of /u o ʊ/ vowel classes is prominent in Southern American English (SAE). This study was focused on Louisiana, a southern state where not all SAE features have had a strong hold. Dynamic formant trajectories of the target vowels were analyzed because different vowel fronting mechanisms were proposed for SAE and mainstream American English (Koops, 2010), and because formant trajectories of back vowels are rarely examined (Stanley and Renwick, 2020). Thirteen White and five Black college-aged females produced 24 monosyllabic words with vowels of each class as a part of a larger study. In White speakers, mean normalized F2 values at midpoints of /u o ʊ/ were not different from each other or from the central vowel /ʌ/, used as a reference. In Black speakers, /ʊ/'s F2 was similar to /ʌ/, but /u o/'s F2 was lower than in /ʌ/. In all speakers, formant trajectories of /u o/ were back-gliding, while /ʊ/ was center-gliding. Trajectory lengths of /u o/ were longer in White than in Black speakers. These results suggest that /u/ fronting was mainstream-like (nucleus fronting and a backglide) and that /u o/ fronting was more advanced in White than in Black female speakers.

1pSC20. Effect of cumulative distance in motor space on segment-sized slips in 3-word phrases. Melissa A. Redford (Linguist., Univ. of Oregon, Eugene, OR, redford@uoregon.edu), Carissa A. Diantoro, and Maya Davis (Linguist., Univ. of Oregon, Eugene, OR)

Slips, or speech errors (e.g., "*teep a cape*" for "*keep a tape*"), have been described as the disordered arrangement of discrete elements, giving rise to the hypothesis that these arise from misordering during phonological encoding. An alternative hypothesis is that slips are articulatory timing errors, conditioned by the same motor factors that underlie coarticulation. This hypothesis motivates the present study, which used a modified slip procedure to elicit errors from 40 native English-speaking adults. Participants spoke 3-word phrases that were constructed as mini tongue twisters, with alternating word onsets and repeating rhymes. These phrases were designed to condition slips to a greater or less extent according to coarticulatory-adjacent processes (e.g., palatalization). A metric of cumulative distance-traveled in motor space during phrase production was also calculated: the degree of change in vocal tract shapes across the phrase, estimated using data generated within the tubesounds model in ArtiSynth. Phrases designed to more strongly condition slips were expected to cover less cumulative distance in motor space than phrases designed to less strongly condition slips. The distance-traveled metric did not always align with this expectation, but was nonetheless a better predictor of presence/absence of slips than phrase condition in analyses on preliminary data.

1pSC21. Acoustic cues to features for Italian vowels as a function of lexical stress and tonal prominence. Ian Chan (Harvard Univ., 10 Colonel George McLaren Dr., Markham, ON L6C 0L3, Canada, ianchan@college.harvard.edu), Alec DeCaprio, Javier Arango (Harvard Univ., Cambridge, MA), Luca De Nardis (Sapienza Univ. of Rome, Rome, Italy), Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA), and Maria-Gabriella Di Benedetto (Sapienza Univ. of Rome and Massachusetts Inst. of Technol., Rome, Italy)

This study investigated how acoustic cues to features in Italian vowels are affected by lexical stress and tonal prominence. One aspect of an analysis on the LaMIT (Lexical Access Model for Italian) project, this study analyzed the impact of prosodic environment on acoustic cues to features. The LaMIT corpus consists of 100 unique sentences uttered twice by four native Italian speakers [Di Benedetto *et al*., "Speech recognition of spoken Italian based on detection of landmarks and other acoustic cues to distinctive features," in 179th ASA Meeting (2020)]. This preliminary experiment focused on vowel /a/. Duration, F0, height, and frontness (measured by F1 and F2 formant frequencies) were estimated for 1098 /a/ instantiations. Every sentence was also annotated for tonal prominence with TOBI (DeCaprio *et al*., "Using TOBI labels to document patterns of prosodic prominence and their acoustic effects in the LaMIT database," in 181st ASA Meeting (2021)]. Preliminary results show that lexical stress and tonal prominence of /a/ generally have a measurable impact on the vowel's acoustic realization. Specifically, F1 values of stressed /a/ are greater than those of unstressed /a/; likewise, F1 values of tonally prominent /a/ (always stressed) are greater than those of their stressed but non-prominent counterparts.

1pSC22. Pitch accents and boundary tones in narrow focus statements in Basque Spanish. Nerea Delgado Fernandez (Modern Lang. and Linguist., Florida State Univ., 625 University Way, Tallahassee, FL 32304, nd16@my.fsu.edu), Carolina González, and Lara Reglero (Modern Lang. and Linguist., Florida State Univ., Tallahassee, FL)

This presentation examines the intonation of non-contrastive narrow focus in 15 Spanish-Basque bilinguals from the Basque Country, Spain. A total of 150 semi-spontaneous statements with sentence-final new-information focus were analyzed in Praat following Spanish ToBi conventions (Aguilar *et al*., 2009). We report the pitch contour of prenuclear and nuclear accents and the prevalence of intermediate phrase boundaries (ip) before the focal element, none of which have been previously investigated in this Spanish variety. Results show that while the first prenuclear accent is generally delayed regardless of bilingualism (93%), the second one is delayed more often in Basque-dominant speakers. Rising-falling nuclear contours (L + H* L%), reported for broad focus in Basque Spanish (Robles Puente, 2012) and for narrow focus in other Spanish dialects (Aguilar *et al*., 2009), were most common in Spanish-dominant participants, while Basque-dominant participants tended to show falling contours (L* L%). Pre-focal ip boundaries occurred in 69% of sentences, regardless of bilingualism degree. Ip boundaries were typically rising, mostly involving continuation rises, although high plateaus were also attested. We will connect these novel findings to the intonational characteristics of narrow focus in Bizkaian Basque (Elordieta & Hualde, 2014) and other Peninsular Spanish dialects (Face, 2000; 2008).

1pSC23. Influence of clear speech on F_0 patterns of Mandarin lexical tones. Jack W. Rittenberry (Louisiana State Univ., 5261 Highland Rd. #375, Baton Rouge, LA 70808, jritt18@lsu.edu) and Irina Shport (Louisiana State Univ., Baton Rouge, LA)

When communication becomes difficult, a talker may take on a form of communication known as clear speech (Keerstock & Smilijanic, 2019; Picheny *et al*., 1985). Little is known, however, about clear-speech modifications in tone languages (Zeng, 2017). This study showed how native Mandarin speakers modulate F_0 in tone words through increased variation, raised F_0 onset and offset. Participants read twenty-four sentences varied in four target tones, three times. Sentences were presented in two blocks: casual speech condition (as if talking to friends) and clear speech condition (as if communicating to someone with hearing loss). Data analysis of seven speakers suggested that effects of clear speech on F_0 contours were tone-dependent. As compared with casual speech, in clear speech, F_0 contours of high level (T1), mid-rising (T2), and high-falling (T4) tones were higher within the same tonal context; in contrast, F_0 contours of the low-dipping tone (T3) were lower preceding T1 and T4. F_0 range, onset F_0, and offset F_0 were greater in clear speech mode across most tonal environments. Additional data of eleven speakers has been collected to examine preliminary results and discuss findings in terms of signal-enhancement in clear speech production of lexical tones.

1pSC24. Boundary-induced prosodic strengthening and its function in interactive and read speech. Seulgi Shin (Linguist., Univ. of Kansas, Temple Rd., Flat 1, Templar House, Oxford Ox4 2HR, United Kingdom, seulgi.shin@gmail.com) and Annie Tremblay (Linguist., Univ. of Kansas, Lawrence, KS)

This study investigates the functions of boundary-induced prosodic strengthening relative to speech style, focusing on English plosives and nasals in Intonational-Phrase(IP)-initial and IP-medial positions in interactive and read speech. Two accounts seek to explain the function of prosodic strengthening: *syntagmatic contrast enhancement* (SCE: enhancement of the contrast between neighboring consonants and vowels) (e.g., Cho & Keating, 2001) and *paradigmatic contrast enhancement* (PCE: enhancement of phonological contrasts) (e.g., Georgeton & Fougeron, 2014). For interactive speech, pairs of participants produced words while exchanging information about scenes; for read speech, each participant individually produced words in written sentences. For plosives, Voice-Onset-Time measurements showed SCE in read speech, with plosives being more consonant-like (less voiced) IP-initially than IP-medially, but PCE in interactive speech, with voiced and voiceless plosives being more distinct from each other and from other places of articulation IP-initially than IP-medially; spectral measures showed PCE in both interactive and read speech. These results suggest that speakers make plosives phonologically more distinct from one another in interactive speech to help listeners identify them better. Nasals were more consonant-like (less nasality) IP-initially than IP-medially in both interactive and read speech, showing SCE, attributable to nasals being phonetically and phonologically distinct from non-nasal consonants.

1pSC25. Prosody and the interpretation of English relative clauses. Elyssa Fenton (Modern Lang. & Linguist., Florida State Univ., 625 University Way, PO Box 3061540, Tallahassee, FL 32306, efenton@fsu.edu)

This study investigates the prosodic cues involved in the disambiguation of English relative clauses (RCs). For example, in the sentence "The reporter interviewed the daughter of the senator who was controversial," the underlined RC can modify the higher noun "daughter" or the lower noun 'senator, thus having a high attachment (HA) or low attachment (LA) interpretation. Prior work shows that prosody influences the interpretation of ambiguous structures (Lehiste, 1973; Nespor & Vogel, 1983) but does not examine specific acoustic cues distinguishing between RC interpretations. An elicited production task was conducted involving 40 target HA and LA RCs. Intermediate boundaries at the higher and lower nouns were examined for prosodic phrasing cues including pauses, syllable lengthening, boundary tones, and final creak. Delaying of prenuclear peaks was also considered. Data from 8 speakers show that 70% of HA utterances contained two intermediate boundaries, compared to 57% of LA utterances. Main results show that the majority of first intermediate boundaries were high (H-) (81% for LA, 70% for HA), while second intermediate boundaries were usually low (L-) (78% for LA, 73% for HA). In addition, LA interpretations had prenuclear delayed peaks on the higher nouns more often (50% vs. 36% of HA cases).

1pSC26. Dissimilatory interactions among lexical tones are mediated by speech rate: Evidence from Thai. Francesco Burroni (Linguist., Cornell Univ., 203 Morrill Hall, Ithaca, NY 14850, fb279@cornell.edu) and Sam Tilsen (Linguist., Cornell Univ., Ithaca, NY)

The pitch targets of lexical tones in adjacent syllables exhibit interactions that are reminiscent of the coarticulatory effects typical of segments (Gandour, 1994; Xu, 1994). However, unlike segmental coarticulation, which is primarily assimilatory, lexical tones also exhibit dissimilatory interactions. For instance, tones with a high pitch target have been reported to reach even higher targets before a dissimilar low tone than before a similar high tone; a phenomenon known as pre-low raising (Lee *et al*., 2021). Our contribution focuses on two aspects of tonal coarticulation: the existence of dissimilatory pre-high lowering effects, and the role of speech rate in coarticulatory effects. A production experiment was conducted with 20 speakers of Bangkok Thai, who produced disyllabic combinations of Falling/Rising tones before all the five tones of the language (Mid/Low/Falling/High/Rising) in carrier sentences at a variety of speech rates. Results show the existence of both pre-low raising and pre-high lowering dissimilatory effects in Bangkok Thai. Further, the dissimilatory effects are weaker at higher speech rates. The interaction between speech rate and the strength of dissimilatory effects suggests that tonal dissimilation may be primarily driven by articulatory target planning mechanisms rather than perceptual enhancement (Lee *et al*., 2021).

1pSC27. Evidence for F0 preplanning with delayed stimuli. Seung-Eun Kim (Linguist., Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, sk2996@cornell.edu) and Sam Tilsen (Linguist., Cornell Univ., Ithaca, NY)

A sentence production experiment was conducted to investigate F0 preplanning, specifically examining whether speakers plan pitch targets according to utterance length and whether preplanning limits dynamic control of pitch range. The experiment used a novel elicitation method in which visual stimuli that cue part of the utterance are delayed so that participants initiate the utterance without knowledge of its length. Participants read sentences in which the subject noun phrase was composed of one, two, or three conjoined noun phrases with controlled lexical and phonological content

(e.g. *Eight red weasels and nine green rhinos and eight blue llamas live in the zoo*). On half of the trials with two and three noun phrases, the presentation of the visual stimuli that cued non-initial phrases (e.g., *nine green rhinos and eight blue llamas*) was delayed until after detection of utterance initiation. The effects of length and delay on F0 measures were analyzed. The results showed the length effect; the utterance-initial F0 peak was highest when three phrases were presented initially and lowest when one phrase was presented initially. Moreover, the utterance-initial peak in delayed stimuli was similar to that of the single phrase stimuli. In non-initial phrases, the pitch range of the delayed stimuli was constrained such that the F0 peak of the stimuli without delay was higher than those with delay. The findings suggest that speakers plan F0 targets and range before production, and such planning restricts subsequent control of F0.

1pSC28. Intonational phonology of Farasani Arabic: intonation ignores stress unless a word is focused. Abeer A. Abbas (Linguist., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, abeer1@g.ucla.edu) and Sun-Ah Jun (Linguist., UCLA, Los Angeles, CA)

This study proposes an Autosegmental-metrical model of intonational phonology of Farasani Arabic, an under-documented dialect of Arabic spoken in the Farasan Islands, Saudi Arabia. Tonal patterns of utterances, produced in neutral and narrow focus contexts, were examined by varying the length of a word and a phrase, the location of stress, syntactic structures, and sentence types. Results suggest that Farasani Arabic has three prosodic units higher than a word: Intonational Phrase > Intermediate Phrase > Accentual Phrase (AP). An AP is defined by a rising tonal pattern, [L Ha], without including a pitch accent, even though the language has stress. This is unique among Arabic dialects and also typologically unusual. So far, only a few languages (e.g., Kuot, Uyghur) have been claimed to be an exception to the association between stress and intonational pitch accent. A further unique feature of Farasani Arabic is that a pitch accent (H*) does occur on the stressed syllable of a word when the word is emphasized (in this case, the AP final tone is Low, i.e., [L H* La]). In other words, stress is *partially* involved in the intonation system of Farasani Arabic, further challenging the current models of intonational phonology and prosodic typology.

MONDAY AFTERNOON, 29 NOVEMBER 2021 — COLUMBIA C, 4:00 P.M. TO 5:30 P.M.

Session 1pID

Interdisciplinary: Keynote Lecture: An Unexpected Journey Through Sound

Peggy B. Nelson, Chair
University of Minnesota, Speech, Language and Hearing Sciences, Minneapolis, MN 55455

Chair's Introduction—4:00

Invited Paper

4:05

1pID1. An unexpected journey through sound. Tyrone M. Porter (Dept. of Biomedical Eng., Univ. of Texas at Austin, 107 W Dean Keetom St., TX 78712, tmp6@utexas.edu)

Growing up in Detroit in the 1980s, I never envisioned a career in acoustics…not because I was told I could not but because I wasn't told I could. I did not learn about the academic and professional opportunities in biomedical ultrasound until halfway through my undergraduate program. But as a doctoral student, the lack of Black representation in acoustics raised many doubts and fears…doubts in my own abilities and fears that the acoustic community would not accept me. I was embraced by members of the ASA, which empowered me to take the road less traveled. I am here today to share how this unexpected journey through sound has shaped me personally and professionally and thoughts on how I and the ASA can continue to broaden participation in acoustics.

Session 1eID

Interdisciplinary: Tutorial Lecture on Software Best Practices

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. Software best practices. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu)

Software tools play an increasingly important role in modern day acoustics research. We will share a few core software best practices and tools that can help make research processes and outcomes more reproducible and at the same time facilitate collaboration. We plan to cover coding best practices with emphasis on tests and documentation, which contribute to code that is more reusable and robust to errors. We will also cover the use of code and data repositories for collaborative research and show open cloud-based platforms suitable for sharing research outcomes. We will conclude the tutorial by an open discussion on the practicalities of these guidelines and invite all participants to share their perspectives and experiences with the group. The tutorial will include presentation, demos and hands-on exercises, so please bring your laptop to the tutorial.

Exhibit

An instrument and equipment exhibition will be located in the Columbia Ballroom prefunction area.

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, 29 November, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary beverages.

Tuesday, 30 November, 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: 4 December, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. break serving coffee.

Session 2aAAa

Architectural Acoustics, Signal Processing in Acoustics and ASA Committee on Standards: Recording and Production Spaces

Ronald Eligator, Chair
Acoustic Distinctions, 825 Eighth Avenue, 18th Floor, New York, NY 10019

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. Spaces for recording and production in the streaming age. Marcus R. Mayell (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, mmayell@thresholdacoustics.com), Tim Perez, and Carl P. Giegold (Threshold Acoust., Chicago, IL)

The creation, curation, and publication of recorded audio/video content is no longer restricted to professional recording studios and production entities. More accessible and affordable means of audio/video capture, broadcasting, and distributing (primarily through streaming and web-based methods) have given increase to the prominence of self-produced content. The expectations and demands for recording equipment and environments in an era of increasing on-demand and self-op functionality lead to a new set of design parameters. For spaces like these, complex technological and architectural systems are often required for seamless, uninterrupted, and unhindered content capture and delivery. For other spaces, such as Guarneri Hall in Chicago, the architectural and technological designs meet the needs of both high-quality audio/video recording and live streaming alongside a live "in-room" audience – all achieved within a straightforward design approach and constrained budget. This paper will focus on emerging space types which are designed to meet recording needs with an eye toward efficiency, simplicity, and the context of their delivery media.

8:30

2aAAa2. Bigger than they appear: Electronic architecture in recording studios. Russ Berger (RBDG, 2717 North Surrey Dr., Carrollton, TX 75006, russberger2@gmail.com), Steve Barbar (eCoustic Systems, Belmont, MA), Richard Schrag, and Jon Birney (RBDG, Addison, TX)

A significant concern for recording in acoustically small spaces is that they often sound zingy, splashy, fluttery or resonant, or (with attempts at treatment) small, choked, dry, and lifeless. Either of these extremes is not conducive to a good recording experience for the performer or the recording engineer. At the same time, variable acoustics has been an elusive goal in small spaces. Passive solutions are generally ineffective, since one's perception of a small room's acoustical character is driven primarily by the room volume and the mean free path. Technological advancements in electronic architecture, and specific adaptations of that technology to the challenges of deploying it in small spaces, have made it an affordable application for recording studios. When combined with appropriate natural acoustics, this approach can achieve acoustical results not possible by treatments alone. The apparent volume, surface treatments, and resulting reverberant energy can be modified in real time beyond the constraints of the physical space, both for what the performer experiences within the space, and for what the engineer can capture in a recording. The essential considerations for using electronic architecture in a recording studio are illustrated using examples from recent projects.

8:55

2aAAa3. Beneficial loudspeaker/room/listener interactions. Floyd E. Toole (Harman Int. (retired), 1301 King James Ct, Oak Park, CA 91377, soundnwine@sbcglobal.net)

Sound reproduction of the highest quality requires an optimized combination of loudspeakers, rooms and listeners—electroacoustics, acoustics and psychoacoustics. Domestic and monitor loudspeakers have improved because of research investigating the physical sounds arriving at listeners in small rooms and correlating double-blind subjective ratings of sound quality with relevant anechoic measurements on loudspeakers. Loudspeakers designed using specific anechoic measurements and guidance (e.g., ANSI/CTA-2034-B) reliably yield high sound quality ratings in double-blind listening tests in normally reflective domestic rooms. The relative subjective ratings of loudspeakers remain substantially unchanged when comparison listening tests are conducted in rooms with very different acoustics, although the **absolute** sound quality has changed. Listeners clearly adapt to some of the essential perceptual characteristics of rooms. But there are limits. About 30% of subjective ratings of sound quality is attributable to bass quality. This is different for every arrangement of loudspeakers and listeners in every different room. Seat-to-seat variations at low frequencies are large. Absorption is traditionally used to damp bass resonances. However, for reproduced sound, multiple subwoofers, with or without signal processing, can manipulate room resonances. These schemes, with equalization, can substantially attenuate audible resonances while yielding much reduced seat-to-seat variation, using no passive acoustical devices.

9:20

2aAAa4. Design of a flagship studio suite for a major recording arts educational program. Hyun Paek (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com), Gary W. Siebein, Jennifer Miller, Matthew Vetterick, and Marylin Roa (Siebein Assoc., Gainesville, FL)

The architectural acoustic design of a studio, isolation booth and control room as well as an intimate performance hall for amplified music was undertaken for a major recording arts educational program. The university which already has many recording studios, considered this facility as their "Audio Temple" or the showpiece of the entire campus. Acoustical design criteria for reverberation time, low frequency sound control, room response, diffuseness, background noise and sound isolation were established in conjunction with the faculty who will use the spaces. There are classroom spaces above the studio suite and mechanical equipment located near-by, so sound isolation design was an important part of the design effort. A three-layer sound isolation system was used for each of the rooms. The faculty had specific requirements for the interior acoustical qualities of the rooms. The paper presents comparisons of the rooms as designed with the criteria for acoustical quality described by the users for reverberation, low frequency sound, room response, diffuseness and background noise. Efforts were maintained during design and construction to preserve these qualities through the realities of the value engineering and construction process.

9:45

2aAAa5. Acoustical design of an evolving post production facility at a creative arts college. Hyun Paek (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com), Gary W. Siebein, Matthew Vetterick, Jennifer Miller, and Marylin Roa (Siebein Assoc., Inc., Gainesville, FL)

A private college began to consider building a screening room and production studio in 2010. The original site was in an existing multi-media exhibit building with the studio located under the main computer server for the campus and next to a mechanical room. After bringing this project very close to fruition, the program projections had already outgrown the project. The college decided to expand the program and build a combined post production and recording facility with three sound stages on an empty piece of land on the edge of the campus. The design evolved in conjunction with the sound designer from New York, production consultant from LA, the architect, the acoustical consultant, the college, a contractor and the rest of the design team. Once the design was nearing completion and the building was being constructed, many changes were initiated by the users to reflect their current ideas. Computer models of the spaces were constructed and the acoustical designs tested for each of the evolving solutions. The evolution of the project continued as equipment was being selected for the room. When the project was almost completed a Dolby Atmos system was purchased for the main screening room so that even after construction was almost complete, additional design changes had to be made and evaluated. Post construction acoustical testing and the critiques of the users have all been exceptionally positive given the long and winding road that the project took to completion.

10:10–10:25 Break

10:25

2aAAa6. Converting an empty office to a podcast studio. Indi Savitala (Acoust., CSDA Design Group, 610 E. Franklin Ave., El Segundo, CA 90245, indi@csdadesigngroup.com)

Companies of various types are converting empty offices to podcast studios. The existing office space presents many acoustic challenges: (1) single-glazed office front, (2) proximity to noise producing adjacency, (3) limited available surface area for room finish treatments, and the list goes on. This presentation analyzes an existing office space that was converted to a podcast studio and presents criteria and recommendations that were used to optimize the space for recording.

10:50

2aAAa7. Transformation of a warehouse into a professional production studio. Hyun Paek (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, hpaek@siebeinacoustic.com), Gary W. Siebein, Jennifer Miller, Marylin Roa, and Matthew Vetterick (Siebein Assoc., Inc., Gainesville, FL)

A major film production studio was slated for demolition due to the expansion of a theme park. Audio tracks for many popular films and animated productions were designed, recorded and mixed in the facility. Space for the new studio was included in a large office renovation project that was occurring in several large existing warehouse/office buildings. The acoustical challenges of providing sound isolation between and among adjacent recording, control and editing spaces as well as between the studios and the exterior which included building box-in-box-in-box isolation systems within the existing warehouse posed an interesting challenge. With the unforgiving structural system of the existing warehouse building, thinking both outside and inside of the box was necessary. Fitting the low velocity ductwork in the relatively low ceiling height of the existing structure was not easily accomplished. Each of the existing limitations of the warehouse building was overcome by closely working with the Design Team and the Contractor. Acoustical shaping and finish materials were designed and selected to meet the criteria of the users for reverberation, room response, low frequency sound control, diffuseness and background noise. The users were pleased to work in a studio that allowed the artists and the engineers to work throughout the day into the night and through thunderstorms. The studio has been very well-received by the users, since the resulting sound tracks translated very well when played back in a variety of venues.

11:15

2aAAa8. A study of a renovated broadcast control room with a concave enclosure. Ronald Eligator (Eligator Acoust. Assoc., 825 Eighth Ave., 18th Fl., New York, NY, 10019, religator@eligatoracoustics.com) and Arthur W. van der Harten (Acoust. Distinctions, Stamford, CT)

Broadcast control rooms are monitoring stations which can have many operators. One such room with ellipsoidal plan shaping was recently renovated. The original room had sufficient absorptive wall treatment, such that any focusing effects were less obvious, and the occupants could perform their work in relative comfort. The renovated space included much more computer screen equipment—so much so that the original treatment was covered, and the concave surface shaping was duplicated in hard surfaces. The resulting space was distracting for the occupants, due to whispering gallery effects, and at times could become excessively loud. In this talk, we discuss our process for measuring anomalous effects in the space using room impulse response measurement techniques, as well as our computer modelling approach, which helps us identify the surfaces implicated in the measured anomalies. We will also discuss our proposed solutions, and the challenges associated with each.

Contributed Paper

11:40

2aAAa9. Design of spaces for recording and live broadcast within existing physical design limitations. Dana S. Hougland (Shen Milsom & Wilke, LLC, 1822 Blake St., Ste. 2A, Denver, CO 80202, dhougland@smwllc.com) and Robert Healey (Shen Milsom & Wilke, LLC, Denver, CO)

It is optimum to start the design of studios for live broadcast and recording using ideal proportions. At times existing building conditions limit the design height or configurations that are possible. This paper examines the design of three studios for live broadcast that are confined either by height, size, location, or a previous design. The three studios serve as live broadcast spaces for classical music, contemporary music, interviews, and jazz. Two of the studios include options which integrate live audiences. Design solutions are presented as well as measured performance results. Results will include before and after testing where existing studio acoustic conditions were modified.

TUESDAY MORNING, 30 NOVEMBER 2021 COLUMBIA A, 8:00 A.M. TO 11:00 A.M.

Session 2aAAb

Architectural Acoustics: Student Design Competition

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

Ian Hoffman, Cochair
Peabody Inst.,

Daniel Butko, Cochair
Architecture, The University of Oklahoma, 830 Van Vleet Oval, Gould Hall 265, Norman, OK 73019

The Student Design Competition is sponsored by the ASA Technical Committee on Architectural Acoustics, with generous support from the Wenger Foundation, the Robert Bradford Newman Student Award Fund, and the National Council of Acoustical Consultants. This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significant importance. Submission requirements will include a digital submission for judging and a printed copy of the submission for display at the Seattle meeting.
Design Scenario: The project involves designing a performance hall primarily for opera for a college of moderate size with a very strong music program. Although the main purpose of the hall is to support their opera program, the hall may be used occasionally for symphony orchestra, chamber music, chorus, and dance. See the full announcement for more information.

TUESDAY MORNING, 30 NOVEMBER 2021 ROOM 602, 8:00 A.M. TO 11:00 A.M.

Session 2aAB

Animal Bioacoustics and ASA Committee on Standards: Classifying and Quantifying Natural Soundscapes I

Michael Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Chair's Introduction—8:00

Invited Papers

8:05

2aAB1. Soundscape metrics: What are we measuring? Michael Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@ocr.org)

The coinage of the term "Soundscape" can trace its origin back to composer R. Murray Schafer's soundscape studies in the 1960s, along with his seminal book "Soundscape: The Tuning of the World," which framed the term in the phenomenological context of how we inhabit our acoustical environments. The term remained largely a philosophical conceit based on subjective impressions of inhabited sound environments until it was reined in by way of ISO 12913.1:2014 (Definition and conceptual framework), and ISO 12913.2:2014 (Data collection and reporting requirements). In these standards the conceptual framework includes (1) context, (2) sound sources, (3) acoustic environment, (4) auditory sensation, (5) interpretation of auditory sensation, (6) responses, and (7) outcomes. With the exception of (4) auditory sensation, and (5) interpretation of auditory sensation, these characteristics could all be used in non-human bioacoustic assessments and standards—utilizing the same indicators (measured by instruments). Where our toolbox falls short is in the

"descriptors" because we have not yet found much success translating animal interpretations of their experience of their sound environments. In order to derive some predictable metrics of these, we would need to develop reliable ways of interpreting responses and outcomes correlated to context, sound sources, and acoustic environments.

8:25

2aAB2. The need to standardize soundscape analysis metrics—A case study: Comparing temporal and spectral kurtoses of underwater sounds. S. B. Martin (Halifax, JASCO Appl. Sci., Halifax, NS, Canada, bruce.martin@jasco.com) and Zahra Alavizadeh (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada)

To evolve our ability to compare soundscapes across time, location and research groups, a standard suite of metrics is essential. Significant progress towards establishing standard approaches for quantifying the amplitude of the soundscape has been made in recent years, however advancing qualitative metrics has been challenging. Temporal kurtosis is often proposed as a soundscape metric that quantifies impulsiveness, and that filtering a sound for an animal's audiogram could quantify the impulsiveness for animals with different hearing capabilities. This frequency-weighted kurtosis could be computed by filtering in the time domain, or perhaps by computing the spectral kurtosis over the hearing range. Comparing the two approaches provides a case study for the importance of standardizing analysis methods. Here the temporal and spectral kurtosis of common underwater sounds are compared. The band-limited temporal and spectral kurtoses are correlated for sounds from vessels, natural sounds, and seismic airgun surveys. They are substantially different for impact and vibratory pile- driving and the spectral kurtoses depend heavily on the analysis parameters, a result that is similar to a previous study of the acoustic complexity index. It is concluded that for qualitative soundscape metrics, standardized analysis methods are essential.

Contributed Papers

8:45

2aAB3. Hierarchical classification of multi-corpus acoustic field recordings. Mallory Morgan (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, morgam11@rpi.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Passive acoustic monitoring is one important technique for detecting long-term ecological change. Since audio data contains complex temporal and frequency information, deep learning is an ideal computational framework for automatic processing. In this work, the dataset consists of 64k annotated clips distributed across 23 classes that are derived from 17 000 h of acoustic data consisting of bio-, geo- and anthropophonies recorded at two different sites in the Capital Region of New York State. A hierarchical deep convolutional neural network with one local classifier per parent node provides fine, medium, and coarse-grained label predictions. If a fine-grained label is predicted incorrectly, a correct medium or coarse label still provides valuable taxonomic information. While the flat classifier performs similarly to the hierarchical for single-corpus training scenarios, it achieves higher medium-grained predictions for categories such as avian vocalization. For cross-corpus scenarios, the hierarchical classifier achieves a superior performance overall. These results highlight the importance of neural network architecture customization to suit a particular task and domain; hierarchical structures show potential in open set recognition tasks, where the entire set of possible classes is unknown. [Work supported by NSF Grant No. 1631674 and an RPI HASS Fellowship.]

9:00

2aAB4. The next wave of passive acoustic data management: How centralized access can enhance science. Carrie Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, 325 Broadway, Boulder, CO 80305, carrie.bell@colorado.edu), Samara Haver (Cooperative Inst. for Marine Ecosystem and Resources Studies, NOAA Pacific Marine Environ. Lab. and Oregon State Univ., Newport, OR), Leila Hatch (Gerry E. Studds Stellwagen Bank National Marine Sanctuary, NOAA Office of National Marine Sanctuaries, Scituate, MA), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), Rob Bochenek (Axiom Data Sci., LLC, Anchorage, AK), Robert P. Dziak (Pacific Marine Environ. Lab., NOAA, Newport, OR), and Jason Gedamke (Office of Sci. and Technol., NOAA Fisheries, Silver Spring, MD)

Passive acoustic data collection has grown exponentially over the past decade resulting in petabytes of data that document our ocean soundscapes. This effort has resulted in two big data challenges: the curation, management, and global dissemination of passive acoustic datasets and efficiently extracting critical information and comparing it to other datasets in the context of ecosystem-based research and management. To address the former, the NOAA National Centers for Environmental Information established a passive acoustic data archive, which contains over 100 TB of audio files mainly collected from stationary recorders throughout waters in the U.S. These datasets are documented with standards-based metadata and are freely available to the public. To begin to address the latter, through standardized processing and centralized stewardship and access, we will present a previously unattainable comparison of first order sound level-patterns from archived data collected across three distinctly separate long-term passive acoustic monitoring efforts conducted at regional and national scales: NOAA/National Park Service Ocean Noise Reference Station Network, the NOPP-funded Atlantic Deepwater Ecosystem Observatory Network, and the NOAA-Navy Sanctuary Soundscape Monitoring Project. Further, we will propose the next frontier for scalable data stewardship, access, and processing flow to help the community collaboratively move forward.

9:15

2aAB5. Listening space reduction calculations. David E. Hannay (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC, Canada, david.hannay@jasco.com)

Listening space reduction analysis has been applied recently by several authors to evaluate the relative reduction of distance or space within which listeners can effectively detect and recognize important sound sources, due to increased acoustic masking caused by a known increase in masking noise. The approach balances the increased masking noise level by reducing source-listener distance so that propagation loss is commensurately reduced. Its recent applications have not considered all important parameters, including listener hearing sensitivity and directionality, listener cognitive signal processing gain, non-uniform acoustic propagation loss, and temporal and spatial distributions of masking noise variability. Here we derive the listening distance and space expressions from the sonar equation, accounting for the parameters above. Listening space reduction calculations are insensitive to some uncertain parameters, including source level and detection threshold (for non-human listeners) that are required for communication space calculations. The method is applicable to any sound type, including predator sounds, prey sounds, conspecific communications, and natural sounds. A modified version of the method is derived for echolocation sounds, and a corresponding echolocation space reduction factor is derived.

Invited Papers

9:30

2aAB6. Global and regional soundscape modeling for ships and wind. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Christopher Verlinden, James Murray, Andrew Heaney, and Tessa Munoz (Appl. Ocean Sci., Fairfax Station, VA)

There has been a growing understanding of the importance of the ocean soundscape on marine ecosystem health. With sound traveling to great distances in the open ocean, particularly at lower frequencies (below 200 Hz) ships that are a great distance away can impact the local sound levels. There is a need to be able to understand the relative portion of the measured sound levels which can be attributed to anthropogenic sources (ships, seismic survey vessels, navy sources, pile driving) and to natural sound sources (wind and waves, marine mammals, earthquakes, ice). Modeling approaches provide the opportunity to evaluate the excess noise for a given region (anthropogenic sound levels above the natural or ambient levels). A set of results are presented where the shipping and wind levels are computed for a single measurement position, a regional scale (Atlantic Outer Continental Shelf, the Arctic) and for the entire world's oceans. Modeling results for the increase in shipping noise levels in the Arctic Ocean due to reduced ice extent and a longer summer transit season are presented, as well as for the reduction on global sound levels due to the initial onset of COVID-19.

9:50–10:05 Break

10:05

2aAB7. Using principal component analysis for marine soundscape research: A case study from Taiwan. Shane Guan (Environ. Studies Program, Bureau of Ocean Energy Management, 45600 Woodland Rd., Sterling, VA 20166, guan@cua.edu) and Tzu-Hao Lin (Biodiversity Res. Ctr., Academia Sinica, Taipei, Taiwan)

Recent years have seen enormous increases in research on marine soundscape, as underwater acoustic sensors are becoming more affordable, coupled with a growing interest and increased concerns on underwater noise impacts on marine fauna. However, when faced with large amounts of data, there are significant challenges regarding how to analyze large datasets effectively and accurately to make informed decisions. In this study, we used principal component analysis (PCA) to analyze the temporal, spatial, and spectral patterns of the soundscape from major contributing sources in a shallow water coastal environment at two locations in Taiwan in early and late summer of 2012. Acoustic intensity levels were analyzed in three frequency bands: 150–300 Hz, 1.2–2.4 kHz, and 3–6 kHz. The PCA results show higher acoustic intensity in the 1.2–2.4 kHz band at night than during daytime hours, and relatively high acoustic intensity levels overall in the 150–300 Hz frequency at all times. In addition, a large temporal relationship was observed between early and late summer in the 1.2–2.4 kHz band. The results show that PCA has the potential to provide a simple but reliable way to quantify the dynamics of marine soundscapes.

Contributed Paper

10:25

2aAB8. Real-time detection and classification of underwater soundscape signals via a convolutional autoencoder. Val Veirs (Beam Reach SPC, 7044 17th Ave. NE, Seattle, WA 98115, vveirs@coloradocollege.edu) and Scott Veirs (Beam Reach SPC, Seattle, WA)

Underwater sounds come with a wide range of frequency components and with a wide range of time intervals. For example, individual killer whale (*Orcinus orca*) calls are typically one or two seconds long. Humpback whale (*Megaptera novaeangliae*) calls range from second long moans to half-hour songs. Speed boats can pass a hydrophone in seconds to minutes. Large ships create signals above background for many minutes. Here fixed 100x100 dimension spectrograms having exponentially increasing time windows of 3, 15, 75, 375, 1875, 9375 seconds are calculated and classified in real time via an autoencoder used as an acoustic pattern detector followed by a supervised convolutional neural network with softmax output. The autoencoder was trained on underwater sound ($\sim$1 Tb) at Orcasound Lab, San Juan Island, WA. The system is designed to classify incoming real-time signals to one or more of twelve different classes (background, killer whale, Southern Resident, Biggs killer whale, humpback whale, breaking waves, kayak paddles, speedboat, displacement boat, large ship, large ship with shaft rub, large ship with repeating sound pattern). The design and efficacy of this detection/classification system will be reported.

Invited Paper

10:40

2aAB9. Large-scale assessment of masking potential by underwater ship noise the North Sea. Jakob Tougaard (Bioscience, Aarhus Univ., Zoophysiology, Bldg. 1131, Aarhus 8000, Denmark, jat@bios.au.dk), Thomas Folegot (Quiet Oceans, Brest, France), Christ de Jong (TNO, The Hague, The Netherlands), Emily T. Griffiths (Bioscience, Aarhus Univ., Aarhus, Denmark), Alexander M. von Benda-Beckmann (TNO, The Hague, The Netherlands), Nathan D. Merchant (CEFAS, Lowestoft, United Kingdom), and Niels A. Kinneging (Rijkswaterstaadt, Amsterdam, The Netherlands)

The JOMOPANS project created one year of soundscape maps of the North Sea, based on ship information from the Automatic Identification System (AIS) and a ship source model. This assessed the potential impact of ship noise on the marine ecosystem through the masking of communication and other relevant acoustic signals. Specifically, the excess noise was quantified, which express the

deterioration of the signal-to-noise ratio caused by the ship noise and hence the reduction in maximum communication range. The dominance of ship noise over natural sources was quantified as the percentage of time where excess was 20 dB or higher. A decrease in signal-to-noise ratio of 20 dB compared to a reference condition (ambient plus weather) without ships indicates a substantial decrease in communication range, and hence a substantial risk of impact on the ecosystem. Maps of excess and dominance in different frequency bands showed the largest ship contribution in the lowest frequency bands and in the most trafficked areas, with less affected areas in shallow waters, including the Dogger Bank. These maps, in combination with distribution maps of sensitive species, form the basis for the joint assessment of environmental status of the North Sea by the surrounding countries.

TUESDAY MORNING, 30 NOVEMBER 2021 ELWHA B, 8:20 A.M. TO 11:20 A.M.

Session 2aAO

Acoustical Oceanography and Signal Processing in Acoustics: Acoustical Oceanography Using Ocean Observatory Systems II

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

Andone C. Lavery, Cochair
AOPE, Woods Hole Oceanographic Institution, 98 Water Street, Woods Hole, MA 02543

Felix Schwock, Cochair
Electrical and Computer Engineering, University of Washington, 185 Stevens Way, Paul Allen Center—Room AE100R, Seattle, WA 98195

Contributed Papers

8:20

2aAO1. Using Axial Base cabled array sensor data to study the prediction of second-order pressures from nonlinear surface wave interactions associated with microseisms. Umesh A. Korde (Johns Hopkins Environ. Health & Eng., Johns Hopkins Univ., Baltimore, MD) and Michael S. McBeth (Res. and Appl. Sci., Naval Information Warfare Ctr. Atlantic, NASA Langley Res. Ctr., 8 North Dryden St., M.S. 473, Hampton, VA 23681, michael.s.mcbeth@navy.mil)

There are currently no published experimental measurements of second-order pressures in the water column from nonlinear surface wave interactions. Microseism researchers have focused on predicting and measuring microseisms directly without measuring second-order pressures in the water column. We considered several Ocean Observatory regional cabled arrays as sources for water column measurements to compare with analytical model second-order pressure predictions based on hindcast directional wave spectra. We selected the Axial Base cabled array for its location and suite of instruments including two hydrophones, 3D velocity sensor, and seismometer near the seafloor. Here we present results for predicted second-order pressures along with hydrophone measurements at the Axial Base site over different times of a year (2019–2020) to demonstrate a correlation. Validating a prediction model for second-order pressures based on directional surface wave spectra using the OOI cabled array sensor data could spur development of wave energy harvesting devices in the deep ocean and other applications.

8:35

2aAO2. Data-driven approaches for ray-based ocean acoustic tomography. Priyabrata Saha (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA, priyabratasaha@gatech.edu), Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., Atlanta), Jihui Jin, Justin Romberg, Saibal Mukhopadhyay (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ray-based acoustical tomography typically provides estimates of local ocean sound speed fluctuations from precise measurements of acoustic travel times fluctuations (with respect to a reference environment) between multiple sources and receiver arrays along with a model of the ray propagation in the reference (i.e., fluctuation-free) environment. Traditionally, this is achieved using a sensitivity kernel, a Born–Frechet linearization of the non-linear relationship between ray arrival-times and sound speeds fluctuations. However, computing the full sensitivity kernel can be computationally intensive and is restricted to only small sound-speed fluctuations (due to the linearization procedure) and does not inherently leverage any available history of the arrival time or sound speed fluctuations which may have been collected prior to performing the inversion procedure. As an alternative solution for ray-based acoustical tomography, we investigated the use of data-driven methods, such as linear regression, nearest neighbors, neural networks, and kernel (ridge) regression, to directly learn the non-linear relationship between sound speed fluctuations and ray arrival time variations

using previously collected measurements. Additionally, the experimental training data set is augmented by generating synthetic SSPs based on empirical orthogonal function analysis of the historical SSP data. This approach is demonstrated for short vertical bottom-mounted arrays deployed in littoral environments.

8:50

2aAO3. Trans-dimensional inversion for sediment attenuation using modal dispersion data collected by an underwater glider. Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W2Y2, Canada, minj@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This paper inverts modal-dispersion data collected by an underwater glider to estimate a layered seabed geoacoustic model including sediment attenuation as well as sound speed and density. The data considered here were collected during the 2017 Seabed Characterization Experiment (SBCEX17) conducted on the New England Mud Patch. The distance between the (combustive) sound source and the hydrophone-equipped (Teledyne Webb Research Slocum) glider was approximately 8 km. The frequency range considered was from 20 to 392 Hz, and 6 modes were resolved from the data using warping. Previous work has demonstrated that the positions of dispersive acoustic modes in the time-frequency plane provide information to estimate seabed sound-speed and density profiles. Including the relative energy distribution in the time-frequency plane (normalized modal amplitudes and/or amplitude ratios between modes) provides additional information to also estimate sediment attenuation, which is carried out here using a trans-dimensional Bayesian inversion. In addition, uncertainties of the seabed sound speed and density with/without modal energy information are compared. [Work supported by ONR.]

9:05

2aAO4. Estimating ocean variables using ambient noise interferometry. John Ragland (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA, jhrag@uw.edu) and Shima Abadi (Univ. of Washington, Seattle, WA)

Ambient noise interferometry is a method of estimating the time domain Greene's function (TDGF) by correlating and averaging sound from adjacent hydrophones. These averaged cross correlations are called noise cross correlation functions (NCCFs). Noise interferometry has been used in ocean acoustics to estimate ocean variables such as water temperature by measuring propagation times. In this talk, a 6-year NCCF stack between two Ocean Observatories Initiative cabled array hydrophones is investigated. The hydrophones are separated by 3.2 km, and are bottom mounted at a depth of 1500 meters. They are located in the caldera of the Axial Seamount volcano approximately 470 km off the Oregon coast. The TDGF estimate from the NCCF stack contains clear, multipath arrivals. Fluctuations in the arrival time over the six years are on the order of a single time bin. However, if the signal to noise ratio is high enough, it is possible to measure the arrival times with more accuracy than a single time bin. In this presentation, signal processing methods to accurately estimate the propagation time will be explored, as well as the viability of inverting these multipath arrival times to estimate ocean variables such as water temperature. [Work supported by ONR.]

9:20

2aAO5. Rapid estimation of empirical Green's functions for passive acoustic characterization of dynamic shallow-water environments. Tsu Wei Tan (Dept. Marine Sci., ROC Naval Acad., No. 669,Junxiao Rd., Zouying Dist., Kaohsiung 81345, Taiwan, ttan1@nps.edu), Oleg A. Godin, and Yi-fan Shen (Phys. Dept., Naval Postgrad. School, Monterey, CA)

Applications of acoustic noise interferometry to passive remote sensing of the ocean and improvement of sonar performance predictions rely on retrieval of empirical Green's functions (EGFs) from synchronized measurements of ambient sound at spatially separated points. Usually, long noise averaging times of many hours and even days are employed to achieve precise passive measurements of the ray or normal mode travel times that are necessary for evaluation of the environmental parameters with oceanographically relevant accuracy. However, long averaging times have limited utility when nonlinear internal waves or other oceanographic processes cause significant, time-dependent variations in sound propagation conditions. Analysis of the noise records acquired during the Shallow Water 2006 experiment reveals that EGFs can be reliably retrieved from cross-correlations of noise recorded over periods as short as 10 minutes by moored receivers horizontally separated by 40–50 ocean depths. The measured EGFs evolve as the sound speed field changes. Suitability of the EGFs for acoustic remote sensing is demonstrated by geoacoustic inversion of the passively measured normal mode dispersion curves and comparison to previously reported inversion results. Rapid EGF retrieval has far-reaching implications for operational use of the underwater acoustic noise interferometry. [Work supported by NSF.]

9:35–9:50 Break

9:50

2aAO6. Quantifying the influence of the range and motion of shipping sources on the ray-based blind deconvolution (RBD) algorithm. Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Office 131, Atlanta, GA 30332, rtouret@gatech.edu), Nicholas C. Durofchalk, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The ray-based Blind Deconvolution (RBD) can estimate the relative arrival-times of the channel impulse responses (CIRs) between a shipping source (acting as a shallow source of opportunity) and a remote receiver array (e.g., a short bottom-mounted vertical line array (VLA)) [*Durofchalk & Sabra, JASA Vol. 147(3), p. 1927–1938, 2020*]. Previous RBD studies primarily estimate the phase of the unknown source by using beamforming on the (most energetic) direct path, which is only observable at short ranges for a shipping source in typical downward refracting environments. Alternatively, when dealing with long range sources of opportunity and only bottom-interacting multipath arrivals can reach the VLA (i.e., exceeding the range when the direct path is observable), the general performance of the RBD algorithm remains to be evaluated. In this study, numerical simulations and experimental validation (using short bottom mounted VLAs) are used in conjunction to investigate the maximal detection range of shipping sources of opportunity per multipath arrivals as a function of frequency, Doppler effect and anisotropy of the shipping source, and the implications for the accuracy of the relative arrival time of the CIR estimated with RBD.

10:05

2aAO7. Modeling and observation of low-frequency propagation into warm core ring on the New England shelf slope. Emma Reeves Ozanich (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 86 Water St., Woods Hole, MA 02540, eozanich@whoi.edu), Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Falmouth, MA), and Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, RI)

Analytical and computational modeling is used to examine propagation of low-frequency sound into a warm core ring oceanic front. When the source is placed outside the front and the receiver is placed inside it, the sound field consists of the refracted path and the horizontal lateral wave, which decays exponentially into the front. The analytical solution is presented in 2D (x, y) assuming an adiabatic vertical mode solution in z, wherein the vertical mode-dependence of the propagation paths into the front are considered. A 3D computational acoustic model examines the additional effects of sloping bathymetry and frontal geometry. The modeled results are compared to experimental measurements of into-front propagation during the March 2021 New England Shelf Break Acoustics Experiment using a low-frequency ($f_c = 115$ Hz), shallow towed chirp signal received on an L-shape vertical and horizontal hydrophone array. [Work supported by the Office of Naval Research.]

10:20

2aAO8. Real-time acoustic modeling in the New England Shelf Break Experiment. Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, RI), and Weifeng G. Zhang (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

During the New England Shelf Break Acoustics Experiment (NESBA) in May of 2021, signals emitted and received by instruments on deployed moorings were successfully modeled while at sea. Using a combination of physical oceanography models and data collected from CTD measurements to determine the sound speed, both parabolic equation (PE) and ray tracing methods were used to model signals received by hydrophones as part of the experiment. Qualitative comparison between model output and hydrophone data was used to inform model adjustments to better represent the acoustic data, ultimately producing a promising data/model agreement during the cruise. Post-cruise research involved incorporation of additional data from the NESBA cruises and model adjustments to more adequately reflect the environmental properties of the experiment site. The success of the initial real-time modeling tests and subsequent improvements to the techniques support their use in future field experiments. [Work supported by the Office of Naval Research.]

10:35

2aAO9. An acoustic observatory for sperm whales in the Eastern Mediterranean Sea. Emmanuel Skarsoulis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, N. Plastira 100, Heraklion, Crete GR-70013, Greece, eskars@iacm.forth.gr), George Piperakis, Emmanuel Orfanakis, Panos Papadakis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol. - Hellas, Heraklion, Greece), and Michael Kalogerakis (Elec. & Comput. Eng., Hellenic Mediterranean Univ., Heraklion, Greece)

A deep-water acoustic observatory for real-time detection and 3D localization of sperm whales has been developed, deployed and operated for 3-month periods in summer 2020 and 2021 off SW Crete in the Eastern Mediterranean Sea. Detection and localization rely on receptions of regular clicks at 3 hydrophones suspended from surface buoys at depths of ∼100 m and ∼1 km apart. Travel times of significant arrivals extracted onboard the buoys are transmitted, together with other supporting data, via mobile broadband to a land-based analysis center where the data from all buoys are combined to enable detection and 3D localization of vocalizing animals exploiting direct and surface-reflected arrivals and using a Bayesian approach. The large separations between hydrophones result in small localization uncertainties for ranges up to ∼6 km, whereas, on the other hand, they pose significant challenges related, e.g., to peak association and synchronization between the buoys. The design and implementation, as well as results from the operation of the system will be presented. [Work supported by Ocean Care – SAvEWhales project.]

10:50

2aAO10. Ranging to fin whale calls at full ocean depths using single ocean bottom seismometers. Rose Hilmo (Oceanogr., Univ. of Washington, 1851 NE Grant Ln., Seattle, WA 98195, wader@uw.edu) and William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA)

Fin whale calls are routinely recorded on ocean bottom seismometers (OBSs), but the networks are often too sparse for multi-station call localization. Single station ranging methods are thus important for efforts to utilize OBSs for population density studies. Ranging using the timing and amplitude of multipaths has been demonstrated in relatively shallow (∼2500 m) settings, but in sedimented regions it is challenging to detect the first multipath at small ranges because seafloor reflections have low amplitudes at high incidence angles and subseafloor reflections often have higher amplitudes. To enhance the identification of weak multipaths, we consider all calls in 10-min time windows together. Calls are detected using spectrogram cross-correlation to yield a time-dependent recognition score. The time difference between direct and first multipath arrivals is identified by (1) autocorrelation of the recognition score and (2) spectrogram cross-correlation of the sum of spectrograms for all detected calls. The range is estimated by comparing this time with the predictions of ray tracing. We have tested these methods at five deep (4200–6000 m) OBS sites with contrasting seafloor properties. The results are promising but show the importance of understanding the nature of subsurface reflections.

11:05

2aAO11. Monitoring fin whale calls using Ocean Observatories Initiative Regional Cabled Array. Xuyang Wang (School of Oceanogr., Univ. of Washington, 1851 NE Grant Ln., Seattle, WA 98195, wxuyang1024@gmail.com), Rose Hilmo, William S. Wilcock, Kathleen Gonzalez, and Mouffee Borras (School of Oceanogr., Univ. of Washington, Seattle, WA)

The Ocean Observatory Initiative (OOI) Regional Cable Array (RCA) and the Ocean Networks Canada NEPTUNE cabled observatory provide an opportunity for long-term studies of the distribution and behavior of calling marine mammals. We will present the results of a study to build a database of "20-Hz" fin whale calls that spans from 2015 to present. Fin whale calls were collected by bottom-mounted low-frequency hydrophones and seismometers deployed on the OOI RCA and are detected automatically by spectrogram cross-correlation with a detection kernel that sweeps down linearly from ∼26 Hz to ∼19 Hz with a duration of ∼0.8 s. The kernel is tuned to detect the high amplitude portions of both the high- and low-frequency calls of the doublet calling pattern which is currently the dominant fin whale song in this area. Preliminary results show significant interannual variations at a given site in the distribution of calls during the fall and winter calling season. The onset of calling appears to be significantly earlier near the continental slope off central Oregon than at Axial Seamount, 450 km offshore. There also appears to be a higher overall rate of calling nearer the continental slope.

Session 2aBAa

Biomedical Acoustics, Signal Processing in Acoustics, Physical Acoustics, and Engineering Acoustics: Nonlinear Acoustic Propagation: Theory, Simulations, Experiment, and Applications I

Vera A. Khokhlova, Cochair
Physics Faculty, University of Washington/Moscow State University, Moscow 119991, Russian Federation

Mark F. Hamilton, Cochair
Walker Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1591

Chair's Introduction—8:00

Contributed Papers

8:05

2aBAa1. Large-aperture 256-element fully populated random array for therapeutic ultrasound applications: Experimental study of its functionality and characterization of the radiated field. Oleg A. Sapozhnikov (Phys. Faculty, Moscow State University, Leninskie Gory, Moscow 119991, Russia, olegs2@uw.edu), Pavel B. Rosnitskiy (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Shamil A. Asfandiyarov, Dmitry A. Nikolaev, Sergey A. Tsysar (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), and Vera A. Khokhlova (Phys. Faculty, Moscow State University, Leninskie Gory, Moscow 119991, Russia)

Recently, a new method has been proposed for designing random high-intensity focused ultrasound (HIFU) arrays with the maximum possible filling factor and polygonal elements of equal area, based on capacity constrained tessellation [Rosnitskiy *et al*., IEEE Trans. UFFC (2018)]. Following this approach, a 256-element piezocomposite array with a radius of curvature of 150 mm, a diameter of 200 mm, and an operating frequency of 1.2 MHz was designed at the Laboratory for Industrial and Medical Ultrasound (LIMU) of Moscow State University. The array was subsequently manufactured by Imasonic (France). It can be driven using a Verasonics ultrasound engine. Electrical matching was performed for each element using matching network created by Image Guided Therapy (France). This report presents the system design procedure and experimental results for array characterization using acoustic holography and acoustic power measurements using the radiation force balance. In addition, a theoretical study was carried out using the "HIFU beam" simulator developed at LIMU (freely available at www.limu.msu.ru) to assess the array's ability to achieve highly nonlinear regimes with shocked pressure waveforms. The possibility of using the array for HIFU applications such as acoustic levitation and manipulation, histotripsy and transcostal HIFU therapy is discussed. [Work supported by RSF 19-12-00148.]

8:20

2aBAa2. Elastic properties of human hematoma model and its sensitivity to histotripsy liquefaction. Ekaterina M. Ponomarchuk (Phys. Faculty, Lomonosov Moscow State Univ., Leninskie Gory, 1, b. 2, Moscow 119991, Russian Federation, msu.ekaterina.ponomarchuk@gmail.com), Pavel Rosnitskiy, Sergey A. Tsysar (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Maria M. Karzova, Anastasia V. Tyurina (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Kseniya D. Tumanova (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Alexey V. Kadrev (Medical Res. and Educational Ctr., Lomonosov Moscow State Univ. / Diagnostic Ultrasound Div., Russian Medical Acad. of Continuous Professional Education, Moscow, Russian Federation), Sergey V. Buravkov (Faculty of Fundamental Medicine, Lomonosov Moscow State Univ., Moscow, Russian Federation), Pavel E. Trakhtman, Nicolay N. Starostin (National Medical Res. Ctr. for Pediatric Hematology, Oncology and Immunology, Moscow, Russian Federation), Oleg A. Sapozhnikov, and Vera A. Khokhlova (Univ. of Washington/Moscow State Univ., Moscow, Russian Federation)

The efficiency of various histotripsy approaches for mechanical tissue ablation has been reported to depend on tissue elastic properties. Here the efficiency of histotripsy liquefaction of human hematomas is investigated in dependence on their shear modulus value and retraction degree. Recalcified anticoagulated blood samples (200 ml) served as an *in vitro* hematoma model. Shear wave elastography was used to measure changes of the samples shear modulus during clotting at different temperatures and during 10-day retraction. Boiling (2.5 ms-pulses) and hybrid (200 μs-pulses) histotripsy lesions (2 MHz operational frequency) were produced and sized in samples of varying retraction degrees. The clotting time decreased from 113 min to 25 min with the increase in blood temperature from 10 to 37°C. The shear modulus increased to 0.53 ± 0.17 kPa during clotting and remained constant within 10-day storage at 2°C, whereas the volume of clotted samples decreased by 57% due to retraction. The lesions produced with the same histotripsy protocols in retracted samples were smaller in size versus freshly clotted ones correlating with the increase in retraction degree. The results demonstrate that the hematoma sensitivity to histotripsy liquefaction is not fully defined by its shear modulus. [Work supported by NIH R01GM122859, RFBR 20-02-00210, FUSF, and "BASIS" Foundation 20-2-10-10-1, 20-2-1-83-1.]

8:35

2aBAa3. Doppler ultrasound observations and metrics of boiling histotripsy treatment progression. Minho Song (Mech. Eng., Univ. of Washington, 1013 NE 40th St., CIMU, Portage Bay Bldg., Seattle, WA 98195, songmh@uw.edu), Gilles P. Thomas, Vera A. Khokhlova, Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Adam Maxwell (Urology, Univ. of Washington, Seattle, WA), and Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA)

Boiling histotripsy (BH) is a mechanical tissue fractionation method that uses millisecond-long nonlinear HIFU pulses with shock fronts. The BH treatment generates bubbles and tissue debris that move within the liquified lesion due to acoustic radiation force and streaming. Since the velocity of the debris is expected to depend on the lesion size and fractionation completeness, it could provide a quantitative metric of the treatment progression. In this study, the motion of bubble remnants and tissue debris immediately following BH pulses was investigated by high-pulse repetition frequency (PRF) plane-wave color Doppler ultrasound (PRF 9–15 kHz, 50–100 pulses) in *ex vivo* bovine myocardium and porcine liver. A 256-element 1.5 MHz spiral HIFU array with a coaxially integrated ultrasound imaging probe (ATL P4-2) produced the 10ms BH pulses to create a volumetric lesion in tissue. As tissue liquefaction progressed, the velocity was observed to be directed towards the HIFU transducer, monotonically increased, and then saturated at about 1 m/s. Further, when the lesion was fully liquefied and its axial size was comparable to that of the HIFU focal spot, the motion changed direction to the opposite, with velocity independent of tissue type, consistent with streaming. [Work supported by NIH R01EB7643, R01GM122859, R01EB025187, and RSF20-12-00145.]

8:50

2aBAa4. Boiling histotripsy dose for mechanical ablation of human prostate tissues with different elastic properties. Pavel Rosnitskiy (Phys. Faculty, Lomonosov Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, pavrosni@yandex.ru), Sergey A. Tsysar, Maria M. Karzova, Ekaterina M. Ponomarchuk, Anastasia V. Tyurina, Kseniya D. Tumanova (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Sergey V. Buravkov (Faculty of Fundamental Medicine, Lomonosov Moscow State Univ., Moscow, Russian Federation), Alexey V. Kadrev (Medical Res. and Educational Ctr., Lomonosov Moscow State Univ./Diagnostic Ultrasound Div., Russian Medical Acad. of Continuous Professional Education, Moscow, Russian Federation), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Univ. of Washington/Moscow State Univ., Moscow, Russian Federation), Adam Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Andrey Chernyaev (Pulmonology Res. Inst., Federal Biomedical Agency of Russia, Moscow, Russian Federation), and Vera Khokhlova (Univ. of Washington/Moscow State Univ., Moscow, Russian Federation)

Boiling histotripsy (BH) is a promising technology for transrectal mechanical ablation of localized prostate cancer. BH uses sequences of milliseconds-long pulses to generate high-amplitude shocks in acoustic pressure waveforms at the focus, resulting in non-thermal liquefaction of pathological tissue. Despite successful pilot experiments on BH ablation of fresh human prostate, a critical problem remains on determining the threshold values of the BH-dose (the number of pulses per focus for specific exposure protocols) sufficient for tissue ablation. This threshold depends on the stiffness of tissue, which varies in different patients and different prostate tissue types. Here, BH lesions were induced in *ex vivo* prostate tissue samples ($n = 14$) with different values of shear modulus and using different BH doses. The 2-MHz transducer operated with exposure parameters: 10-ms pulses, 1% duty cycle, and 10–40 pulses/point. After sonication, the samples were analyzed histologically to assess cell destruction. The experiment demonstrated varying completeness of tissue liquefaction for various BH-dose depending on the tissue stiffness. [Work supported by RSF 21-72-00067.]

9:05

2aBAa5. Low-intensity pulsed ultrasound induces metabolic lipolysis of adipocytes. Sangnam Kim (Aerosp. and Mech. Eng., Univ. of Notre Dame, 213B Multidisciplinary Res. Bldg., Notre Dame, IN 46556, skim52@nd.edu) and Sangpil Yoon (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN)

We used low-intensity pulsed ultrasound (LIPUS) to uncover underlying lipolysis mechanism of 3T3-L1 adipocyte. Ultrasound pulses with the center frequency of 2 MHz were applied for 10 min/day to promote the lipolysis of white adipocyte. Lipolysis is the process of breaking down lipids and the metabolic pathway through which lipid triglycerides are hydrolyzed into a glycerol and three fatty acids. The proliferation of 3T3-L1 adipocytes was detected by cell viability XXT assay, RT-PCR and Western blot. The Triglyceride Assay quantitatively measures triglycerides levels in differentiated adipocytes. LIPUS reduced the triglycerol content significantly in adipocytes. The differences between transcriptional genes and metabolites were analyzed by transcript analysis and metabolomic profiling experiments. The results of RT-PCR, and Western blot after LIPUS showed the increased lipolysis. The expression of genes related to lipolysis-related factors, ATGL and HSL, was up-regulated. In addition, cellular RNA-sequencing (RNA-Seq) showed an increase in lipolysis and adaptive immune cells under ultrasound stimulation. The algorithm and additional analysis reveals a gene network that is predicted to be associated and potentially drive adipocyte lipolysis and heterogeneity. LIPUS can promote the lipolytic capacity of 3T3-L1 cells in the differentiated state. This study provides an important experimental and theoretical basis for the clinical application of LIPUS in promoting the lipolysis of 3T3-L1 adipocytes.

9:20

2aBAa6. Suppressing tissue clutter in the presence of motion and nonlinear propagation of ultrasound. Geraldi Wahyulaksana (Biomedical Eng., Erasmus MC, Wytemaweg 80, Rotterdam, Zuid Holland 3015CN, The Netherlands, g.wahyulaksana@erasmusmc.nl), Luxi Wei, Jasper Schoormans, Antonius F. van der Steen, and Hendrik Vos (Biomedical Eng., Erasmus MC, Rotterdam, The Netherlands)

Visualizing blood flow in small vessel with ultrasound imaging is a useful diagnosis tool for evaluating organ condition. The usage of encapsulated microbubbles that have non-linear response as contrast agents, in combination with pulsing schemes that suppress the linear tissue signal can improve the sensitivity and specificity of ultrasound flow imaging. However, the nonlinear propagation when ultrasound travels through tissue and microbubble clouds reduce contrast detection. Moreover, tissue motion causes clutter artifacts and imprecise combination of the multi-pulse transmission scheme, impairing image quality. We have previously developed an independent component analysis (ICA)-based clutter filter for high framerate imaging, which performed better than the widely used singular value decomposition (SVD) filter in presence of fast tissue motion. In this study, we examined the performance of ICA filter in combination with power modulation (PM) pulsing schemes to detect flow signal. We performed an *in vitro-* measurement that emulated ultrasound propagation through cloud of microbubbles and fast tissue motion. The results showed that the combination of PM and ICA filter achieved up to 6dB contrast-to-background ratio improvement compared to only PM. This improvement could be significant for myocardial perfusion imaging, where rapid tissue motion and nonlinear propagation through clouds of microbubbles are expected.

9:35–10:05 Break

10:05

2aBAa7. Characterization of nonlinear field of a dual-mode linear array for image-guided applications of pulsed high-intensity focused ultrasound. Vera A. Khokhlova (CIMU/APL, Univ. of Washington, Seattle, va.khokhlova@gmail.com), Petr V. Yuldashev (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Randall P. Williams (CIMU/APL, Univ. of Washington, Seattle, WA), Maria M. Karzova, Azamat Z. Kaloev, Fedor A. Nartov (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), and Tatiana D. Khokhlova (Div. of Gastroenterology, Univ. of Washington School of Medicine, Seattle, WA)

Recent studies have shown that nonlinear pulsed high-intensity focused ultrasound (pHIFU) is capable of inducing cavitation, without the need for contrast agents, to enhance drug delivery in tissues with poor perfusion. Here, acoustic characterization of a dual-mode linear array designed for image-guided cavitation-enhanced drug delivery applications is performed. The array consists of 64 elements (1.07 MHz, 14.8×51.2 mm^2, 0.8 mm pitch) and is controlled by an ultrasound system (V1 Verasonics) configured with an external power supply for pHIFU excitation. Acoustic holography measurement data were collected for varying focus position electronically steered in the transverse and axial directions and used for setting a boundary condition in the acoustic model. Nonlinear simulations of the array field are performed using the Westervelt equation at increasing power outputs of the array. Modelling results are validated by comparing with pressure waveforms measured in the focus using a fiber optic hydrophone. It is shown that the amplitude of the shock fronts formed at the focus increases for larger focusing angles of the field and is sufficient to maintain the therapeutic level of cavitation. Simulation results are in good agreement with the measurement data. [Work supported by NIH R01EB023910 and RSF 20-12-00145.]

10:20

2aBAa8. Dependence of cavitation behavior induced by a pulsed high-intensity focused ultrasound array on focal waveform parameters and transducer F-number. Randall P. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA, rpw1@uw.edu), Maria M. Karzova, Petr V. Yuldashev (Lomonosov Moscow State Univ., Moscow, Russian Federation, Moscow, Russian Federation), Vera A. Khokhlova (Lomonosov Moscow State Univ., Moscow, Russian Federation, Seattle, WA), Bryan W. Cunitz (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Kyle P. Morrison (Sonic Concepts, Inc., Bothell, WA), and Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA)

Pulsed high-intensity focused ultrasound (pHIFU) is capable of inducing cavitation without the need for contrast agents, which can enhance drug transport in tissues with poor perfusion. We report on the experimental characterization of cavitation induced by a new dual-mode ultrasound array designed for image-guided pHIFU therapies. The 64-element array (1.071 MHz, aperture of 14.8×51.2 mm^2 and pitch of 0.8 mm) is driven by the Verasonics V1 ultrasound system, configured for pHIFU excitation. Acoustic properties of the beam in the focal region were characterized through hydrophone measurements in the linear and nonlinear operating regimes while steering the beam azimuthally and axially. A high-speed camera was used to observe the cavitation behaviors induced in tissue-mimicking gel phantoms near the focus of the beam for different pHIFU exposure parameters. As the focal pressure increases, a change in cavitation behavior occurs, from the appearance of single, stationary bubbles, to groups of proliferating bubbles. The pressure at which the transition from stationary to proliferating cavitation was observed corresponds to the onset of shocks in the focal region. Beams with larger F-numbers were shown to induce cavitation at lower pressures than beams with lower F-numbers. [Work supported by NIH R01EB023910 and RSF 20-12-00145.]

10:35

2aBAa9. Aberration-inducing body wall phantom for high-intensity focused ultrasound applications. Alex T. Peek (Appl. Phyiscs Lab., Univ. of Washington, Seattle, WA, apeek@uw.edu), Gilles P. Thomas (Appl. Phyiscs Lab., Univ. of Washington, Seattle, WA), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Pavel Rosnitskiy, Petr V. Yuldashev (Lomonosov Moscow State Univ., Moscow, Russian Federation, Moscow, Russian Federation), Daniel Leotta (Appl. Phyiscs Lab., Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Univ. of Washington/Moscow State Univ., Moscow, Russian Federation)

Aberrations induced by soft tissue inhomogeneities distort acoustic fields and often complicate HIFU therapies. An aberration-generating phantom was designed and fabricated to mimic the acoustic effects of a porcine body wall. A rippled surface pattern replicating inhomogeneities of size scales relevant to the porcine body wall was transferred to a large phantom made of fat-like and muscle-like gel materials. The acoustic distortions produced by the phantom and an *ex vivo* porcine body wall were characterized by hydrophone measurements with a 256-element 1.5 MHz focused ultrasound array (diameter = 144 mm, F# = 0.83). Insertion of the phantom generated large distortions for both linear and nonlinear beam focusing including displacement of the focus (up to 0.3 mm transversely), enlargement of the focal region, and reduction in focal pressures due to aberration (up to 2.5-fold peak positive pressure reduction). The phantom and porcine body wall produced similar levels of aberration and attenuation effects. An aberration correction approach was leveraged to isolate the quantitative effect of aberration and to restore shocks in nonlinear waveforms in the focus. Boiling histotripsy treatments through the phantom or body wall were subsequently demonstrated in a hematoma sample. [This work was supported by NIH R01EB7643 and R01EB25187, and RSF 20-12-00145.]

10:50

2aBAa10. *In vivo* phase aberration correction for high intensity focused ultrasound therapy with a 256-element spiral array. Gilles P. Thomas (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, gthom@uw.edu), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Yak-Nam Wang, Stephanie Totten (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), George R. Schade (Dept. of Medicine, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Univ. of Washington/Moscow State Univ., Seattle, WA), and Vera A. Khokhlova (Univ. of Washington/Moscow State Univ., Moscow, Russian Federation)

One of the challenges of transcutaneous high intensity focused ultrasound therapies, especially ones relying on shock formation such as boiling histotripsy (BH), is the deterioration of focusing from phase aberration induced by soft tissue heterogeneities within the acoustic path. Here, a methodology to perform phase aberration correction *in vivo* is proposed. A custom BH system consisting of a 1.5 MHz phased array of 256 elements connected to Verasonics V1 system was used in pulse/echo mode targeting liver and kidneys in swine. Phase shift on each element needed to correct for aberration was estimated by maximizing the value of the coherence factor on the backscattered signals received from the focal region and iteratively repeating the pulse emission with corrected phases. The process required 3 to 10 iterations to converge to a solution, which necessitated the implementation of tracking the moving target via additional pulse/echo sequences. The performance of the method was evaluated by comparing the transducer driving voltage needed for shock-forming conditions to generate a boiling bubble at the target with and without aberration correction. Results show that aberration correction effectively lowers the voltage required to reach boiling by 8%–25%. [Work supported by NIH R01EB007643, R01GM122859, R01EB25187, P01DK043881, and RSF 20-12-00145.]

11:05

2aBAa11. Directional dependence of nonlinear piezoelectric surface acoustic waves in lithium niobate. John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Dept. of Medicine, Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA 15261, jmc345@pitt.edu) and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

A model for the propagation of finite-amplitude surface acoustic waves (SAWs) in a piezoelectric material [Hamilton *et al*., J. Acoust. Soc. Am. **100**, 2567(A) (1996)] is used to investigate the dependence of SAW nonlinearity on crystal cut and propagation direction in lithium niobate ($LiNbO_3$). The model is based on Hamiltonian formalism, in which the boundary conditions at the free surface need only be evaluated at linear order, thus simplifying computation of the nonlinear problem. In contrast with the scalar coefficient of nonlinearity that appears in evolution equations for bulk plane waves, nonlinearity of SAWs is described by a matrix for harmonic interaction due to the variation of depth dependence of the SAW field with frequency. Components of the nonlinearity matrix are generally complex-valued as a result of material anisotropy and piezoelectricity. Variation with propagation direction of the (1,1) component of the nonlinearity matrix, which governs second-harmonic generation, is presented for $LiNbO_3$ crystals that are X-cut, Y-cut, and Z-cut. Relative contribution of each nonlinear mechanism (nonlinear elasticity, nonlinear piezoelectricity, electrostriction, and nonlinear dielectricity) to the nonlinearity matrix is examined. Distortion of an initially sinusoidal waveform during propagation is demonstrated by integrating coupled spectral evolution equations for the harmonic amplitudes.

11:20

2aBAa12. Demonstration of nonlinear interaction of crossed turbulent streaming jets. Jenna M. Cartron (Phys., U.S. Naval Acad., 3519 Forest Haven Dr., Laurel, MD 20724, jcartron11@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

An experimental apparatus was built for demonstration purposes involving two mutually perpendicular crossed ultrasonic turbulent streaming jets in water. A 2D Particle Imaging Velocimeter (LaVision) measured the turbulent flow generated by their interaction. Experiments were performed in an open 10 cc acrylic water tank. The apparatus consisted of two Langevin ultrasonic transducers—both equipped with stepped horns with section diameters: 1/2, 1/4 and 1/8 in.[2]. A kit developed by Sonics and Materials included a 40-kHz oscillator driving a 20 W amplifier (impedance matched to the 40 kHz transducer CV245) and stepped horn (Part # 630-0594). Two such units were needed to generate the crossed ultrasonic turbulent streaming jets. In the viewing plane the axes of each streaming jet was—**i** cos30—**j** sin30, + **i** cos30—**j** sin30, respectively. The horn tips (1/8 inch) (slightly submerged in the water) also generated small cavitation bubbles. Lighthill's theory of acoustic streaming (1978) for a single source will be discussed. Lighthill recognized the importance of "Stuart streaming," Reichardt's turbulent jet measurements, Schlicting's turbulent jet theory (involving eddy viscosity) and Squire's exact solution for a nonlinear laminar jet flow. Earlier, Lighthill predicted that a turbulent jet would generate aerodynamic sound.

11:35

2aBAa13. Nonlinear scattering of crossed ultrasonic beams in the presence of a turbulent bubbly jet flow in water. Katherine A. Haas (Biology, St. Mary's College of Maryland, 1106 Broadview Dr., Annapolis, MD 21409, kahaas@smcm.edu) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

Experiments involve the nonlinear scattering of mutually perpendicular crossed ultrasonic beams of primary CW frequency components: $f_1 = 2.240$ MHz, $f_2 = 2.293$ MHz; interacting nonlinearly with turbulent bubbly shear flow generated by a circular water jet (nozzle exit diameter D = 1.6 mm and volume flow rate 32 cc/s). The jet axis is perpendicular to the crossed beam axes. The nozzle exit is downward (5 mm above the surface). The water pump's suction port was at the bottom center of the 10 cm cubic acrylic water tank (wall thickness 4.8 mm). Identical 1.27 cm diameter quartz transducers (T_1 and T_2) were epoxied at the centers of adjacent vertical tank sidewalls generate the primary beams (in the **i** or **j** directions). Identical 4.5 MHz receiving transducers (6.35 mm diam PZT-4) measure nonlinear scattering in the forward or backscattered directions. Receiver R_f located in the plane formed by the primary acoustic beam axes) detects in the **i** cos 45 + **j** sin 45 direction. Receiver R_b was located in the opposite direction. In the absence of turbulence virtually no radiated sum frequency component exists. Doppler shift, frequency broadening, skewness and kurtosis from the received sum frequency power spectrum were measured.

Session 2aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics III

Sangpil Yoon, Chair
Aerospace and Mechanical Engineering, University of Notre Dame, 151 Multidisciplinary Research Building, Notre Dame, IN 46556-4634

Contributed Papers

9:00

2aBAb1. A focused limited-diffraction beam for high-resolution imaging. Jian-yu Lu (Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606, jian-yu.lu@ieee.org)

Unfocused limited-diffraction beams such as Bessel beams and X waves have been studied extensively to obtain a large depth of field for medical imaging. In this study, a focused zeroth-order Bessel beam $j_0(\alpha r)$, where $\alpha = 1202.45$ m^{-1} and r is the radius, produced with a 50 mm diameter and 2.5-MHz transducer and with an acoustic lens of 50-mm focal length (*f*-number = 1) was used to obtain a C-mode image of an object in water (0.6 mm wavelength). The results of the study showed that at the focal distance, the lateral resolution of the image obtained with an unapodized transducer in both transmission and reception was lower than that obtained with an unapodized beam in transmit but the j_0 Bessel beam was used in receive. Beam plots of the unapodized beam and the j_0 Bessel beam at the focal distance showed that the lateral beamwidths were about 0.78 and 0.59 mm, respectively. The 0.59-mm beamwidth of the focused Bessel beam is better than that of the diffraction limit ($1.22 \times$ wavelength $\times$ *f*-number = 0.732 mm). The decreased beamwidth of the j_0 Bessel beam results in a higher lateral image resolution while the sidelobes of the beam were suppressed by the focused unapodized transmit beam.

9:15

2aBAb2. Ultrasound coherent plane-wave compound imaging: Image quality evaluation in phantoms and breast lesions. Gijs Hendriks (Medical UltraSound Imaging Ctr., Radboud Univ. Med. Ctr., P.O. Box 9101, Nijmegen 6500HB, The Netherlands, gijs.hendriks@radboudumc.nl), Gert Weijers, Chuan Chen (Medical UltraSound Imaging Ctr., Radboud Univ. medical Ctr., Nijmegen, The Netherlands), Madeleine Hertel (Siemens Heathcare GmbH, Forchheim, Germany), Chi-Yin Lee (Siemens Ultrasound, Issaquah, WA), Peter Dueppenbecker, Marcus Radicke (Siemens Heathcare GmbH, Forchheim, Germany), Andy Milkowski (Siemens Ultrasound, Issaquah, WA), Hendrik Hansen, and Chris L. de Korte (Medical UltraSound Imaging Ctr., Radboud Univ. Med. Ctr., Nijmegen, The Netherlands)

Coherent plane-wave compound imaging (CPWCI) can achieve high temporal resolution (e.g., to prevent breathing artefacts in volumetric scanning). It has to be verified that CPWCI using a limited number of steering angles (n_a) can achieve at least similar image quality as conventional focused imaging (CFI). The aim is to compare image quality between CPWCI and CFI in phantom and in breast lesions. In CPWCI, plane-wave channel data were recorded ($n_a = 15$, ± 11 deg) by a Sequoia Ultrasound system (10L4, 14L5; Siemens Healthineers); beamformed (delay-and-sum (DAS), Lu's-fk and Stolt's-fk) for each steering angle, and coherently compounded. For comparison, images were recorded by CFI (10, 35 mm focus). Image quality metrics were obtained in images of a multipurpose phantom (CIRS Model057). We have just initiated a reader study ($n = 200$) to investigate how these phantom results translate to *in vivo*. Phantom results showed that contrast sensitivity (CS) and resolution (CR), lateral resolution (LR) and contrast-to-noise ratio (CNR, depth < 45 mm) were similar for CPWCI and CFI (10, 35mm) for both transducers, whereas CNR (>45 mm), LR and penetration were improved. In CPWCI, Lu's f-k and DAS resulted in optimal CS (10L4) and LR, and CNR, respectively. In the ongoing reader study, image quality of breast lesions will be evaluated.

9:30

2aBAb3. Improving ultrasonic evaluation of osteochondritis dissecans using a cadaveric model. 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), Kun-Hui Chen (Dept. of Orthopedics, Mayo Clinic, Rochester, MN), Hyungkyi Lee, Shuai Leng (Dept. of Radiology, Mayo Clinic, Rochester, MN), James Fitzsimmons, Shawn O'Driscoll (Dept. of Orthopedics, Mayo Clinic, Rochester, MN), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Osteochondritis dissecans (OCD) is a joint disease that is prevalent among youth athletes. Medical ultrasound is currently limited in its ability to detect and monitor OCD due to limitations of current Delay-and-Sum (DAS) reconstruction algorithms. In this work, we used a Delay-Multiply-and-Sum (DMAS) reconstruction algorithm to improve medical ultrasound's ability to diagnose OCD. We artificially generated OCD lesions in unfixed cadaveric elbows and imaged them with a research ultrasound machine. We also imaged these cadaveric limbs with x-ray computed tomography (CT) with a high-resolution acquisition. These CT images were used as a gold standard for OCD geometry. We reconstructed the ultrasound images using both DAS and DMAS algorithms and compared them quantitatively by taking several profiles across the images of the artificial OCD lesions. The image metrics that we measured within these profiles were OCD lesion contrast, bone interface clarity, and lesion crack thickness accuracy. We found that the DMAS algorithm improved OCD lesion contrast by up to 26% when compared to DAS. We also found that the DMAS algorithm reduced the variability of bone interface clarity overall. Future work will involve an *in vivo* study, as well as optimization, of the DMAS algorithm for OCD lesion imaging.

9:45

2aBAb4. Z-section and 3D rendering imaging using 130 and 50 MHz ultrasound and bacteria generated ultrasound contrast agents. Sangnam Kim (Aerosp. and Mech. Eng., Univ. of Notre Dame, 213B Multidisciplinary Res. Bldg., Notre Dame, IN 46556, skim52@nd.edu), Gyoyeon Hwang, and Sangpil Yoon (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN)

We present an ultrasound z-sectioning imaging approach to visualize small vasculature. The major difference between confocal and conventional microscopy is the sectioning capability to generate three-dimensional images by stacking multiple sections. To visualize small vessels, we used 130 and 50 MHz ultrasonic transducers and gas vesicles (GV) isolated from bacteria as contrast agents. We developed 130 and 50 MHz ultrasonic transducers using lithium niobate. The focus and aperture of 130 MHz transducer are 1.5 and 1 mm and 50 MHz transducer has 3 mm aperture and 4 mm

focus. These transducers were attached to ultrasound biomicroscopy for scanning phantoms and mouse liver specimen *ex vivo* with 1 micrometer steps. Saved radiofrequency data was post-process to reconstruct B-mode images and mid-band fit (MBF) maps. Four types of GVs were isolated in house. GV sizes were ranging from 200 to 600 nm. Three GV species present unclustered morphology but GVs isolated from one bacteria species present clustered morphology. Distinct MBF maps were created from clustered and unclustered GVs which were suitable for multiplexed imaging. The results demonstrated that MBF and GV based ultrasound imaging approach can image thin slices of tissue specimen with multiplexed imaging capability.

10:00

2aBAb5. Color Doppler imaging of pure crystals in hyperbaric conditions. Eric Rokni (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, ezr144@psu.edu) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The color Doppler ultrasound twinkling artifact is a rapid color shift that appears on some kidney stones and other pathological mineralizations during ultrasound imaging. Twinkling on kidney stones is primarily attributed to scattering from stabilized microbubbles as increasing hydrostatic pressures has been shown to decrease twinkling. However, it is unknown whether microbubbles are also present and cause twinkling on other forms of pathological mineralizations or pure crystals. In this study, 5 cholesterol and 5 calcium phosphate crystals were grown *in vitro* from supersaturated solutions and imaged in a custom-built, hydrostatic pressure chamber with a Philips/ATL L7-4 transducer and Vantage-128 research ultrasound system. Hydrostatic pressure was increased to a maximum of 14 MPa while saving IQ data at 1 fps for evaluation of Doppler power. On all 10 imaged crystals, twinkling was found to decrease by approximately 90% when pressures reached 4 MPa absolute. Twinkling reappeared when the pressure was returned to atmospheric levels. These results suggest that bubbles are present on pure crystals and give rise to the twinkling artifact. [Work supported by NSF CAREER 1943937 and NSF GRFP DGE1255832.]

10:15

2aBAb6. Comparison of the relative performance of three ultrasonic backscatter parameters measured *in vivo* at the femoral neck. Brent K. Hoffmeister (Rhodes College, 2000 N Parkway, Memphis, TN 38112, Memphis, TN 38112-1624, hoffmeister@rhodes.edu), Sarah I. Delahunt (Rhodes College, Austin, TX), Kiera L. Downey, Ann M. Viano, Will R. Newman, Doni M. Thomas, Loukas A. Georgiou (Rhodes College, Memphis, TN), Aubrey J. Gray (Rhodes College, Frederick, MD), Evan N. Main, and Gia Pirro (Rhodes College, Memphis, TN)

The global impact of osteoporosis as a major public health problem has generated interest in developing ultrasonic techniques that can be used to screen populations for this bone disease. The goal of this study was to assess the relative performance of three ultrasonic backscatter parameters: apparent integrated backscatter (AIB), frequency slope of apparent backscatter (FSAB) and frequency intercept of apparent backscatter (FIAB). Measurements were performed at the left and right femoral necks of 88 healthy volunteers using an ultrasonic imaging system equipped with a 3 MHz convex multi-element transducer. Backscatter signals from the femoral neck were captured for analysis. AIB was determined from the frequency averaged power in the backscatter signal compensated for the frequency dependent response of the measurement system. FSAB and FIAB were determined from the slope and intercept, respectively, of a line fitted to the compensated spectrum. Linear regression analysis was used to compare measurements performed at the left and right femur. All three parameters demonstrated similar and highly significant ($p < 0.000001$) correlations between left and right side measurements ($R_{AIB} = 0.62$, $R_{FSAB} = 0.56$, $R_{FIAB} = 0.51$) indicating that they are equally sensitive to naturally occurring variations in the ultrasonic properties of the femoral neck.

10:30–10:45 Break

10:45

2aBAb7. Foam gratings as an alternative to customised acoustic lenses. Luke A. Richards (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus, Oxford, Oxfordshire OX3 7DQ, United Kingdom, luke.richards@eng.ox.ac.uk), Eleanor P. Stride, and Robin O. Cleveland (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Tissue inhomogeneity can affect the accuracy with which ultrasound can be focused into the body. In existing clinical systems, phase corrections are typically made using ultrasound arrays, but with the advent of 3D printers there has been increased interest in producing patient-specific acoustic lenses. Acoustic lenses offer a more compact and affordable system than an equivalent array and may be suited for therapeutic applications where there is well defined target location. This study explored the properties of gratings made of foam machined with holes as an alternative to acoustic lenses. The approach was investigated with an analytical model, a full-wave numerical model, and experimental measurements. A grating is demonstrated that mimics a conventional ultrasound lens, modulating the phase of transmitted ultrasound while maintaining uniform amplitude. The performance of a foam grating is compared to lenses made of PDMS or 3D printed resin. Using two gratings, independent control of amplitude and phase is demonstrated, with increased insertion loss. The primary advantages of this technique over conventional lenses are rapid manufacture (<30 min) and the ability to control both the amplitude and phase of the transmitted ultrasound. Potential applications include generation of complex ultrasound fields for patient specific treatments in ultrasound therapy.

11:00

2aBAb8. Detection of cellular mineralization using the Doppler ultrasound twinkling artifact. Lucas Ruge-Jones (Penn State Univ., 201E Appl. Sci. Bldg., University Park, PA 16802, ldr5225@psu.edu), Lisa Berntsen, Daniel Hayes, and Julianna Simon (Penn State Univ., University Park, PA)

Heterotopic ossification (HO), or the growth of bone in soft tissue where bone does not usually exist, can occur after any musculoskeletal trauma. While the exact cause of HO is unknown, it can take 2i–3 months before HO can be detected on x-ray or computed tomography (CT). The color ultrasound Doppler twinkling artifact is a rapid color shift that was discovered when imaging kidney stones. Here, our objective is to determine whether twinkling can be used to detect the earliest signs of cellular mineralization in HO. A Vantage-128 research ultrasound system with Philips/ATL L7-4, L12-5, and Vermon L22-14 transducers was used to image human bone marrow-derived stem cells (hBMSCs) cultured in osteogenic media and supersaturated solutions of calcium phosphate. Preliminary results show a 50% increase in Doppler magnitude and a 250% increase in Doppler variance in the presence of cellular mineralization or calcium phosphate crystals compared to images of cells without minerals or in non-crystalline solutions. The L7-4 was found to be more sensitive to twinkling from mineralization than the higher frequency transducers. The increase in Doppler magnitude and variance with cellular mineralization suggests twinkling may be a sensitive early detector of HO. [Work supported by DoD CDMRP PR201164.]

11:15

2aBAb9. Experimental comparison of acoustic backscattered field from a single fiber and media with pseudo-aligned fibers in longitudinal and transverse planes. Mohammadreza Kari (Medical Phys., UW Madison, 1122-F2 WIMR, 1111 Highland Ave., Madison, WI 53705, mkari@wisc.edu), Ivan Rosado-Mendez (Medical Phys., UW Madison, Mexico City, Mexico City, Mexico), Helen Feltovich (Medical Phys., UW Madison, Provo, UT), and Timothy J. Hall (Univ. of Wisconsin, Madison, Madison, WI)

Acoustic backscatter spectrum analyses of soft tissues are being developed to provide quantitative information beyond that offered by B-mode images. Common assumptions used in some approaches are that the medium is homogeneous, isotropic, and produces diffuse scattering conditions. However, in some soft tissues, the presence of fiber-like structures causes the

acoustic backscatter to be anisotropic. In this work, we compare the backscattered fields produced by a single fiber, a phantom containing many pseudo-aligned fibers, and biceps muscle. Measurements were performed with single-element immersion transducers using a laboratory benchtop pulse-echo setup, and different linear array transducers on a Siemens Acuson S3000 scanner. The phantom and muscle were imaged in longitudinal and transverse planes to study the effect of fiber orientation on the backscattered field. The results showed that the normalized backscattered power is different in longitudinal and transverse planes. This can result from differences in the sonicated portion of the fibers when varying their orientation with respect to the 3D diffraction pattern. Future work will further study this phenomenon with single fibers and transducers with different focal configurations, and investigate the use of the point-wise signal-to-noise ratio and the generalized spectrum to detect the presence of coherent scattering conditions.

11:30

2aBAb10. Measuring signal quality in low power wearable ultrasound imaging. Samuel A. Acuña (Bioengineering, George Mason Univ., 4400 University Dr., Ste. 3100, Fairfax, VA 22030, sacuna2@gmu.edu), Ahmed Bashatah, Parag V. Chitnis, and Siddhartha Sikdar (Bioengineering, George Mason Univ., Fairfax, VA)

Frequency Modulated Continuous Wave (FMCW) ultrasound is an emerging imaging method adapted from radar that can be implemented in wearable ultrasound systems due to the reduced size, power, and signal processing requirements. Acoustic reflections are encoded in the frequency domain, and thus evaluating signal quality currently requires extracting frequency information from a time-varying acquisition signal, which requires additional signal processing. Question: Can we differentiate FMCW ultrasound signal quality using an acquired FMCW time-series? We collected FMCW ultrasound images with a custom ultrasound transducer contacting and losing contact with tissue, to simulate moments of poor signal quality. We used two potential time-domain signal quality metrics: the mean absolute value (MAV) of the signal, and Sample Entropy. Sample Entropy is an information theory method for assessing complexity in a time-series. Both MAV and Sample Entropy could differentiate when the transducer was in good contact with the tissue. MAV required less processing time, but Sample Entropy had less noise when the transducer lost contact. FMCW ultrasound imaging is a promising imaging method that has application in wearable ultrasound systems, and FMCW ultrasound signal quality can be assessed in the time-domain without extracting frequency information. [Work supported by NIH 5U01EB027601-02.]

11:45

2aBAb11. Haze Artifact reduction in cardiovascular synthetic aperture imaging in use of intertransmission coherence factor. Teiichiro Ikeda (Innovative Technol. Lab., FUJIFILM Healthcare Corp., 1-280, Higashi-Koigakubo, Kokugunji-shi, Tokyo 1858601, Japan, teiichiro.ikeda.hv@fujifilm.com), Chizue Ishihara (I), Misaki Hiroshima (Innovative Technol. Lab., FUJIFILM Healthcare Corp., Kokubunji-shi, Tokyo, Japan), Masnori Hisatsu, and Hiroshi Kuribara (Ultrasound Diagnostic Systems Div., FUJIFILM Healthcare Corp., Tokyo, Japan)

In cardiovascular medical ultrasound applications, the use of spatially decimated focused-transmission synthetic aperture (SA) imaging is promising to achieve both high SNR and high-frame-rate. But unlike prevailing planar/divergence SA, the use of focused transmission where the virtual source of SA is placed in the imaging area for the better penetration with up to 160-mm field-of-view result in severe haze artifacts in the heart chambers. We hypothesize the haze is caused by the grating lobe due to the decimated focused transmission of SA and propose a method for reducing artifacts by using the intertransmission coherence factor (ITCF) [1] focusing on the fact that SA grating lobe signal is less coherent than the main lobe signals. Simulation and *in vivo* experiment revealed that there is a strong relation between the grating lobe and haze artifact. When imaging a human heart (under approval of institutional review board) using 4-times transmission decimation (equivalent to 160 frames/s) with a phased array probe, the ITCF method was able to reduce the grating artifacts within heart chambers greatly by more than 15 dB, while maintaining the intensity of myocardium, LV/LA wall, mitral valve, and other tissue/organ structures. [1] Ikeda *et al*. IEEE Trans. UFFC, doi: 10.1109/TUFFC.2021.3088678 (2021).

Session 2aCA

Computational Acoustics and Signal Processing in Acoustics: Validation and Verification (V&V) for Acoustics and Vibrations Simulations

Sheri Martinelli, Cochair
Applied Research Laboratory, The Pennsylvania State University, 225 Science Park Rd/, M/S 9800L, State College, PA 16803

Andrew S. Wixom, Cochair
Applied Research Laboratory, Pennsylvania State University, P.O Box 30, Mail Stop 3220B, State College, PA 16801

Chair's Introduction—8:00

Invited Paper

8:05

2aCA1. Model validation in the test and evaluation community: Current practices. Edward Pees (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, edward.pees@navy.mil)

Computer modeling and simulation (M&S) continues to play a growing role in the acquisition lifecycle of commercial systems throughout every stage from developmental testing to operational evaluation. During system development, M&S may be used to evaluate options for meeting performance requirements. For operational systems, M&S may be used to enable the design of physical experiments aimed at assessing operational effectiveness, suitability, or durability. When performing actual system testing, M&S is often useful for reinforcing conclusions drawn from experimental measurements by supplementing test results, or simulating test cases that are experimentally inaccessible. Given the high value of decisions that may be based on the outcomes of these and other types of simulations, the need for a rigorous verification and validation (V&V) process becomes evident. This presentation provides an overview of the basic design and analysis techniques that are being promoted by the operational testing community to perform quantitative comparisons of simulation and experimental data. As advances in computing technology lead to more extensive use of M&S for supporting system assessments, stakeholders will require more extensive use of defensible validation practices. The focus of this presentation, therefore, is to provide an overview of the process, required data, and statistical techniques recommended for validating complex acoustic models.

Contributed Paper

8:25

2aCA2. Validation of a sonic boom model through atmospheric turbulence with flight test data. 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)

This paper presents a validation of an extended Burgers equation that accounts for mean scattering effects of turbulence in the atmospheric boundary layer (ABL). The scattering effects are calculated from a scattering wavenumber formulated using the multiple scattering theory and are incorporated as an additional linear term in a Burgers equation commonly used to model sonic boom propagation. The scattering wavenumber represents the interaction of an acoustic wavefield, including backscattering, with a turbulent field and gives rise to signal attenuation and distortion. Numerical models obtained using the extended Burgers equation are compared to measured sonic booms from conventional supersonic aircraft. The acoustic measurements are supplemented by measurements of the turbulence conditions as well as the thickness of the ABL. Results show that the mean effects of scattering on sonic booms smooth out shocks in the waveforms, reduce the peak amplitudes, and thus reduce the loudness. The results also indicate that the loudness reduction is dependent on the variance, correlation length, and thickness of the turbulent field. The performance of the extended Burgers equation is compared to that of existing signal processing-based approaches developed to represent turbulence effects on sonic booms through the ABL.

Invited Paper

8:40

2aCA3. Focus boom simulation benchmark cases for the 182nd ASA Meeting May 2022. Sriram Rallabhandi (Aeronautics Systems Anal. Branch, NASA Langley Res. Ctr., Rm. 190-25, Mailstop 442, Hampton, VA 23681, sriram.rallabhandi@nasa.gov), Juliet A. Page (US Dept. of Transportation, Cambridge, MA), and Gerald Carrier (ONERA, Palaiseau, France)

The low-boom prediction committee comprised of several research agencies and companies across the world plans to organize a special session dedicated to prediction and comparison of focused sonic booms. This presentation will document and discuss the benchmark cases that will be studied and compared during this special session. The case description would include the underlying near-field CFD data along with relevant flight conditions and mission trajectory information for the participants to simulate and discuss their individual focus boom predictions. Details of the CFD simulation conditions and solution procedure to obtain off body pressure waveforms on a cylindrical surface around the aircraft at a few selected points along the flight trajectory will be provided. The rationale for the choice(s) of the mission trajectories to result in focus booms near the ground for comparison using multiple independent implementations will be provided as well.

Contributed Papers

9:00

2aCA4. On mitigating the opacity of digital twins. Gregory A. Banyay (Appl. Res. Lab., Penn State Univ., University Park, State College, PA 16802, gregory.banyay@protonmail.com)

Digital twins, or variants thereof, are becoming ubiquitous across many disciplines of computational science and engineering modeling and simulation. Despite the increased use of this concept, industrial practitioners and academic researchers alike lack consensus as to what exactly constitutes a digital twin, much less how one might quantify associated uncertainties to support the verification, validation, and ultimately credibility assessment thereof. Furthermore, the complexities involved in modeling structural vibration and acoustics can exacerbate the epistemic uncertainties of this opaque portrait. We therefore posit that use of goal structured notation (GSN) in the context of a credibility assessment framework (CAF) provides one viable means by which key stakeholders in both the development and deployment of digital twins can collectively work towards increased clarity. One can expect that achieving greater clarity for requirements of digital twins, particularly those of phenomenological relevance to structural dynamics and acoustics, computational, experimental, and operational disciplines can more optimally combine to achieve desired engineering outcomes.

9:15

2aCA5. A Helmholtz equation implementation for finite element method based on FEniCS project. Roberto Velasco-Segura (UNAM, Inst. for Appl. Sci. and Technol., Circuito Exterior S/N, Ciudad Universitaria, Mexico City 04510, Mexico, roberto.velasco@icat.unam.mx) and Cristian U. Martínez-Lule (UNAM, Faculty of Sci., CDMX, Mexico)

Many systems in acoustics have relevant stationary states, which can be described using the Helmholtz equation. In this work we present an implementation of the Helmholtz for Finite Element Method based on FEniCS Project. Even when this is a single equation, the modeling of different systems requires different versions of it, regarding: domain of the involved variables, dimensionality, symmetry, boundary conditions, and homogeneity of the media (including parameters related to attenuation). Accordingly, these versions require different parts of the code to be adjusted. We present a set of examples, with increasing complexity to incorporate a variety of situations, and a convergence test against analytic solutions for some representative cases, which show some of the limitations of the code. Even when this code is presented with simple examples for clarity, its usage is not limited to this scope, it has successfully been used for research projects, which are briefly mentioned as well. The code is provided with the same open source license as FEniCS Project. [This work was supported by UNAM-PAPIIT TA100620.]

9:30

2aCA6. Uncertainty quantification of quadratically nonlinear local damage size quantification using nonlinear ultrasonics. Pravinkumar R. Ghodake (Mech. Eng., Indian Inst. of Technol., Bombay, IIT Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Backscattered and forward scattered harmonic waves from local damage modeled as quadratically nonlinear material in a 1D bar are used to solve a deterministic inverse problem to find the local damage size by Ghodake (2020). Due to highly nonlinear ($|\sin(x)/x|$) nature of the proposed forward problem, the constrained nonlinear deterministic optimization problem was solved using non-gradient-based algorithms. The forward model is a function of input frequencies, linear wave velocity, and local damage size. In practice, slight uncertainty of these parameters gives a completely different solution. To address this problem, uncertainty quantification of the modified forward problem is carried out along with solving the inverse problem using reliability-based design optimization (RBDO) and Bayesian Inverse (BI) approaches. In RBDO studies, Quantile Monte Carlo (QMC) approach gives more reliable and accurate results in comparison with approaches such as reliability index approach (RIA), performance measure approach (PMA), sequential optimization and reliability assessment (SORA), and inverse first-order reliability method (FORM) integrated with optimization methods such as interior-point (IP) and sequential programming (SQP). Multiple models are solved using the BI approach. Support vector machines regression (SVR), polynomial chaos expansions (PCE), and Kriging surrogate models are used along with RBDO and BI to reduce computational time.

9:45–10:00 Break

Invited Papers

10:00

2aCA7. Diffraction problems in an architectural acoustics benchmark. Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de) and Lukas Aspoeck (IHTA, RWTH Aachen Univ., Aachen, NRW, Germany)

In 2016/2017, the first round robin on auralization in room acoustics was performed. This project allowed participants to contribute results of their simulation model for typical scenarios of sound propagation and room acoustics. The outcome of the round robin and details of the related scene database were published in two papers [Brinkmann *et al*. J. Acoust. Soc. Am. 145 (4), 2746 (2019); Brinkmann *et al*., Appl. Acoust. **176**, 107867 (2021)]. As concerns diffraction effects, however, just a few integrated room-acoustic simulation tools could contribute to the comparison. In this paper, the reference measurements for the diffraction cases are described, and results obtained with simulation models are discussed in three categories: a finite screen on rigid ground, a finite panel in free field and an array of panels in free field.

10:20

2aCA8. Application of Monte-Carlo simulations for validating adaptive psychophysical procedures. Yi Shen (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246, shenyi@uw.edu)

In psychoacoustics, data collection from human listeners is commonly done by adaptively varying the stimulus parameters according to the data collected from preceding trials during an auditory task. Compared to the method of constant stimuli, the adaptive procedures allow the adjustment of stimuli for individual listeners to achieve a desired performance range. The adaptive procedures may be designed to estimate performance thresholds, psychometric functions, or the parameters of psychophysical models. The algorithms within these procedures that govern the selection of stimulus parameters may be based on heuristics or Bayesian active learning, and they may differ in terms of computational complexity, accuracy, reliability, stability, and rate of convergence. Choosing an appropriate adaptive procedure for a specific application can be assisted by Monte-Carlo simulations, in which candidate procedures are run on simulated listeners whose responses are governed by ground truth psychometric functions or psychophysical models. The estimated model parameters can be then compared to the ground-truth parameters to evaluate the performance of each adaptive procedure. In this presentation, an introduction of this approach will be provided with examples from psychoacoustics. The analysis techniques that address accuracy, reliability, stability, and rate of convergence will be discussed.

10:40

2aCA9. Shallow water transmission loss inversion for seafloor characterization. James Albritton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX) and Aaron M. Gunderson (Appl. Res. Labs., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, aaron.gunderson01@gmail.com)

A Bayesian inference algorithm has been developed for seafloor geoacoustic parameter determination through analysis of transmission loss measurements taken in shallow water environments. It uses a parallel tempering scheme for improved parameter space sampling and bias reduction, and evaluates ray acoustic forward models in Bellhop. The algorithm is verified by application to synthetic data generated using finite element and ray acoustic propagation models for frequencies from 0.5 to 5 kHz. Verification is assessed through the ability to recover inversion parameter values prescribed within the synthetic data, and inspection of ambiguous inversion situations. Initial inversion parameters include bulk sediment density, sound speed, and attenuation, as well as ripple height and wavelength for some seafloors. The inversion yields posterior probability distributions for each inversion parameter, which inform parameter resolvability and covariance. Algorithm accuracy, efficiency, and applicability to data from different ocean environments will be discussed. [This work has been supported by the Office of Naval Research, Task Force Ocean.]

Contributed Papers

11:00

2aCA10. Finite impedance boundary conditions determined *in situ* for modeling underwater acoustical measurement in laboratory tanks. Cameron T. Vongsawad (Phys. and Astronomy, Brigham Young Univ., 820 W 555 N, Orem, UT 84057, cvongsawad@byu.edu) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In underwater acoustics, laboratory tank sound propagation models may be improved using *in situ* characterization of water tank boundaries. An underwater acoustic tank model is developed with finite impedance boundaries determined *in situ* from impulse response measurements calculated from the frequency deconvolution of swept-sine signals following techniques commonly used in room acoustics. Impulse responses were used to obtain the acoustic absorption of the acrylic tank walls through reverse Schroeder integration determining the reverberation time and solving the Eyring equation for absorptive effects. This approach improves modeling of sound propagation in water tanks and is validated with numerical test cases and measurements. An improved tank model considering measured finite impedance boundaries will further develop the ability for more realistic data simulation in water tanks which will be used to design machine learning refinement.

11:15

2aCA11. Validating a phase-inversion procedure to assess the signal-to-noise ratios at the output of hearing aids with wide-dynamic-range compression. Donghyeon Yun (Speech and Hearing Sci., Indiana Univ. Bloomington, 1603 E. 3rd St. 216, Bloomington, IN 47401, dongyun@iu.edu), Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and Jennifer Lentz (Speech and Hearing Sci., Indiana Univ. Bloomington, Bloomington, IN)

Even for the same signal-to-noise ratio (SNR) at the input of a hearing aid, the output SNR may vary due to nonlinear compression algorithms. To quantify the output SNR of hearing aids, a phase-inversion technique is commonly used. The current study was designed to determine whether the phase-inversion technique accurately estimates the SNR for a simulated hearing aid. A simulated hearing aid was used to allow the explicit control of the compression algorithm and the calculation of the actual SNR at its output. The actual output SNRs were then used as the ground truth to evaluate the estimated output SNRs using the phase-inversion technique. Connected speech added to background noise (two- or twenty-talker babble), presented at SNRs of –10 to +10 dB, was used as test stimuli at the input of the simulated hearing aid. Four representative audiograms were used to set the gain and compression ratio for the simulated hearing aid according to a standard prescription method. In separate conditions, the hearing-aid compression was either fast- or slow-acting. Results showed that the estimated and actual output SNRs were not significantly different and their differences were within ±0.12 dB across all test conditions, indicating satisfactory agreement.

11:30

2aCA12. Using the lattice Boltzman method and GPU computing to model flow in organ pipes. Connor N. Kaplan (Rollins College, Winter Park, FL, ckaplan@rollins.edu), Jack D. Gabriel, Adrien David-Sivelle, and Whitney L. Coyle (Rollins College, Winter Park, FL)

The lattice Boltzman method (LBM) has been used to study oscillating air flow in musical instruments since the early 1990s. Most of the work has focused on flow into organ pipes and flow behavior around the instrument's labium. The current work intends to use GPU computing to simulate oscillations in 3D modeled and printed organ pipes of different cross-sectional geometries. These pipes will be used in experimental validation measurements of flow at the pipe exit (see TCMU general session for this work). The domain space required to visualize each outlet of air (around the labium and the pipe's open end) simultaneously with a sufficient resolution was larger than could fit in our GPU's memory. The resolution and space requirements of the simulations has required the use of high-memory GPUs (a physical machine as well as Google Cloud GPU have been tested). The physical and computational parameter optimization, GPU processes, and the simulation difficulties encountered will be discussed. Finally, preliminary comparisons between experiment and model will be shared.

11:45

2aCA13. Acoustic and thermal numerical modelling of transcranial ultrasound thermal therapy. Andrew Drainville (LabTAU INSERM 1032, 151 Cours Albert Thomas, Lyon 69003, France, andrew.drainville@inserm.fr), David Moore, John Snell (Focused Ultrasound Foundation, Charlottesville, VA), Sylvain Chatillon (LIST, CEA, Gif sur yvette, France), Frederic Padilla (Focused Ultrasound Foundation, Charlottesville, VA), and Cyril Lafon (LabTAU INSERM 1032, Lyon, France)

Ultrasound has long been recognized as a therapeutic tool with potential to treat disorders throughout the brain. However, the attenuation and distortion of acoustic waves caused by the skull has been a significant challenge to the development and clinical application of transcranial ultrasound therapies. Numerical models, such as the use of finite difference method, can be prohibitively expensive in required computational resources. While some numerical models can be computationally prohibitive, the ray-tracing modelling method used in the CIVA Healthcare simulation platform has been shown to be an efficient and computationally inexpensive method of modelling transcranial focused ultrasound, and can be coupled to Pennes' bioheat transfer equation to model temperature in biological tissue. The present work quantifies the sensitivity of transcranial ultrasound to clinically relevant uncertainties in the physical properties of the skull (i.e., shear conversion, wave speed), and combines the ray-tracing acoustic model with a numerical thermal model to provide a versatile and unified software platform for use in clinical ultrasound treatment planning. Results are compared with experimentally measured thermal rise from transcranial focused ultrasound using a clinical Insightec ExAblate 650 kHz 1000-element system, and shown to exhibit a high degree of accuracy. [This work is supported by the Focused Ultrasound Foundation.]

Session 2aEA

Engineering Acoustics: General Topics in Engineering Acoustics

Michael R. Haberman, Chair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Chair's Introduction—10:20

Contributed Papers

10:25

2aEA1. Electric vehicle additive sounds: Detection results from an outdoor test for sixteen participants. Michael Roan (ME, Penn State, 1430 Linn St., State College, PA 16803, mjr110@psu.edu), Luke Neurauter (VTTI, Blacksburg, VA), Michael Beard (Virginia Tech, Blacksburg, VA), and Marty Miller (VTTI, Blacksburg, VA)

Electric vehicles are significantly quieter than standard internal combustion engine vehicles. Although this is a benefit to the acoustic soundscape, it presents a safety concern, particularly to the vision impaired. Because of this, governments around the world have mandated that artificial sounds be added to electric vehicles in order to improve pedestrian safety. In the United States, these regulations are embodied in a National Highway Transportation Safety Administration document, FMVSS-141. This document provides guidance on the 1/3 octave-band frequency content, overall SPL, and measurement procedures that must be followed for a vehicle to be certified. This talk will present the results of a listener test with 16 participants who pressed a button upon detection of an approaching vehicle equipped with a FMVSS-141 compliant additive sound. Results are presented at probability of detection versus distance. This provides a unique view of the effectiveness of these sounds and additional aspects of this problem such as additive sound contribution to noise pollution and listener false alarm rates. Lastly, the talk will cover difficulties with this type of outdoor testing and recommendations for conducting them in a virtual environment.

10:40

2aEA2. Cryogenic testing of microphones and preamplifiers for MEMS quench detection in superconducting magnets. Reid Baris (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, reid.baris@tufts.edu), Mischael Anilus, Zijia Zhao (Mech. Eng., Tufts Univ., Medford, MA), Steve Chau (Tanner Res., Monrovia, CA), Wayne Liu (Tanner Res., Duarte, CA), Amish Desai, Michael Emerling (Tanner Res., Monrovia, CA), Robert Littrell, Craig Core, Stacey Tang (Vesper Technologies, Inc, Boston, MA), Luisa Chiesa (Mech. Eng., Tufts Univ., Medford, MA), Makoto Takayasu (Plasma Fusion and Sci. Ctr., MIT, Cambridge, MA), and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA)

High Temperature Superconducting (HTS) tapes are needed for high field magnets in high-energy physics and nuclear fusion. However, quench events, in which the superconductor loses its superconductivity, can cause significant and costly damage to the device if allowed to propagate. Consequently, rapid quench detection methods are crucial. Existing methods such as voltage taps can be less effective in HTS applications, as the normal zone propagation velocities of HTS devices are slow, drastically increasing quench detection time (Iwasa, Cryogenics, 2003). We propose an alternative method that utilizes a linear array of MEMS microphones embedded in the cryogenic coolant channel (Takayasu, IEEE Transactions on Applied Superconductivity, 2019). The system requires amplifiers and acoustic sensors capable of operation at cryogenic temperatures as low as 4.2 K, although operation at 20 K may be sufficient. We report on cryogenic testing of the Vesper microphone preamplifier ASIC, as well as opamp based preamplifiers using the OPA838, LT1464, AD8591, and AD8057 ICs. The amplifiers are being used with the Vesper VM4 MEMS microphone to achieve acoustic measurements in gaseous helium at temperatures as low as 8 K. A charge amp scheme using the AD8591 achieved the best results to date. [Support from DOE STTR DE-SC0019905.]

10:55

2aEA3. Lumped parameter modeling of two dimensional mechanical and acoustical components. Carter J. Childs (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., The Penn State Univ., University Park, PA 16802, cjc357@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Multidomain lumped parameter models used to predict transducer performance are an efficient way of modeling full system responses. To increase the accuracy of such models, transmission lines can be used. Discretizing the line into small spatial segments allows for lumped parameters to be used for modeling partial differential equations. Segmentation in one dimension has long been used in circuit simulators and multidomain analysis codes. This work presents a solution in two dimensions for both mechanical and acoustical physical domains. For example, the analysis of the mechanical motion of membranes and the acoustical behavior of thin cavities can be analyzed. This enables increased accuracy in modeling a transducer as the shape function of the membrane is accounted for. Further, the mechanical and acoustical physics can be coupled for an appropriate Multiphysics interaction. This is a simple demonstration of the possibilities that could be achieved using a spatially discretized multidomain models for more complex transducers.

11:10

2aEA4. Pinhole measurements of hydroacoustic pressure fluctuations beneath wall-bounded turbulent flows. Jane H. Kim (Naval Architecture and Marine Eng., Univ. of Michigan, 1231 Beal Ave., W.E. Lay Automotive Lab., Rm. 2010, Ann Arbor, MI 48105, janehjki@umich.edu)

The common flush-mounted-transducer measurement method for underwater surface pressure fluctuations results in transducer-size-limited spatial resolution, which may be inadequate for the fluctuations produced beneath a turbulent boundary layer (TBL). For water moving over a smooth surface at speeds of a few meters per second, the relevant length scales may be less than 100 mm with fluctuation magnitudes far less than 1% of the free-stream dynamic pressure. Unfortunately, such size and sensitivity requirements cannot simultaneously be met with common instrumentation. However, mounting microphones behind pinholes is a possible solution for wall-bounded airflows. This presentation covers the effect of pinhole mounting pressure transducers for measuring surface pressure fluctuations in wall-bounded turbulent flows in water. The experimental work is conducted in a turbulent channel flow at varying flow speeds (0.5 to 7 m/s) using pressure transducers with a 5-mm-diameter sensitive area that are flush mounted and

mounted behind variable size pinhole openings. The measured pressure-fluctuation power spectra are compared to each other and to results from past direct numerical simulations, in the 10 Hz to 10 kHz frequency range, to determine the utility of pinhole mounting for underwater TBL pressure-fluctuation measurements. [Sponsored by ONR.]

11:25

2aEA5. Towards aeroacoustic measurements under hyperbaric condition. Constantinos S. Kandias (Aerosp. Eng., Penn State Univ., 229 Hammond Building, University Park, PA 16802, csk5332@psu.edu), Nick Morgan, Andrew Shields, Eric Greenwood, and Mark Miller (Aerosp. Eng., Penn State Univ., State College, PA)

Full scale experimental aeroacoustic measurements are often impractical due to the prohibitively high costs associated with full scale models and large facilities capable of far-field acoustic measurements. Although small scale models can provide useful acoustic data, this scaling is limited by the inability to achieve full scale equivalent aerodynamic conditions at smaller scale. However, aerodynamic similarity can theoretically be achieved using much smaller models under hyperbaric conditions. While promising, microphones and anechoic materials used in acoustic experimentation are not explicitly designed for use in such conditions, and so their behavior must be verified. To characterize the response of condenser microphones at elevated pressures, a pistonphone style calibrator was developed to operate over a range of pressures. Likewise, for the anechoic materials, an impedance tube was developed to evaluate their performance at elevated pressures. The performance of these systems under atmospheric pressure and hyperbaric conditions is discussed, along with details of the experimental method, and observed microphone response, and acoustic insulation performance at high pressure.

11:40

2aEA6. Aluminum nitride piezoelectric micromachined ultrasound transducers with applications in sonic anemometry. Freidlay Steve (Mech. Eng., Tufts Univ., Medford, MA), Robert Littrell, Craig Core (Vesper Technologies, Inc., Boston, MA), Don Banfield (Astrophysics and Planetary Sci., Cornell Univ., Ithaca, NY), and Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu)

Recent results demonstrate the promise of sonic anemometry for accurate, three dimensional, high update relative wind measurements in high altitude (stratospheric) balloons and for planetary science missions on Mars, Venus, and other environments (Banfield, JASA 2016, White, ASA 2020, White, AIAA 2020). For these applications, low weight and small size is desirable. We propose an aluminum nitride piezoelectric micromachined ultrasound transducer (PMUT) as a candidate for a small ($\sim$1 cm^3) low weight ($\sim$10 g) sonic anemometry system. Two PMUT designs were fabricated using Vesper Technologies MEMS microphone process by Vanguard Semiconductor International on 1.4×1.4 mm^2 die. The first device has 270 μm diameter diaphragms and operates at 610 kHz, the second 190 μm diameter diaphragms and operates at 890 kHz. By operating at these high frequencies, the acoustic path length can notionally be reduced to 1 cm, while still retaining sufficient phase resolution to measure wind with 1 to 10 cm/s accuracy. The substantial challenge is producing sufficient signal to overcome acoustic absorption at these elevated frequencies. This design is aggressively exploratory in nature. The PMUT devices have been fabricated and operate, but the sonic anemometry system has not yet been demonstrated. [Work supported by NASA PICASSO NNX16AJ24G, NSF 1934553 and NSF 2110083.]

Session 2aMUa

Musical Acoustics and Physical Acoustics: Nonlinearities in Musical Acoustics

Nicholas Giordano, Cochair
Physics, College of Sciences and Mathematics, Auburn University, Auburn, AL 36849

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—9:00

Invited Papers

9:05

2aMUa1. Abstract synthesis representation of nonlinear behaviour in coupled resonators. Tamara Smyth (Music, UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0021, trsmyth@ucsd.edu)

In this work, abstract synthesis techniques known as frequency/amplitude/phase modulation (FM, PM, and AM, respectively) are explored in musical acoustic systems involving coupled oscillators, nonlinear mechanical resonators and feedback. Though these systems are frequently modeled having acoustic/physical parameters (and units), there is a strong musical benefit of an abstract representation in which input parameters control more perceptual aspects of the produced sound. Previously coined "loopback FM is an extension of the well known FM synthesis technique in which a carrier oscillator "loops back to modulate its own frequency, resulting in behaviour that is characteristic of nonlinearities. With reference to loopback FM, along with its strongly related feedback PM/AM, nonlinearities that influence sounding frequency, spectral bandwidth and amplitude envelope are explored with musical instrument examples.

9:25

2aMUa2. Reproducible excitation of string instruments using a robotic arm. Montserrat Pàmies-Vilà (Dept. of Music Acoust.—Wiener Klangstil (IWK), Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, mdw - Inst. 22 IWK, Vienna 1030, Austria, pamies-vila@mdw.ac.at), Alexander Mayer, and Vasileios Chatziioannou (Dept. of Music Acoust. - Wiener Klangstil (IWK), Univ. of Music and Performing Arts Vienna, Vienna, Austria)

In string instruments, while the vibrations of the wooden bodies can be mostly described as linear, the excitation mechanism is usually nonlinear. For instance, the frictional forces that excite bowed-string instruments have undergone both theoretical and experimental studies due to the underlying complexity. In order to analyze the vibratory behavior and the radiation pattern of musical instruments, it is often necessary to excite the instruments repetitively under controlled conditions, for which artificial means are customarily used. In this work, the sound production in string instruments is studied with the usage of a computer-controlled robotic arm, which provides high precision in positioning as well as controlled motion with 6 degrees of freedom. Together with a customized 3D-printed clamp-system, it is possible to hold and move objects in order to mimic the gestures of instrumentalists, equipping the arm, for example, with a bow or a plectrum. This systematic experimental approach allows to repeatedly reproduce the player-instrument interaction, either by accurately recreating the player actions or by precisely adjusting one or several control parameters. In this study, the possibility to introduce such a robotic arm for the investigation of string instruments is presented and exemplified using different case studies.

Contributed Papers

9:45

2aMUa3. Nonlinearities in the singing voice. Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

The vocal ligament and the thyroarytenoid muscle are the vibrating elements that control fundamental frequency in singing. Their stress-strain relations are highly nonlinear. For a 3 octave f_o range, it requires an exponential rise in stress of the form $e^B\varepsilon$, where ε is the strain. The exponent B must be greater than 8.0 if there is a 2:1 vocal fold length change. There are also strong interaction nonlinearities. Maximum acoustic and aerodynamic power is transferred from the source to the airway if the glottal impedance approaches the airway impedance. The airway impedance can be controlled with the epilaryngeal airway, which acts like the mouthpiece of a trumpet.

10:00

2aMUa4. Spectral content of the attack portion of the tone produced by a soprano recorder. Nicholas Giordano (Phys., Auburn Univ., College of Sci. and Mathematics, Auburn, AL 36849, njg0003@auburn.edu) and Jared W. Thacker (Phys., Auburn Univ., Auburn, AL)

We have used Navier-Stokes-based modeling to calculate tones produced by a full-size soprano recorder. Of all the wind instruments, the player of a recorder has relatively little impact on the tone that is produced. However, one thing that the player does control is the way that the blowing pressure is increased from zero at the start of a tone to a final value that is maintained until the end of the tone. We show how the ramp-up time for the blowing pressure has a profound effect on the spectral content of the attack portion of the tone. Using a correlation analysis, we explore this spectral content and consider how it can, in certain blowing regimes, be an important and audible contributor to the tone color produced by a recorder. [Work supported by NSF grant PHY1513273.]

10:15

2aMUa5. Preliminary results from a time domain model of a clarinet-like system. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, sct12@psu.edu)

A time domain model of a clarinet-like system has been built using the Simscape multidomain modeling system. This model largely replicates physical features that have been included in previous models by others. The air column has constant inside diameter. The tube has thirteen tone holes. The first three holes are spaced to give approximate chromatic tone spacing. The remaining holes have the same spacing as the last chromatic pair, and all the holes are sized to give a constant tone hole lattice cuttoff frequency of 1500 Hz. The reed is modeled as a single degree of freedom oscillator that terminates the "closed" end of the tube. Blowing pressure is provided by a mouth cavity of fixed volume. This model allows calculation transient behavior that becomes steady state if the input pressure remains constant. We present results of startup transients, slow crescendo and decrescendo playing, and oscillation behavior in response to opening and closing the tone holes. The current model does not accurately match the design of any particular instrument, but it does demonstrate an ability to calculate the behavior of some interesting playing behaviors of an instrument.

TUESDAY MORNING, 30 NOVEMBER 2021 — QUINAULT, 10:45 A.M. TO 11:40 A.M.

Session 2aMUb

Musical Acoustics: Acoustics of Brass Instruments

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

Thomas R. Moore, Cochair
Department of Physics, Rollins College, Department of Physics, Box 2743, Rollins College, Winter Park, FL 32789

Invited Papers

10:45

2aMUb1. Measurements and simulations of sung multiphonics on a trombone played by an artificial mouth. Christopher Page (School of Phys. and Astronomy, Univ. of Edinburgh, Edinburgh, United Kingdom), Joël Gilbert (Laboratoire d'Acoustique UMR CNRS 6613, Université du Mans, Le Mans, France), Alan Woolley (School of Phys. and Astronomy, Univ. of Edinburgh, Edinburgh, United Kingdom), and Donald M. 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)

Playing a sung multiphonic on a brass instrument is a specialised technique in which the performer sounds one pitch through lip vibration while simultaneously singing a different pitch. The result is typically a complex sound in which several different pitches are audible. A full acoustical description of the sound generation process would include the mutual interactions of the two nonlinear flow valves (vocal folds and lips), the player's windway resonances, and the resonances of the instrument. Velut *et al.* (JASA 2016) reported measurements of mouth pressure, pressure inside the mouthpiece, and radiated sound while a trombone player performed various sung multiphonics. A simplified time domain simulation model replacing vocal folds and windway by a sinusoidal source was partially validated. In the present work the human player has been replaced by an artificial mouth, allowing the "sung" mouth pressure to be provided either by a sinusoidally excited loudspeaker or a human singer. Results are broadly in agreement with those of Velut *et al.*, and with time domain simulations using sinusoidal forcing in the mouth cavity.

11:05

2aMUb2. Constrained continuation of a physical model of brass instrument. Vincent Fréour (YAMAHA Corp., R&D Div., C.N.R.S. Laboratoire de Mécanique et d'Acoustique, 4, Impasse Nikola Tesla, Marseille 13013, France, vincent.freour@music.yamaha.com), Louis Guillot (Aix Marseille Univ., CNRS, Centrale Marseille, LMA UMR7031, Marseille, France), Hideyuki Masuda (YAMAHA Corp., R&D Div., Hamamatsu, Japan), Christophe Vergez, and Bruno Cochelin (Aix Marseille Univ., CNRS, Centrale Marseille, LMA UMR7031, Marseille, France)

Constrained continuation is a method allowing some constraints to be applied to bifurcation diagrams, while relaxing some parameters of the model, which then become new unknowns of the system. Applied to a physical model of wind instruments, this method allows some constraints to be applied to a solution branch, and the evolutions of the relaxed parameters to be computed along this solution branch. We apply this approach to a physical model of trumpet, where constraints on the playing frequency and on the amplitude of the internal pressure with respect to the mouth pressure are established from experimental measurements on a trumpet player. By relaxing different parameters of the model, it is possible to approximate the behavior of the trumpet player using constrained continuation, and to quantify the variations of the model parameters that are necessary to fulfill the constraints. Application to different trumpet models reveal differences between instruments, that are confronted to some preliminary perceptual evaluations.

Contributed Paper

11:25

2aMUb3. Calculation of the trumpet impedance with various mouthpieces. Sarah Babione (Phys., Berry College, 500 E Spring St., Oxford, OH 45056, sbabione2@gmail.com) and Herbert Jaeger (Phys., Miami Univ., Oxford, OH)

This project explored computationally how mouthpieces of various dimensions affect the acoustic input impedance for the entire trumpet. The impedance was calculated using a cone approximation. The trumpet was divided into cones shorter than the wavelengths considered. The measurements for the trumpet were taken from the literature. Each cone was further divided into 1 mm-long slices in order to get an accurate value for the thermal and viscous wall losses. The load impedance at the open bell was taken as zero. The input impedance of the trumpet with mouthpiece was calculated iteratively such that the input impedance of one cone was used as the load impedance for the next cone. Calculations were done for frequencies up to 2000 Hz and for a variety of mouthpiece geometries but with the same trumpet sections. The effect of the mouthpiece geometry on the instrument's impedance spectrum will be discussed.

TUESDAY MORNING, 30 NOVEMBER 2021 301 (L)/304 (O), 8:00 A.M. TO 9:15 A.M.

Session 2aNSa

Noise and ASA Committee on Standards: General Topics on Noise and Standards

James E. Phillips, Chair
Intertek, 4703 Tidewater Ave., Suite E, Oakland, CA 94601

Contributed Papers

8:00

2aNSa1. Relationship between vessel speed and sounds levels in the Bering Strait. Erica D. Escajeda (School of Aquatic and Fishery Sci., Univ. of Washington, 1122 NE Boat St., Seattle, WA 98105, escajeda@uw.edu), Kathleen Stafford, Rebecca Woodgate, and Kristin L. Laidre (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Vessel speed limits have been proposed as a means of reducing underwater ship noise, however it is unclear how effective such a measure would be in the Bering Strait, a natural bottleneck for ships transiting into the Arctic from the Pacific Ocean. In this study, we examine how ship noise varies with vessel type and speed using Automatic Identification System (AIS) data collected from vessels traveling through the strait along with acoustic recordings from three moored hydrophones. We matched recordings with ship noise to individual vessels that passed within 100 km of each hydrophone in June through November 2013–2015. A total of 69 sound files were analyzed and matched with individual vessels, with tug ($n = 21$) and cargo ships ($n = 18$) as the most common vessel type observed in our dataset. Sound levels for each vessel were calculated and compared as a function of vessel type and speed. The results of our study could inform policymakers and managers on the effectiveness of vessel speed limits on reducing ship noise in a sensitive Arctic habitat.

8:15

2aNSa2. Measurements of ship noise using an acoustic camera: A first survey. Johan A. Bocanegra, Davide Borelli (Dept. of Mech. Eng., Univ. of Genoa, Genova, Italy), and Corrado Schenone (Dept. of Mech. Eng., Univ. of Genoa, 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.

8:30

2aNSa3. An analysis of angular directivity for small caliber firearm noise. Steven C. Campbell (Ball Aerosp., 2875 Presidential Dr., Ste 180, Beavercreek, OH 45324, sccampbe@ball.com), Alan T. Wall (Sensory Systems Branch, Air Force Res. Lab., Wright Patterson Airforce Base, OH), William J. Murphy (Div. of Field Studies and Eng., Noise and Bioacoustics Team, National Inst. for Occupational Safety and Health, Cincinnati, OH), Reese D. Rasband (Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Accurately describing the sound field generated by an impulsive source is non-trivial. Conventional methods for analyzing impulsive noise of firearms consists of deploying one or more circular microphone arrays centered on the muzzle of the source. Currently, much debate exists around the design of these microphone arrays (radius, quantity of sensors, angular spacing, etc.) and the resulting effect on capturing source directivity of the impulse event. An in-depth investigation of data collected on small caliber firearms operating in outdoor environments is performed with an emphasis on describing source directivity as a function of (1) array radius, (2) quantity of microphones, and (3) angular spacing. Ultimately, given a sufficient circular microphone array, the angular directivity of an impulsive source can be generalized using analytical curve fitting functions.

8:45

2aNSa4. Plumb-full of hush to the Brim: Park sound levels per American National Standards Institute S12.100-2014. Davyd H. Betchkal (Natural Sounds and Night Skies Div., National Park Service, PO Box 9, Denali Park, AK 99755, Davyd_Betchkal@nps.gov)

In this talk we explore the use of ANSI/ASA S12.100-2014, the contemporary standard to *'define and measure the residual sound in protected natural and quiet residential areas'*. Key provisions are digested using examples from measurement sites within The National Park System. Special attention will be paid to the acoustic form of natural areas in contrast to urban areas (Secs. 3.12 and 3.13), avoiding bias from transients (Secs. 5.2 and 5.3), and acknowledging measurements censored by instrument background noise (Secs. 5.9 and 6.4). Adoption of standard practice in these few areas could vastly improve the accuracy of impact assessments for public lands nationwide.

9:00

2aNSa5. Transmission loss calculation of a pipe with dissipative silencer containing porous materials. Haesang Yang (Dept. of Naval Architecture and Ocean Eng., Seoul National Univ., 1 Gwanak-ro, Gwanak-gu, Seoul 08826, Republic of Korea, coupon3@snu.ac.kr), Jongmoo Lee, Chul-Won Lee, and Woojae Seong (Dept. of Naval Architecture and Ocean Eng., Seoul National Univ., Seoul, Korea (the Republic of))

Various types of silencers have been used to reduce noise transmitted through pipe system. In particular, a dissipative silencer containing absorbing materials inside has an ability of noise control by transferring acoustic/vibrational energy to thermal energy. In this study, the acoustic performance of a dissipative silencer lined with two-layer absorbing materials is investigated. Herein, the absorbing materials in the form of a polymer foam are considered as a porous medium composed of a solid polymer skeleton and internal pores. From the boundary conditions at the rigid wall and the interfaces between the porous materials, eigenvalues and eigenfunctions in the dissipative silencer are determined. Finally the transmission loss is obtained through the mode matching method. The effects of geometry and acoustic properties of the porous medium are also discussed.

Session 2aNSb

Noise, Architectural Acoustics, Engineering Acoustics, and Structural Acoustics and Vibration: Building Systems Noise and Vibration Control: Beyond ASHRAE Chapter 49 I

Brandon Cudequest, Cochair
Threshold Acoustics, 141 W Jackson Blvd. Suite 2080, Chicago, IL 60604

Joseph Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, Ste. 209, Woodbridge, NJ 07095

Chair's Introduction—9:45

Invited Papers

9:50

2aNSb1. Duct, duct, goose chase? Revisiting algorithms for duct flow noise. Brandon Cudequest (Threshold Acoust., 141 W Jackson Blvd. Ste. 2080, Chicago, IL 60604, bcudequest@thresholdacoustics.com)

The American Society for Heating, Refrigeration, and Air Conditioning Engineers (ASHRAE) has several practical resources for flow noise prediction. There are reference tables for maximum air velocities for a given background noise rating as well as tables for maximum "free area" speeds for supply diffusers or return registers. For other duct fittings, engineers may use a variety of algorithms to predict octave band sound power levels. Experiments have demonstrated a direct relationship between flow noise and duct size, aspect ratio, and pressure drop. Currently, few prediction methods accurately take these factors into account because the underpinning algorithms are based on a limited set of ductwork sizes. To better understand how these factors impact airflow generated noise, ASHRAE has conditionally approved Work Statement 1919: The effects of duct size and aspect ratio on flow noise in elbows. This paper will review past literature, highlight shortcomings of current prediction methods, and discuss the proposed methodology for upcoming research.

10:10

2aNSb2. Evaluation of Ashrae background noise criteria through *in situ* measurement. Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), John Lo Verde, and Wayland Dong (Veneklasen Assoc., Santa Monica, California, CA)

Chapter 49 of the ASHRAE Applications Handbook establishes background noise design criteria for typical interior occupancies. These criteria are referenced and commonly used as an acoustical basis of design for codes, such as the International Green Construction Code, and for rating systems, such as LEED, as well as other project documents and design standards. They are set forth to determine the level of occupant comfort from mechanical systems. The authors have observed that compliance with these standards is not always necessary to avoid complaints. In fact, the benefit of compliance with the standards is limited to the protection of the acoustician, rather than meeting needs for the users of the space. This observation has motivated the authors to assess the established ASHRAE criteria as compared to measured interior sound levels with HVAC systems in operation. The authors will present objective data associated with noise measurements in specific HVAC applications. Using data from subjective determinations, the authors will propose a system for determination of occupant reaction that can be considered.

10:30

2aNSb3. Applications of building information modeling software to HVAC noise prediction. Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182, samuelunderwood@huskers.unl.edu) and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

This paper presents a methodology and workflow for utilizing Building Information Modeling (BIM) software to expedite the calculation of duct borne noise according to Chapter 49 of the ASHRAE Handbook – HVAC Applications. Using Dynamo script for Autodesk Revit, relevant building mechanical system parameters are extracted to perform noise calculations with reduced manual input. Resulting predictions for receiver sound pressure levels are compared to other software tools available in the industry. Advantages and limitations of this approach are weighed to assess the utility of Building Information Models in the context of the noise prediction workflow. The results of this effort support the development of more robust duct noise prediction tools that are not only more accurate but also more user-friendly.

10:50

2aNSb4. Improving fan inlet conditions to reduce noise. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Zachery O. L'Italien (McKay Conant Hoover, Westlake Village, CA)

This presentation will review various examples of fan inlet conditions that caused excessive noise in different frequency ranges, and system modifications for noise reduction that involved a few intermediate steps to maximize effectiveness and sometimes to reduce initial client resistance. Examples will include several large air handlers, a healthcare facility exhaust fan, and some informal experiments.

11:10

2aNSb5. Integrated heating, ventilation, and air conditioning noise control solutions. Chris Dziedzic (Noise Control, Price Industries, 638 Raleigh St., Winnipeg, MB R3X 0L7, Canada, chrisd@priceindustries.com)

There are a wide variety of acoustical considerations with respect to commercial building heating, ventilation and air conditioning (HVAC) systems. This presentation will review and discuss integrated solutions and noise control products that can be utilized to solve noise problems during the design process. Air Moving and Air Distribution devices can affect the overall acoustics of a space and when selected properly, help to reduce the background sound with the space. There are also a wide variety of noise control products available to help mitigate and attenuate sound transmission and transfer from the noise source to the receiver within commercial buildings. This presentation will review some of the most commonly used noise control products and their overall design characteristics to provide understanding on how they operate.

11:30

2aNSb6. Paint a clearer picture with *in situ* auralizations. Shane J. Kanter (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, skanter@thresholdacoustics.com), Chris Springthorpe (Threshold Acoust., Chicago, IL), Nicolaus T. Dulworth, Robin Glosemeyer Petrone, and Scott Pfeiffer (Threshold Acoust., Chicago, IL)

Communication is at the heart of everything we do as consultants. We are constantly striving for new and better ways to guide our clients to make informed decisions. *In situ* auralizations of major pieces of future mechanical / electrical equipment allows major stakeholders and decision makers to get an aural sense for various equipment configurations, enclosure types, and barrier wall buildups prior to installation. After the auralization, board members and homeowners from past projects had a richer understanding of the anticipated experience at the property line and nearby classroom spaces when compared to reviewing graphs and charts alone. We will explore the value of, and challenges with, this *in situ* auralization method, as well as some future improvements to further validate the process through the lens of two past projects—New Trier High School and Lake Bluff Middle School in Illinois.

Session 2aPA

Physical Acoustics, Engineering Acoustics, ASA Committee on Standards, Structural Acoustics and Vibration, and Noise: Wind Noise

Gregory W. Lyons, Cochair
Information Technology Laboratory, U.S. Army Engineer Research and Development Center, 3909 Halls Ferry Road, Vicksburg, MS 39180

W. C. K. Alberts, Cochair
CCDC-Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783

Richard Raspet, Cochair
NCPA, University of Mississippi, 145, Hill Drive, University, MS 38677

Chair's Introduction—8:00

Invited Papers

8:05

2aPA1. Correlation properties of wind noise pressure on the ground surface. 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)

Recent studies of the seismic displacement of the ground use the theory developed by Jiao Yu to predict the pressure spectrum of the wind noise on the ground. The prediction of the displacement due to this pressure employs the ground properties and the spatial correlation properties of the pressure on the ground. The theory is successful at predicting the amplitude of the seismic displacement at the surface but does not predict the correct behavior with depth. These calculations used the low frequency correlation functions measured by Shields [JASA **1**(17), 3489–3496 (2005)]. The sensors in Shields' study were on the ground surface and may have been subject to stagnation pressure contributions in addition to the surface pressure fluctuations. A new measurement of the surface pressure correlation using 12 microphones mounted in two perpendicular arms, flush to the ground surface, has been performed. Since the arms cannot be moved to align with the wind, new analysis methods have been employed to extract the correlation coefficients along and perpendicular to the wind direction. Initial analysis indicates that the correlation coefficients are significantly smaller than those of Shields and the convection velocity is slightly larger than the .7 *U(2.0 m) used by Yu.

8:25

2aPA2. Wind noise performance analysis of a prototype compact infrasound array. Garth Frazier (NCPA, Univ. of Mississippi, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu) and David Harris (Hyperion Technol. Group, Inc., Tupelo, MS)

Previously, presentations have been given at ASA meetings that demonstrated the potential for compact arrays of infrasound sensors achieving high gain against wind noise under appropriate environmental conditions without the use of large windscreens. These presentations focused on using specific models for the spatiotemporal correlation function of the wind noise as part of the processing methodology, and they did not directly address the features of the compact array design that influence performance. This presentation will study the effects of several compact array features on performance using a recently published cross-spectral density model for wall-bounded turbulent pressure fluctuations. Additionally, data from several experiments will be presented including data from a prototype compact infrasound array.

8:45

2aPA3. An experimentalist's overview of wind reduction systems in infrasound measurements. Jeremy Webster (Los Alamos National Labs, LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544, jwebster@lanl.gov)

The largest limiting factor in infrasound measurements is wind noise. This noise is caused by pressure fluctuations generated by turbulent flow passing over an acoustic sensor. The size scales of these turbulent fluctuations vary from centimeters to tens of meters and larger. This, coupled with the wide variety of possible measurements, means that there is no single best solution to this problem. Instead, wind reduction systems should be tailored to best fit the intended measurement. In this talk, I will give a brief review of wind noise from an experimentalist's point of view, and will discuss some of the physical principles to consider when choosing a wind noise reduction system. This will include spherical and domed wind screens, porous hoses and pipe arrays, probe microphones, and terrain and foliage considerations.

9:05

2aPA4. Update on using metal domes for wind noise reduction at infrasonic frequencies. Carrick Talmadge (NCPA, Univ. of Mississippi, Oxford, MS, clt@olemiss.edu)

The utility of infrasound for long range monitoring applications such as remote chemical explosions, or natural sources of infrasound such as tornadoes, volcanoes or hurricanes, continues to be hampered by the high noise floors experienced in the presence of high winds. Wind noise is typically filtered using pipe arrays. But these generate significant distortion, making them non-ideal for signals with significant frequency content above a few Hz, and do not produce sufficient noise reduction during windy measurements even at their optimal frequency band (0.5–1 Hz), making their utility generally limited to low-wind-noise periods. We present here an update on resesarch performed by the National Center for Physical Acoustics (NCPA) using field sites near Oxford (University of Mississippi Field Staiton) and at the Sandia National Laboratory Facility for Acceptance, Calibration, and Testing (FACT) site. The 20-foot dome at the FACT site is collocated with an IMS 18-m rosette filter and a wind noise filter developed by the University of Alaska Fairbanks. We compare in this presentation the relative performance of traditional pipe arrays to the 20-ft dome. The performance of smaller wind domes, including 42-in.^{2} perferorated metal domes developed at the NCPA, are also discussed.

9:25

2aPA5. Large porous wind barriers as a mitigation technique for turbulence induced infrasonic noise. JohnPaul R. Abbott (None, 501 Prospect Blvd., Apt. 25D, Frederick, MD 21701, johnpaul.abbott@gmail.com)

An infrasonic wind barrier is a large porous device that effectively reduces wind noise by disrupting and minimizing the flow near the pressure sensor, and transferring the local and direct turbulence induced noise interactions from the pressure sensor to the surface of the barrier where the sum total of the poorly correlated noise induced turbulence interactions are reduced by area averaging. In addition to effective wind noise reduction these devices maintain acoustic waveform fidelity, work in adverse conditions, and can reduce noise frequencies with associated turbulence scales sizes that are several times the size of the barrier itself. This paper will review wind barrier research conducted at the National Center for Physical. This will include a discussion on the mechanical properties of the barrier that lead to effective noise reduction including its porosity, size, shape, and number of filtering layers. Barriers designed and studied at the National Center for Physical Acoustics and by others are compared and discussed. A simple empirical model predicting wind noise reduction based on the mechanical properties of the barrier and the turbulence flow interactions located inside, on the surface, and outside of the barrier is given

9:45–10:00 Break

10:00

2aPA6. Comparison of wind filter designs for infrasound sensors. Sarah McComas (U.S. Army Res. and Development Ctr., 3909 Halls Ferry Rd. ATTN: CEERD-GS-S, Vicksburg, MS 39180, sarah.mccomas@usace.army.mil), Chris Hayward, Stephen Arrowsmith, Brian Stump (Southern Methodist Univ., Dallas, TX), and Mihan McKenna (U.S. Army Res. and Development Ctr., Vicksburg, MS)

Sensors in infrasound arrays measure the total acoustic field that contains signals of interest as well as coherent and incoherent noise. Modern array designs utilize sensors that have a quantified response that extends across the acoustic spectrum from the infrasound passband (<20 Hz) into the low-end audible passband (<1000 Hz). Mechanical wind filters applied to individual sensors are an integral part of array designs that reduce incoherent noise and improve the signal-to-noise ratio of coherent signals. Common forms of these filters are porous hose rosettes and domes, which provide spatial averaging of coherent energy over the dimension of the mechanical filter. These filters also provide additional noise reduction by limiting the sensor's sensitivity to the effects of the turbulent boundary layer. This presentation will focus on direct comparison of four wind filter designs: 4- and 10-hose porous hose rosettes, a 1-m fabric dome, and a high-frequency shroud. The analysis presented will quantify similarities and differences in the wind filter performance that led to recommendations for specific use cases dependent on expected environmental conditions. [Permission to publish was granted by the Director, Geotechnical and Structures Laboratory.]

Contributed Papers

10:20

2aPA7. Keep it simple: A right conical infrasound windscreen from off-the-shelf components. W. C. K. Alberts (CCDC-Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, william.c.alberts4.civ@mail.mil)

Windscreens for higher frequency infrasound are a continued topic of discussion because the ideal has yet to be found, i.e., an inexpensive, easily shipped, compact windscreen with a flat 3 Hz to >20 Hz passband and nearly zero transmission loss. Recent research has shown that the robust porous hose, porous fabrics, and perforated metal domes are capable of filling many of those needs. However, each technology has drawbacks that necessitate further research to arrive at a greater understanding of the windscreen and to approach the ideal. Guided by this previous infrasound windscreen research, a right conical windscreen was constructed from easily obtained commercial products and subsequently tested beside previously characterized windscreens. The presentation will discuss results of the experimental program to include comparisons of power spectral densities and transfer functions.

10:35

2aPA8. Application of spectral-based wind noise reduction method to acoustical measurements. 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), Shane V. Lympany, and Matt Calton (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Wind-induced microphone self-noise is a non-acoustic signal that may contaminate outdoor acoustical measurements, particularly at low frequencies, even when using a windscreen. A recently developed method [Cook *et al.*, JASA Express Lett. **1**, 063602 (2021)] uses the characteristic spectral slope of wind noise in the inertial subrange for screened microphones to

automatically classify and reduce wind noise in acoustical measurements in the lower to middling frequency range of human hearing. To explore its uses and limitations, this method is applied to multiple acoustical measurements collected in both urban and rural environments. The method can be applied to one-third octave band spectral data with different frequency ranges and sampling intervals. By removing the shorter timescale data at frequencies where wind noise dominates the signal, the longer timescale acoustical environment can be more accurately represented. While considerations should be made about the specific applicability of the method to particular datasets, the wind reduction method allows for simple classification and reduction of wind-noise-contaminated data in large, diverse datasets. [Work supported by U.S. Army SBIR.]

TUESDAY MORNING, 30 NOVEMBER 2021 COLUMBIA C, 8:00 A.M. TO 11:20 A.M.

Session 2aPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Auditory Perception of Stationary and Moving Sounds I

Cynthia F. Moss, Cochair
Psychological and Brain Sciences, Johns Hopkins University, 3400 N. Charles Street, Ames 200B, Baltimore, MD 21218

Michaela Warnecke, Cochair
Audio Team, Facebook Reality, 9845 Willows Road NE, Redmond, WA 98052

Chair's Introduction—8:00

Invited Papers

8:05

2aPP1. Sound-source localization when listeners and/or sound sources move. William A. Yost (College of Health Solutions, ASU, PO Box 870102, ASU, Tempe, AZ 852870102, william.yost@asu.edu) and M. Torben Pastore (College of Health Solutions, Arizona State Univ., Tempe, AZ)

In the dynamic everyday world, sound sources and/or listeners move when animals attempt to localize sound sources. Determining sound-source locations in these dynamic scenarios requires several computations by the brain. The auditory-spatial cues (Interaural Time Differences, ITDs; Interaural Level Differences, ITDS; and spectral cues associated with the head-related transfer function, HRTF) indicate the location of sound sources relative to the head. When the source and/or listener move, these cues change and by knowing the position of the head, the brain could determine which moved and where it moved based on the auditory-spatial cue changes. If interaural cues (ITDs or ILDs) are used for sound-source localization, cone-of-confusion errors occur, and head motion can resolve front-back, cone of-confusion errors. The elevation of a sound source, not effected by cone-of-confusion errors, might be able to be determined when both the source and listener move. This presentation will describe work in the Spatial Hearing Lab at ASU related to how sound source and listener motion affect sound source-location judgements in these scenarios. [Research supported by grants from NIDCD, WAY: R01-DC015214 and MTP: F32-DC016808.]

8:35

2aPP2. On the role of head rotation in solving the HRTF inverse problem. W. Owen Brimijoin (Audio Team, Facebook Reality Labs, 9845 Willows Rd. NE, Redmond, WA 98052, owen.brimijoin@fb.com) and Michaela Warnecke (Audio Team, Facebook Reality Labs, Redmond, WA)

The sound arriving at the eardrum is a complex composition of the sound source spectrum, the acoustic properties of the listener's environment, and the directionally dependent filtering of the listener's head-related transfer function (HRTF). Listeners only ever have access to this mixture, and it remains unknown how they are nonetheless capable of disentangling features intrinsic to the sound source from those created by the source's position in the environment. We hypothesized that head movements, by changing only the HRTF-related components of an arriving sound, inform a listener's judgement of source spectrum. We tested this with virtual acoustics, presenting listeners with a speech sound rendered at an elevated frontal location and asked them to judge which of two subsequent copies of the same sound rendered at the horizon had spectral noise added to them. The test sounds were either head-locked, world-locked, or moved inverted to the listener's movements, and were rendered with either individual or non-individual HRTFs. Error patterns across HRTF and motion conditions appear to confirm the hypothesis that head rotation plays a role in the estimation of sound source spectrum, and further indicate that listeners have expectations about how the signal at the ear should change as the head moves.

9:05

2aPP3. How human listeners make sense of crowded acoustic scenes. Maria Chait (Ear Inst., UCL, UCL Ear Inst., 332 Gray's Inn Rd., London WC1X 8EE, United Kingdom, m.chait@ucl.ac.uk)

The notion that sensitivity to the statistical structure of the environment is pivotal to perception has recently garnered considerable attention. I will discuss a series of recent experiments in which we investigated this sensitivity in the context of crowded acoustic scenes. Stimuli are artificial 'soundscapes' populated by multiple (up to 14) simultaneous streams ("auditory objects") comprised of tone-pip sequences, each with a distinct frequency and pattern of amplitude modulation. Sequences are either temporally regular or random. We show that listeners' ability to detect abrupt appearance or disappearance of a stream is facilitated when scene streams were characterized by a temporally regular fluctuation pattern and that this ability is preserved during healthy aging. Remarkably, listeners benefit from regularity even when they are not consciously aware of it. These findings establish that perception of complex acoustic scenes relies on the availability of detailed representations of the regularities automatically extracted from multiple concurrent streams.

9:35

2aPP4. Encoding, reconstruction and representation of natural vocal communication signals. Bradley Theilman (Neurosciences Graduate Program, UC San Diego, La Jolla, CA) and Tim Gentner (Psych., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, tgentner@ucsd.edu)

How complex, natural communication signals are represented in the activity (spiking) patterns of neural populations is not well understood. I will describe data from a series of experiments that examine the spatiotemporal pattern of song-evoked spiking in large populations of simultaneously recorded neurons in the secondary auditory cortices of European starlings (Sturnus vulgaris), a species of songbird. Single neurons in these regions display composite receptive fields that incorporate large numbers (a dozen or more) orthogonal spectro-temporal features matched to the acoustics of species typical songs. This observation implies that the independent spiking response of any single neuron at any point in time is somewhat ambiguous with respect to the stimulus. I will discuss how this ambiguity can be resolved at the population level by considering an alternative representation of underlying latent stimulus space and the coincident pattern of firing among small groups of neurons. I show how this approach allows for high-fidelity reconstruction of spectrographic representations of novel birdsong from spiking responses. I then show how these coincident patterns of spiking can be measured directly in arbitrarily large neural populations, without reference to receptive fields, and conclude by discussing the implications of this combinatorial population encoding for the nature of perceptual representation.

10:05–10:20 Break

10:20

2aPP5. Expectation of self-generated sounds drives predictive processing in mouse auditory cortex. David Schneider (Ctr. for Neural Sci., New York Univ., 4 Washington Pl., Rm. 771, New York, NY 10012, david.schneider@nyu.edu), Nicholas Audette, and WenXi Zhou (Ctr. for Neural Sci., New York Univ., New York, NY)

Behavior exerts a strong influence over sensory responses in the brain. In the auditory cortex, neural responses to self-generated sounds are suppressed, suggesting that prediction may play a critical role in local sensory processing. However, it is unclear whether this phenomenon derives from a precise movement-based prediction or how it affects the neural representation of incoming stimuli. We address these questions by designing a behavioral paradigm where mice learn to expect the predictable acoustic consequences of a simple forelimb movement. Dense neuronal recordings from auditory cortex revealed suppression of neural responses that was strongest for the expected tone during a narrow time window surrounding the specific sound-associated movement. Predictive suppression was concentrated in layers 2/3 and 5, preceded by the arrival of movement information in deep layers, and driven by motor-sensory coupling rather than task engagement, behavioral relevance, or reward association. Despite large-scale suppression at the population level, the auditory cortex retained a sparse representation of the expected self-generated sounds, while a substantial population of auditory cortex neurons in L2/3 and L5 encoded prediction errors. These findings illustrate that expectation, learned through motor-sensory experience, drives layer-specific predictive processing in the mouse auditory cortex.

10:50

2aPP6. Bats use predictive strategies to track moving auditory objects. Angeles Salles (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames Hall, Ste. 232, Baltimore, MD 21218, angiesalles@gmail.com), Clarice Diebold, and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Big brown bats rely on acoustic information from the echoes of their own vocalizations to hunt insects in flight. Unlike many predators that use vision as their primary sensory system, bats compute the 3D position of insect prey from discrete echo snapshots, which yield interrupted sensory information about the target trajectory. Furthermore, this information may be further diminished by clutter in the environment and erratic flight maneuvers of prey. The question arises about the strategies employed by bats to successfully track and intercept their prey. We devised an ethologically inspired behavioral paradigm to directly test the hypothesis that echolocating bats build internal prediction models from dynamic acoustic information to anticipate the future location of moving auditory targets. Collectively, our results demonstrate that the echolocating big brown bat integrates acoustic snapshots over time to build prediction models of a moving auditory target's trajectory and enables prey capture under conditions of uncertainty. Furthermore, we build on past work to propose a new model that describes bat flight trajectories of bats employing predictive strategies.

Session 2aSA

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Acoustic Metamaterials I

Bogdan-Ioan Popa, Cochair
Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin,

Alexey Titovich, Cochair
Carderock Div., Naval Undersea Warfare Center, 9500 MacArthur Blvd., West Bethesda, MD 20817-5700

Nathan Geib, Cochair
Mechanical Engineering, University of Michigan, 1587 Beal Ave. Apt. 13, Ann Arbor, MI 48105

Chair's Introduction—8:00

Contributed Papers

8:05

2aSA1. Asymmetric lateral beaming of sound in a bianisotropic phononic crystal. Xiaoshi Su (Toyota Res. Inst. of North America, 1555 Woodridge Ave., Ann Arbor, MI 48105, xiaoshi.su@toyota.com) and Debasish Banerjee (Toyota Res. Inst. of North America, Ann Arbor, MI)

Willis materials or bianisotropic acoustic materials that couples pressure to particle velocity provide rich physics for sound manipulation. It has recently been shown that tailored Willis acoustic scatterers with resonance effects can display significant cross-coupling between the monopole and dipole responses. In this talk, we show that the multiple scattering in a bianisotropic phononic crystal composed of Willis acoustic scatterers also leads to interesting coupling behavior. The asymmetric structure in our design is non-resonant in the frequency range of interest. However, the nonlocal effect due to the multiple scattering among the scatterers leads to coupling between quadrupole and dipole responses. The phononic crystal is tuned to match the second-order Bragg scattering with two leaky guided modes. We show that a normally incident wave can be steered to propagate along the positive and negative directions perpendicular to the incident direction with different amounts of energy. Simulation and experimental results will be discussed.

8:20

2aSA2. Extreme effective material parameters enabled by non-local Willis coupling. Steven R. Craig (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta 30318, Georgia, scraig32@gatech.edu), Phoebe J. Welch, and Chengzhi Shi (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

While traditional acoustic metamaterial designs based on monopolar and dipolar resonance realize effective densities and bulk moduli that go beyond the those of natural materials, their reliance on local resonance ultimately limits their ability to access extreme effective parameter values. Here, we use an active metamaterial with non-local Willis coupling to drastically extend the accessible range of the effective density and bulk modulus by at least two orders of magnitude compared to a non-Willis metamaterial that relies on local resonance. Our design introduces Willis coupling with two sensor-transducer pairs on both sides of the metamaterial that detects incident waves and superimposes an asymmetric active acoustic signal on an otherwise passive unit cell. The asymmetry of the active signal results from feedback control circuits with unequal gains and/or phases. In our setup, an asymmetric feedback control circuit induces the non-local Willis coupling that potentially increases the effective length of wave propagation and extends the range of effective density and bulk modulus of the metamaterial. Active metamaterials with extreme effective material parameters via non-local Willis coupling will enable improved control over acoustic propagation and yield valuable advancements in biomedical imaging, noise control, and invisibility cloaking technology.

8:35

2aSA3. Willis coupling in micromorphic elasticity. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu), Benjamin M. Goldsberry, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic and elastic metamaterials (AEMM) are artificial materials whose performance is engineered to exceed naturally occurring materials and conventional composites, often leveraging complex microstructural geometry and strong dispersion. Generalized continuum (GC) theories, such as micromorphic elasticity, are extensions of classical elasticity that capture the effects of microstructure via inclusion of internal kinematic variables and/or higher-order gradients of displacement, and have received renewed interest as potential reduced-order models for AEMM. An alternative AEMM modeling framework is the elastodynamic homogenization theory of Willis, which yields an effective medium whose constitutive relations contain stress-acceleration and momentum-strain-rate couplings. Despite the recent popularity of the GC and Willis methods, a comprehensive description of their relationship remains lacking. In the present work, we derive an explicit connection between the Willis and GC models. Specifically, we augment the variational formulation of Mindlin [*Arch. Ration. Mech. Anal.*, **16**, 51–78 (1964)] to account for microscale asymmetry of mass density and demonstrate that the resulting boundary value problem may be expressed in the form of a Willis material. This approach helps to elucidate the physical origins of Willis coupling and may offer insights

about how other emergent material properties arise from microstructural constituents. [Work supported by ONR.]

8:50

2aSA4. Breaking reciprocity through nonlocal coupling in elastodynamic systems. Nathan Geib (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Saint Joseph, MI 48109, geibnath@gmail.com), Bogdan-Ioan Popa, and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

A commonly studied structure in active metamaterial research is an elastodynamic beam outfitted with piezoelectric patches with electronic circuitry used to modify the beam's local material properties. These active beams have been used to validate a variety of conceptual strategies, including those to reduce vibrations, enhance flexural wave sensing, and break reciprocity. Here, we show how the coupling of adjacent piezoelectric patches can be used to create a nonlocal metamaterial featuring the highly nonreciprocal transmission of flexural waves. We discuss how a modified Bernoulli–Euler beam theory was used to design a controller to drive a piezoelectric patch using the signal from a spatially separate patch. We discuss conditions for stability and present experimental data validating model predictions.

9:05

2aSA5. Asymmetric Bragg scattering of water-borne sound by identically oriented asymmetric inclusions. A. J. Lawrence (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, ajlawrence@utexas.edu), Yahui Xue (Dept. of Mech. and Aerosp. Eng., Southern Univ. of Sci. and Technol., Berkeley, CA), Xiang Zhang (Nanoscale Sci. and Eng. Ctr., Univ. of California, Berkeley, Berkeley, CA), Preston S. Wilson, and Michael R. Haberman (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

We report on the modeling, fabrication, and characterization of an innovative design for a heterogeneous material consisting of spherical scatterers with asymmetric material distribution embedded in a thin viscoelastic layer. The material properties of the scatterers are uniformly oriented with respect to the layer thickness. The inclusions consists of unequal parts iron (Fe) and polyetheylene glycol (PEG) with non-uniform distribution of Fe within each inclusion to leading to coupling between monopole and dipole scattering response from individual Fe-PEG scatterers in the long wavelength limit when embedded in a polydimethylsiloxane (PDMS) matrix. Experiments were conducted from 0.5–1.5 MHz in an ultrasonic tank on two different samples.

9:20

2aSA6. Improved sound diffuser performance using spatiotemporally modulated metasurfaces. Janghoon Kang (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, jh3010.kang@utexas.edu) and Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Conventional sound diffuser designs consist of a periodic arrangement of variable depth wells which result in a quasi-random spatial distribution of the reflected phase and minimal energy loss. Diffuser performance depends on geometric parameters such as well depth, number of elements per diffuser period, and in-plane dimensions. We previously showed that coiled-space acoustic metasurfaces (AMS) can significantly reduce diffuser depth while maintaining the performance when compared to classical quadratic residue diffusers (QRD) [Kang and Haberman, *J. Acoust. Soc. Am.*, **149**(4), A141 (2021)]. However, all conventional and AMS diffuser performance is limited by geometry due to the linear relationship between frequency and wavelength. Here we consider spatiotemporal modulation of an AMS to control and improve sound diffusion using a technique inspired by nonreciprocal vibrations in Euler beams [Goldsberry *et al.*, *Phys. Rev. B* **102** (2020)]. A theoretical model is developed using the classical grating equation augmented with a surface impedance that is modulated as a travelling wave function. Properly selected modulation parameters lead to increased diffusion coefficients associated with scattering into harmonics of the temporal modulation frequency and nonreciprocal scattering patterns dependent on the spatiotemporal modulation. Diffuser performance is demonstrated and investigated through the angular dependence of the scattered field magnitude.

9:35

2aSA7. A Green's function approach for nonreciprocal vibrations of finite elastic structures with spatiotemporally modulated material properties. Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bgoldsberry@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The spatiotemporal modulation of material properties in acoustic and elastic metamaterials is one way to realize dynamic nonreciprocity for mechanical wave propagation and vibration of finite systems. A recent study on nonreciprocal vibrations of finite elastic beams has shown that strong nonreciprocal input-output responses resulting from the spatiotemporal modulation only occur for a small subset of modulation parameters, which may complicate the design of nonreciprocal systems [Goldsberry *et al.*, Phys. Rev. B **102** (2020)]. In this work, we derive a Green's function approach for finite elastic systems with spatiotemporally modulated material properties, which mathematically describes the modulated system response for a given set of boundary conditions. As a case example, we investigate two-dimensional flexural vibrations of a plate whose stiffness is modulated in a traveling wave form. We then investigate the first order corrections to the unmodulated system by utilizing an asymptotic expansion in the small parameter defining the modulation strength. Finally, we find conditions on the material and modulation parameters that yield a large degree of nonreciprocity. The present analysis leads to potential applications in acoustic communications, vibration suppression, and energy harvesting. [Work supported by NSF EFRI.]

9:50

2aSA8. Weak-form homogenization of two and three-dimensional systems. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil) and Kyle Dunn (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH)

A homogenization method based on a weak formulation of the dynamic equation for one-dimensional systems was recently developed and published [Muhlestein, *J. Acoust. Soc. Am.* **147** (2020)]. An important aspect of that homogenization method requires the weak form be written explicitly in terms of the acoustic pressure and the particle velocity, which are continuous quantities at all points in one-dimensional systems. However, for two and three-dimensional systems only the normal component of the particle velocity need be continuous at interfaces, and simply averaging the mass density (as was suggested by the one-dimensional analysis) leads to an inaccurate prediction of the effective mass density even in the quasi-static limit. In this paper we describe a more accurate homogenization method for higher-dimensional systems. A critical result is that the particle velocity does not necessarily approach a macroscopic limit, even for quasi-static signals, and that material properties should be interpreted as applicable for an ensemble-averaged particle velocity. Analytical estimates for the effective material properties are given for stratified media. A numerical simulation is then presented comparing the behavior of a stratified system with the behavior of a homogenized system.

10:05–10:20 Break

10:20

2aSA9. Weak-form homogenization of one-dimensional systems with hidden degrees of freedom. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Acoustic metamaterials are synthetic structures that exhibit specific, often exotic, effective material properties for long-wavelength stimuli. One of the principal methods that exotic material properties, such as zero or negative mass density or bulk modulus, may be obtained is through hidden

degrees of freedom. An example of a hidden degree of freedom is a resonator attached in shunt to a duct and modeling the system as just a duct with effective material properties. In this example, the resonator is "hidden" from the model by incorporating its (rather profound) effect on the system into the material properties. This paper demonstrates how hidden degrees of freedom may be incorporated into a recently developed homogenization scheme that utilizes the weak form of the dynamic system [Muhlestein, *J. Acoust. Soc. Am.* **147** (2020)]. Two specific examples will be considered: the shunt resonator, as described above, and a membrane inside a duct. The results are compared with other standard homogenization methods.

10:35

2aSA10. Study of three-dimensional elastic metamaterials based on a lattice system. Zachary Mumme (Eng. Sci., Bethany Lutheran College, 700 Luther Dr., Mankato, MN 56001, zmumme@blc.edu), Jiadi Fan (Aerosp. Eng. and Mech., Univ. of Minnesota, Minneapolis, MN), and Sheng Sang (Eng. Sci., Bethany Lutheran College, Mankato, MN)

In this paper, a three dimensional (3D) mass-in-mass lattice system is proposed to understand how elastic waves propagate through 3D metamaterials. In our model, we included spring elements that protrude in all three dimensions, along with spring elements protruding in only one or two dimensions to properly analyze the system's elastic properties. Due to the dispersion equation containing four variables, it must be graphed in three different perspectives to produce plots in which the stop bands' values can be interpreted with ease. In essence, the stop bands are produced by aligning the graphed structure along each orthogonal plane. By modifying the spring elements' constants, each of the stop bands can be shifted into higher or lower frequency ranges independent of each other, theoretically providing complete control over how a wave propagates through the metamaterial. From this, real world applications can be developed such as absorbing elastic waves in undesired directions, focusing elastic waves to specific points, guiding waves along complicated paths, and creating elastic superlenses.

10:50

2aSA11. Symmetry-protected, zero-energy disclination modes and their observation in an acoustic lattice. Yuanchen Deng (Graduate Program in Acoust., he Penn State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27606, ypd5099@psu.edu), Wladimir Benalcazar (Dept. of Phys., The Penn State Univ., State College, PA), Zeguo Chen (Dept. of Phys., Hong Kong Baptist Univ., Hong kong, Hong Kong), Mourad Oudich (Penn State Univ., State College, PA), Guancong Ma (Dept. of Phys., Hong Kong Baptist Univ., Hong Kong, Hong Kong), and Yun Jing (Graduate Program in Acoust., The Penn State Univ., State College, PA)

Building upon the bulk-boundary correspondence in topological phases, disclinations-crystalline topological defects that induce a curvature singularity in a lattice-have been harnessed for trapping modes within the bulk of synthetic lattices. These trapped modes are useful in photonic and sonic crystals because the bulk surrounding a defect generates a bandgap that isolates the mode from the gapless free space. However, in current realizations, these modes are not protected from delocalization because so far, experimentally realized mode-trapping disclinations have broken chiral symmetry. As such, their modes do not lie at mid-gap and can hybridize with bulk modes to form resonances, losing their confinement. Here, we devise a fundamentally new paradigm that allows the protection of modes bound to disclinations that preserve chiral symmetry. These disclination modes are consequently pinned at the mid-gap, which in turn result in their protected maximal confinement. By presenting a protection mechanism that rests on the interplay between the topology of the lattice and the point group symmetry of defects and by experimentally probing these modes in judiciously designed chiral-symmetric acoustic lattices, our work demonstrates the existence of protected modes within the bulk of synthetic lattices.

11:05

2aSA12. Elastic topological pumping for surface acoustic waves. Mourad Oudich (Penn State Univ., 201 Appl. Sci. Bldg., University Park, State College, PA 16802, mxo5236@psu.edu) and Yun Jing (Acoust., Penn State Univ., State College, PA)

This talk is concerned with Thouless pumping of topological states for surface acoustic waves (SAW). We first designed an elastic waveguide analog of high-index optical fiber, that supports highly confined surface acoustic modes with controllable acoustic index. The "elastic fiber" is made of a silica waveguide embedded in a silicon substrate at the surface, which enables the existence of elastic modes propagating on the surface of the substrate, while being confined along the silica waveguide. The evanescent field outside the silica waveguide can be controlled via the geometrical dimensions of the waveguide which directly affects the coupling between adjacent waveguides. We then constructed a topological structure made of a finite number of elastic waveguides that support topological elastic states, and modulated the coupling between the silica waveguides to demonstrate for the first time Thouless pumping for SAW.

11:20

2aSA13. Topology optimization design of metasurfaces to control Lamb wave propagation. Daniel Giraldo-Guzman (Mech. Eng., Penn State, 132 Reber Bldg., University Park, PA 16802, dzg5526@psu.edu), Cliff Lissenden (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Mary Frecker (Mech. Eng., Penn State, University Park, PA), and Parisa Shokouhi (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Metasurfaces are engineered metamaterials with a structured surface used in electromagnetics, optics, elastodynamics, and acoustics. Elastodynamic metasurfaces can control plate or surface waves. Wave control using metasurfaces has applications across length scales from designing electronics to vibration control systems. Previous works have approached the design of metasurfaces through experimentation and parameter-tuned designs, but no systematic design methodology has been proposed yet. This work addresses the design of resonant metasurfaces to forbid the propagation of plate waves. This is achieved through designing the local resonators using a density-based topology optimization method. Specifically, we design resonators that at their base, fulfill a set of continuity conditions. These are equivalent to boundary conditions mathematically proven to provide reflection and mode conversion for Lamb or Rayleigh waves. Thus, the optimization problem is defined as the minimization of a weighted sum of the displacements and stresses that define such boundary conditions. A topology optimization algorithm is developed that includes material interpolation models, penalization schemes, filtering techniques, sensitivity analysis, and solution methods. A complex scheme integrating the density-based mathematical parametrization, finite element models, data processing modules, optimization solvers, and 3D structural visualization is implemented

11:35

2aSA14. Investigation of geometrical factors effects on sound silencing behavior of mass attached membrane-type acoustic metamaterials. Mohammad Aghamiri (School of Mech. Eng., Univ. of Tehran, 1450 Kargar St. N., Tehran 14399- 57131, Iran, m.aghamiri74@ut.ac.ir), Reza Ghaffarivardavagh (MIT Media Lab, Massachusetts Inst. of Technol., Boston, MA), and Mohammad J. Mahjoob (School of Mech. Eng., Univ. of Tehran, Tehran, Iran)

Membrane-type acoustic metamaterials have drawn significant attention in recent years due to their capacity in generating low frequency band gaps. To date, numerous analytical and numerical approaches have been developed to study and evaluate acoustic performance of this type of metamaterials toward improving their sound silencing behavior. In this paper, we have used point-wise matching method to study different geometrical factors of mass attached membrane and investigate how these factors contribute to the sound silencing behavior. Additionally, the full-wave simulation of these exotic mass-attached membranes are carried out to numerically validate the results. The paper herein, provides an insight on how tuning the geometrical parameter of mass-attached membrane yields broadening the silencing bandwidth. Developing wide-band membrane-type metamaterial pave the way for the emergence of myriad applications ranging from noise mitigation to acoustic energy harvesting.

Session 2aSC

Speech Communication and Psychological and Physiological Acoustics: Development of Sensory-Motor Connections for Speech

Tian Zhao, Cochair
Institute for Learning and Brain Sciences, University of Washington, University of Washington, Box 367988, Seattle, WA 98195

Patricia K. Kuhl, Cochair
UW, Malistop 357920, University of Washington, Seattle, WA 357920

Matthew Masapollo, Cochair
University of Florida, 1225 Center Dr., Gainesville, FL 32610

Chair's Introduction—9:00

Invited Papers

9:05

2aSC1. Sensorimotor information flow and the acquisition of speech. Patricia K. Kuhl (Inst. for Learning & Brain Sci. (I-LABS), Univ. of Washington, Seattle, WA, pkkuhl@uw.edu)

In 1999, Doupe and Kuhl published a review of "Birdsong and Human Speech" for the *Annual Review of Neuroscience* that detailed the commonalities and differences between the two species' vocal learning and the acquisition of species-specific communicative signals. As part of the review, we outlined a model for both species learning, suggesting that an initial period of *sensory* learning was followed by a period of *sensorimotor* learning. We argued that auditory experience establishes memory representations of species-typical auditory input, followed by sensorimotor learning, which is guided by the learned sensory information. In this presentation, I will review both behavioral and neural data suggesting that the model needs to be revised for human speech learning. Data suggest that from the earliest moments of life, sensory and motor information for human speech is interwoven. Bidirectional influences have been documented. I will describe a possible neural mechanism that may account for this sensorimotor information flow, and also a revision of the Doupe and Kuhl model.

9:30

2aSC2. Neural dynamics of the motor system during language development. Maggie Clarke (Inst. for Learning and Brain Sci., Univ. of Washington, 1715 NE Columbia Rd., Seattle, WA 98195, mdclarke@uw.edu), Alexis Bosseler, Tian Zhao, and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Motor and sensory cortices are traditionally thought to be functionally separate systems. However, many studies have shown their roles in both action and perception to be highly integrated. In particular, this has been observed in regard to speech, where listening to speech sounds elicits neural activity in motor regions of the brain. Two datasets containing magnetoencephalography (MEG) data from 2-, 6-, 7-, and 11-month-old infants were used to investigate the role of the motor system in speech perception throughout various stages of language development. We examined the relationship between activation in auditory sensory and motor regions of the brain with respect to: (1) the development of neural responses with increasing age, and (2) potential differences between speech and nonspeech auditory signals. Results showed that motor regions are activated during speech perception across all ages. At 2 months of age, infants show activity in both motor and motor planning regions in response to speech, but not nonspeech, acoustic stimuli. Our data suggest that infants' activation of sensory and motor brain regions in response to speech does not require experience producing speech.

Contributed Paper

9:55

2aSC3. Using MEG to assess the neural mechanisms of phonetic distributional learning and future language growth in 2- and 6-month-old infants. Alexis Bosseler (Univ. of Washington, Seattle, WA, bosseler@uw.edu), Maggie Clarke, Kambiz Tavabi, and Patricia K. Kuhl (Univ. of Washington, Seattle, WA)

Infants are sensitive to the distributional characteristics of the speech they hear, and perception can be altered by the distributional properties of language input. Infants listening to languages with distinct distributional properties show altered perception at the age of 6 months based on learning from ambient language input. Perception can also be altered in the laboratory by manipulating the distributional characteristics of speech in a 2-min exposure. However, the brain mechanisms underlying this learning are not well understood. Here we used MEG to explore the neural processes involved in on-line distributional learning. We examined brain responses to ambiguous speech sounds that straddle a /ba-wa/ speech continuum. Infants were tested longitudinally at 2 and 6 months of age not only to assess whether effects of experience can be documented between 2 and 6 months, but also to determine whether individual differences in brain response can predict future language growth. Our results show that differences in the magnitude of the brain response to ambiguous speech sounds in brain regions involved in auditory speech processing, motor planning, and attention change from 2- to 6-months-of- age. Moreover, we found that individual differences in these regions predict language growth up to 27 months of age.

10:10–10:25 Break

Invited Papers

10:25

2aSC4. Speech perception and the sensorimotor system in Infancy. Dawoon Choi (Psych., Yale Univ., 1 Prospect St., New Haven, CT 06511, dawoon.choi@yale.edu) and Janet F. Werker (Psych., Univ. of BC, Vancouver, BC, Canada)

The speech that infants perceive and learn from is highly multisensory. Preverbal infants show multisensory speech sensitivities prior to direct associative experience, for instance, even to non-native speech that they have not experienced before. Sensorimotor influences on auditory perception are of increasing interest in the context of speech perception development. In a series of experiments, we explored whether infants' speech perception is influenced by articulatory-auditory relations. Specifically, we experimentally restricted the movement of infants' relevant articulators during speech perception tasks. We review a series of studies showing both behavioral (eye-tracking) and neural (EEG) evidence that preverbal infants' auditory speech perception is influenced by articulatorily specific sensorimotor input at 6- and 3- months of age. To control for the possibility of learning, we tested both native and non-native (hence not visually nor auditorily familiar) phonetic contrasts. To explore whether the auditory-sensorimotor relation is in place even without feedback from self-produced vocalization, we tested consonant discrimination in infants as young as 3-month, who are not yet babbling and are unable to produce consonant sounds. Our results show that the sensorimotor-auditory link is in place prior to specific experience watching, hearing, or producing the relevant sounds.

10:50

2aSC5. MIPA: A theory of phonological acquisition and speech motor control. Matthew Masapollo (Univ. of Florida, 1225 Ctr. Dr., Gainesville, FL 32610, mmasapollo@phhp.ufl.edu) and Susan Nittrouer (Univ. of Florida, Gainesville, FL)

Despite long-standing interest in the relation of early speech production and phonological development, little is known about how speech motor skills develop in tandem with emerging phonological representations in childhood. Recent experimental work in our laboratory has shown that better speech motor control at young ages is associated with better outcomes at later ages in phonological sensitivity. In this talk we will advance a new, theoretical model to account for these findings. It is termed MIPA (*M*otor *I*nvolvement in *P*honological *A*cquisition) and proposes that highly developed sensorimotor maps for speech support the acquisition of detailed phonological representations. We will begin by providing an overview of the model, its core predictions, and compare it to existing frameworks. We will then describe experimental studies designed to explore the conceptual merits of the model by testing children longitudinally on a set of measures assessing the coordination of vocal-tract gestures, phonological sensitivity, and the relationship between those two skills. Application of the model to the study and treatment of children with developmental disabilities that affect speech production, such as stuttering, as well as language-related disorders arising from poor sensitivity to phonological structure, such as dyslexia, will also be briefly described.

11:15

2aSC6. A neurocomputational view of developmental stuttering. Frank Guenther (Boston Univ., 677 Beacon St., Boston, MA 02115, guenther@cns.bu.edu)

An estimated 5% of children go through a period of stuttering around 3–5 years of age, with stuttering persisting into adulthood in approximately 20% of these children. Research into the neural underpinnings of stuttering have identified a number of structural and functional anomalies in various components of the left hemisphere cortico-basal ganglia-thalamocortical motor loop in children and adults who stutter. This talk will present the theoretical view that this loop is heavily involved in the *initiation* of speech motor programs (rather than encoding sensory-motor transformations that make up the motor programs themselves) and that stuttering is primarily a disorder of speech initiation rather than one of impaired motor programs for speech. Relevant neural and behavioral findings will be reviewed in light of this hypothesis, including findings that seemingly implicate audio-motor transformation difficulties in stuttering. In the current view, these latter findings are interpreted as evidence of difficulty in using auditory information to determine the appropriate time to initiate the next motor program in a speech sequence.

Session 2aSP

Signal Processing in Acoustics, Physical Acoustics, Noise, Biomedical Acoustics, and Underwater Acoustics: Signal Processing for Non-Specialists

Zoi-Heleni Michalopoulou, Cochair
Department of Mathematical Sciences, New Jersey Institute of Technology, 323 M. L. King Boulevard, Newark, NJ 07102

Eric A. Dieckman, Cochair
Mechanical Engineering, University of New Haven, 300 Boston Post Road, West Haven, CT 06516

Invited Papers

9:00

2aSP1. Signal processing for architectural acousticians. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu) and John P. O'Keefe (O'Keefe Acoust., Toronto, ON, Canada)

Architectural acoustics is characterized by its multidisciplinary nature, the multiple disciplines include physical acoustics, engineering acoustics, noise control, psychoacoustics, and signal processing. In recent decades, signal processing has become an integrated body of fundamental knowledge of architectural acoustics. Furthermore, it has advanced technology in architectural acoustics practice. This talk highlights a number of applications in architectural acoustics in which acoustic signals need to be efficiently manipulated and analyzed. In addition, signal processing tools have drastically advanced measurement techniques in architectural acoustics, advanced spatial and temporal analysis of sound fields in enclosures. Signal processing has entered into a wide range of areas, from room-acoustic energy decay analysis, modal analysis, interaural de-correlation for quantifying auditory spaciousness, array signal processing, three-dimensional rendering techniques for auditory displays, time-frequency analysis of impulse response functions, to optimizations, machine-learning and the concept of Pareto Fronts in architectural acoustics. Architectural Acousticians who are equipped with and understand the basic tools of signal processing are able to contribute to advancing knowledge and perform best practices in the architectural acoustics.

9:20

2aSP2. Human and machine learning expertise applied to speech and music signals processing. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Faculty of ETI, Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

Machine learning techniques provide extensive advances in automated signal and data processing in many fields. One of the critical areas in acoustic signal processing is extracting relevant information from the speech signal. The speech signal contains phonemic variation, temporal structure, prosody, timbre, and voice quality. It also includes various aspects of the speaker's profile, such as emotions or sentiments. What is easily discerned and analyzed by a human escapes a machine learning-based approach when dealing with all that complexity at once. One may easily see parallels to musical signal processing as such aspects as temporal structure, timbre, and music quality, etc., are of importance, especially as music also carries an artistic message. An example of such a functional concept of music is when it is associated with the image, as the ultimate goal of a film music composer is to enhance or evoke the audience's emotions. A question appears whether a machine learning technique may discern emotions associated with a given scene the same way as the human does. This paper focuses on the application of machine learning to speech and music signal analyses. It also emphasizes speech and music signal processing covering recent advances in deep learning.

9:40

2aSP3. Machine learning for acoustic source localization in room and ocean acoustics. Peter Gerstoft (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, gerstoft@ucsd.edu), Haiqiang Niu (Inst. of Acoust., Beijing, China), Michael J. Bianco (Univ. of California, San Diego, La Jolla, CA), Emma Ozanich (Univ. of California, San Diego, Woods Hole, MA), and Yifan Wu (Univ. of California, San Diego, San Diego, CA)

Despite decades of development, acoustic source localization has been challenged by mislocations. Instead, we learn the localizations directly from data using machine learning. After a mapping has been learned we test that the mapping generalizes well on test data. We demonstrate the approach in an ocean waveguide as a classification and a regression neural network, in a room acoustic setting we demonstrate how a neural network can alleviate multipath.

10:00

2aSP4. Speech processing in the age of machine learning: not all models and solutions are created equal. John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, 800 W Campbell Rd., Erik Jonsson School of Eng., EC32 P.O. Box 830688, Richardson, TX 75080-3021, John.Hansen@utdallas.edu)

The speech processing field has evolved dramatically in the last decade, especially for non-specialists using speech technology tools. Machine learning based on various neural networks (DNNs, GANs, CNNs, LSTMs, etc.), have emerged as a way of leveraging available training data to formulate new algorithms which have revolutionized performance for many challenging tasks including speech recognition, speaker identification, speech enhancement, etc. However, with such improved modeling capabilities, greater care is needed to ensure resulting solutions take into account fundamental properties of speech production, language, and perception. This talk provides a brief overview of current speech processing advancements using model based/supervised solutions. In particular, after briefly highlighting several approaches to speech processing, three examples are considered that highlight challenges/flaws/issues in applying supervised model-based concepts and machine learning without factoring in underlying foundations present in the original data context. These areas include: (i) word-count estimation for language assessment of child-adult speech—comparing with LENA language system, (ii) dialect identification using available corpora (e.g., "is the secret in the silence?"), and (iii) speech features for forensic audio analysis with noise and nonlinear distortion mismatch conditions. Suggestions for best practices will be highlighted in speech, speaker, and language based processing.

10:20–10:35 Break

10:35

2aSP5. Time reversal acoustics for communications, focusing, and source localization. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N245 ESC, Provo, UT 84602, bea@byu.edu)

Time reversal is a signal processing method used either to communicate privately between two locations, focus high amplitude energy to a selected spatial location, or to locate and characterize an unknown source. It has been used in acoustics for various applications including secure underwater communications, biomedical applications (e.g., acoustic lithotripsy), and for nondestructive evaluation of structures and mechanical parts. Time reversal techniques require that a calibration signal (i.e., Green's function, impulse response, transfer function, etc.) be obtained between the source and receiver location(s). Assuming that conditions do not change, the calibration signal(s) may be used to send wave energy to the target location to provide a coherent reconstruction of a source or constructive interference at that selected location. This presentation will discuss how time reversal techniques work, introduce various types of processing used, and review various applications of the technique. If that does not sound interesting enough, videos will be shown of LEGO minifigures being knocked over with focused vibrations and targeted candles will be blown out with focused loud sound.

Contributed Papers

10:55

2aSP6. Overview and tutorial on the identification and tracking of pitch and frequency. Sarah R. Smith (Dept. of Elec. and Comput. Eng., Univ. of Rochester, 617 Comput. Studies Bldg., 160 Trustee Rd., RC 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

Many diverse areas of acoustics leverage the identification and tracking of frequency over time. Pitch tracking plays an important role in music information retrieval and performance analysis. Frequency contours are also important features in speech processing and bioacoustics applications, and the identification of frequency content in a signal is widely used. From a signal processing perspective, these applications vary drastically with respect to signal duration and frequency, desired resolutions, and amount of signal interference. In some cases, a single pitch is used to characterize a multicomponent sound, while other applications demand independent tracking of individual components. As such, a wide variety of algorithms exist in both the time and frequency domains. This presentation provides a general overview of commonly used methods and available implementations for tracking pitch and/or frequency. Potential sources of error will be emphasized, focusing on the considerations relevant to selecting and optimizing an algorithm for a given application.

11:10

2aSP7. The Hilbert-Huang transform: Improvements and applications in vibro-acoustics. Trent Furlong (The Penn State Univ., 111 Osborn Hall, University Park, PA 16802, trentfurlong@gmail.com)

The Hilbert-Huang transform (HHT) is a relatively new signal processing technique that offers several advantages over traditional Fourier transform analysis. The HHT empirically decomposes a signal into multiple components called intrinsic mode functions (IMF), then returns the instantaneous frequency and energy for each IMF. Having each IMF's instantaneous frequency and energy for every data sample offers greater frequency resolution in both 2-D and 3-D spectral analysis—a clear advantage when compared to the Fourier transform's well-known time-frequency resolution challenge. Because the HHT performs signal decomposition empirically no model or prior knowledge about the system is needed, making the HHT a useful technique for analyzing and giving physical interpretation to nonlinear, non-stationary systems. The Fourier transform, however, inherently only provides physical interpretation to linear, stationary systems; increasing the required expertise for use on nonlinear, non-stationary systems. This presentation will give an overview on the HHT algorithm, common modifications, advantages and disadvantages, how its results compare to the Fourier transform, and examples of successfully using the HHT on vibro-acoustic data for monitoring machine wear.

11:25

2aSP8. Toward improving road traffic noise characterization: A reduced-order model for representing traffic flow dynamics. 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), Shane V. Lympany, and Matt Calton (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

The National Transportation Noise Map predicts time-averaged road traffic noise across the continental United States (CONUS) based on average annual daily traffic counts. However, traffic counts may vary significantly with time. Since traffic noise is correlated with traffic counts, a more detailed temporal representation of traffic noise requires knowledge of the time-varying traffic counts. Each year, the Federal Highway Administration tabulates the hourly traffic counts recorded at more than 5000 traffic monitoring sites across CONUS. Each site records up to 8760 traffic counts corresponding to each hour of the year. The hourly traffic counts can be treated as time-dependent signals upon which signal processing techniques can be applied. First, Fourier analysis is used to find the daily, weekly, and yearly temporal cycles present at each traffic monitoring site. Next, principal component analysis is applied to the peaks in the Fourier spectra. A reduced-order model

using the first 11 principal components represents much of the temporal variability in traffic counts while requiring only 0.13% as many values as the original hourly traffic counts. This reduced-order model can be used in conjunction with sound mapping tools to predict traffic noise on hourly, rather than time-averaged, timescales. [Work supported by U.S. Army SBIR.]

11:40

2aSP9. Similarity matrices applied to differentiate between Lombard and natural speech. Grazina Korvel (Inst. of Data Sci. and Digital Technologies, Vilnius Univ., Vilnius, Lithuania), Krzysztof Kakol (3PGS Software, Gdansk, Poland), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The study aims to present a method of evaluating speech quality in noise and interference conditions based on similarity matrices. For that purpose, sets of words recorded in the presence of disturbing signals along with their clean counterparts are used. First, it is checked to what extent a correct alignment of signals with and without the Lombard effect is important when self-similarity matrices are utilized in the process of feature discerning. Then, self-similarity matrices based on the acoustic parameters concerning speech prosody and alignment are built. Next, a correlation check is performed for feature redundancy. Based on a reduced set of parameters, 2D maps of acoustic features are created for visualization purposes. This is also performed as the cross-check between different languages with and without Lombard speech. Analyses performed shown that self-similarity matrices may be applied for differentiating speech modified by the Lombard effect and non-Lombard utterances. This research is funded by the European Social Fund under the No 09.3.3-LMT-K-712 “Development of Competences of Scientists, other Researchers and Students through Practical Research Activities” measure.

TUESDAY MORNING, 30 NOVEMBER 2021 — 502 (L)/503 (O), 9:00 A.M. TO 11:40 A.M.

Session 2aUW

Underwater Acoustics, Acoustical Oceanography, Structural Acoustics and Vibration, Animal Bioacoustics, and Signal Processing in Acoustics: Acoustics of Underwater Explosions

Peter H. Dahl, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98115-7834

Ross Chapman, Cochair
University of Victoria, University of Victoria, 3800 Finnerty Road, Victoria, V8P5C2, Canada

Chair’s Introduction—9:00

Contributed Papers

9:05

2aUW1. Acoustic pressure oscillations in the decay envelope of the initial shock radiated by an underwater explosion. Andrew R. McNeese (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, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu), David P. Knobles (Knobles Sci. and Anal., Austin, TX), Peter H. Dahl, David Dall’Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Ross Chapman (Univ. of Victoria, Victoria, BC, Canada)

Cole described the initial shock of an underwater explosion as a monotonically decaying exponential, a model widely used [Underwater Explosions, Princeton University Press (1948)]. When viewed on a time scale that includes bubble pulses, virtually all of the published measurement data also appears to follow the shape of a decaying exponential. The authors have made recent recordings of underwater explosions produced by Sound, Underwater Signal (SUS) devices that reveal repeatable oscillations in the decay of the initial shock. At first, these oscillations were considered to be artifacts of the measurement system, but then two different systems designed specifically to avoid such artifacts were used and the same oscillations were found. Chapman’s data from the mid-1980s, which was not originally published on a time scale that could reveal these oscillations, was reexamined and found to exhibit similar oscillations. Cole describes a similar effect for explosions from multiple charges that are not detonated at the same time, but the authors have found no explanation for this effect from a single mass of explosive material. Examples are shown and a possible explanation is proposed. [Work supported by ONR.]

9:20

2aUW2. Vector acoustic properties of explosive underwater sound. 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)

The relation between acoustic pressure and velocity associated with propagating sound from explosive sources is studied. Data originate from the 2017 Sediment Characterization Experiment where the sound source is

38 g SUS, measurement ranges are order 10 km in waters 75 m; and recent experimental studies on the effects of explosive underwater sound on fish off San Diego where the source is 4.7 kg net explosive weight, ranges are order 100 m in waters 20 m. Horizontal components of velocity are resolved into a single radial component and with the vertical component the intensity vector is derived and associated energy streamlines from simulations are presented. It is also shown how radial component of acoustic velocity is phase- locked with pressure, and nearly corresponds to it upon multiplication by the characteristic impedance; including vertical velocity for total velocity magnitude the correspondence with pressure magnitude is within calibration uncertainty. This self-evident finding concerning high-bandwidth signals is worth noting to inform the discussion on the importance of acoustic velocity to fishes. An exception for short ranges is a low-amplitude, narrow band, Scholte wave arrival that decays precipitously away from the water/sediment interface. [Research support by ONR and by the Navy LMR program.]

9:35

2aUW3. Two-stage algorithm for the simulation of thin shocks in multi-relaxing fluids. William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029, william.willis@utexas.edu), John M. Cormack (Dept. of Medicine, Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA), and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

An efficient two-stage algorithm is presented for simulating a weak shock in an arbitrary waveform propagating in a fluid with multiple relaxation mechanisms. The first stage of the algorithm is based on an evolution equation expressed in intrinsic coordinates for nonlinear propagation in a relaxing fluid [Hammerton and Crighton, JFM (1993)], and it includes effects of spherical spreading. Simulation using intrinsic coordinates allows the pressure waveform to become multivalued, avoiding the need to discretize thin shocks. At selected distances, a shock is inserted into the multivalued pressure waveform according to the equal-area rule, rendering the waveform single-valued. Waveforms calculated in this way agree with corresponding time-domain solutions of a Burgers equation augmented to include relaxation, while simulation with intrinsic coordinates requires orders of magnitude less computation time. An analytical solution for shock evolution [Crighton and Scott, Philos. Trans. R. Soc. A (1979)] is used to estimate shock thickness as a function of propagation distance for determining the appropriate distance at which the second stage, the time-domain solution of the augmented Burgers equation, can be used to continue propagation into the far field. [WAW is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

9:50

2aUW4. Precursor arrivals of explosive sounds during the Seabed Characterization Experiment 2021. Lin Wan (Elec. and Comput. Eng., Univ. of Delaware, 139 the Green, 301 Evans Hall, Newark, DE 19711, wan@udel.edu), Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

Travel time and amplitude of precursor arrivals of explosive sounds in shallow water contain information of marine sediments and could be used for geoacoustic inversion. The precursor arrivals have been identified in the received signals from explosive SUS charges detonated during the Seabed Characterization Experiment (SBCEX2021) conducted in the New England continental shelf and slope region in the summer of 2021. These SUS charges were deployed at various ranges along and across 200-m and 250-m bathymetry contours and recorded by two vertical line arrays (VLAs) positioned at 200-m and 250-m water depths. In addition to the two acoustic VLAs equipped with environmental sensors, there were three environmental moorings simultaneously capturing the water column fluctuation of temperature and salinity. In this research, measured precursor arrivals are first extracted and analyzed. Simulated results are then obtained and compared with the measured ones. Finally, a sensitivity study of the precursor arrivals on water column sound speed and sediment parameters (e.g., sound speed and attenuation) is performed. [Work supported by ONR Ocean Acoustics.]

10:05

2aUW5. Geoacoustic inversion using explosive air gun signals received by a single hydrophone in the East Siberian Sea, Arctic Ocean. Dae Hyeok Lee (Dept. of Marine Sci. & Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan 15588, Republic of Korea, edh0921@hanyang.ac.kr), Jee Woong Choi (Dept. of Marine Sci. & Convergence Eng., Hanyang Univ., Ansan, Republic of Korea), Dong-Gyun Han, Hyoung Sul La, and Eun Jin Yang (Div. of Ocean Sci., Korea Polar Res. Inst., Incheon, Republic of Korea)

Goacoustic inversions using low-frequency impulsive sources in shallow water have been recently performed to estimate geoacoustic parameters of sediments. In this talk, we present the geoacoustic inversion results estimated using explosive air gun data received by a single hydrophone in the East Siberian Sea in September 2019. The single hydrophone was deployed by the Korean icebreaker R/V Araon operated by Korea Polar Research Institute. The geoacoustic inversion is carried out by extracting modal dispersion curves from air gun data received from tens of km. A signal processing method called warping is applied to separate normal mode components and then the extracted dispersion curves are compared to the replicas predicted by the KRAKEN normal mode program. Although it was not possible to compare to *in situ* measurements of sediment, our inversion results are valuable in that they provide indirect information about geoacoustic parameters of the Arctic Ocean. [This research was a part of the project titled 'Korea-Arctic Ocean Warming and Response of Ecosystem (K-AWARE, KOPRI, 1525011760)', funded by the Ministry of Oceans and Fisheries, Korea and supported by the National Research Foundation of Korea (NRF) grant funded by the Korea government (MSIT) (2020R1A2C2007772).]

10:20–10:40 Break

10:40

2aUW6. Estimation of seabed attenuation in the New England Mud Patch. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, gpotty@uri.edu), James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Julien Bonnel, Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, RI), and Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

Measurements of acoustic pressure and particle velocity were made during the Seabed Characterization Experiment (SBCEX-2017) in the New England Mud Patch south of Cape Cod in about 70 m of water. This experimental location is characterized by a "soft" layer of surficial sediment consisting of mud. University of Rhode Island and Woods Hole Oceanographic Institution deployed the "geosled" with a four-element geophone array, a tetrahedral array of four hydrophones, and several hydrophone receive units (SHRUs) as data acquisition packages. In addition, a new low frequency source, interface Wave Sediment Profiler (iWaSP) was deployed to excite interface waves (Scholte waves). The geophone array was localized using the known locations of the acoustic sources and noise from the research vessel. Modal arrivals from broadband sources on geophones and hydrophones were used to invert for compressional and shear wave speeds and attenuation in the mud layer and the underlying sand layer. Effect of shear conversion effects on the modal attenuation estimates and it's frequency dependence will be explored. The role of shear conversion along the mud-sand interface will be discussed. [Work supported by the Office of Naval research, code 322 OA.]

10:55

2aUW7. Vector sensor measurements of dispersed arrivals from explosive signals, underwater sound (SUS). David R. Dall'Osto (Acoust., Appl. Phys. Lab., Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In 2017 and 2021, ONR experiments south of New England examined the effect of the environment on sound propagation. A series of small explosive ordinance called SUS, models Mk-64 (1.1 oz) and Mk-82 (1.8 lbs), were deployed at various ranges (1-40 km) from a vector sensor positioned a 1.5 m above the seafloor. These studies focused on two areas: (1) the New

England Mud Patch (NEMP) where 75 m of water overlay a thick 10-m layer of surficial mud; (2) a 60 km south-southwest of the NEMP where the continental shelf transitions to the slope. SUS provide a strong impulse signal, exciting both water-borne and sediment-borne arrivals. Owing to geometric dispersion, a time-frequency representation of the signals reveal distinct arrivals associated with the propagating normal modes. Forward models based on mode theory are presented to demonstrate the effects of bathymetry, oceanography and sediment structure on dispersed SUS signals. The model facilitates both the extraction of vector information, and inverting it to determine geo-acoustic properties of the environment. Measured vector data are also presented, including examples on the continental slope where the 3D propagation effects encountered will need to be considered for successful modeling and inversion.

11:10

2aUW8. Towards a field deployable alternative to underwater explosive charges. Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX, mcneese@arlut.utexas.edu), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Underwater acoustic experiments and surveys often utilize underwater explosions to produce a high amplitude, broadband event capable of penetrating the seabed and propagating to long range. Although underwater explosive charges have been readily utilized for many decades, an alternative to explosives is of interest due to environmental and logistical concerns. Prior experimentation has demonstrated the viability of the Rupture Induced Underwater Sound Source (RIUSS) that utilizes an expendable diaphragm, known as a rupture disk, as an underwater acoustic source. The device functions by placing a rupture disk over a chamber and mechanically breaking the disk (either by striking on demand or via hydrostatic pressure) at a specified depth to produce high-amplitude, broadband waveforms. Recent experimentation has demonstrated the ability to perform ocean propagation measurements from a small, low cost fishing vessel in the New England Mud Patch by deploying expendable RIUSS units along with small, single channel acoustic moorings. Each of the acoustic moorings and expendable RIUSS units were deployed by hand, allowing the measurements to be gathered with few personnel in a low cost manner. Discussion will focus on recent advancements, fielded deployments, and comparison to related explosive charge propagation measurements. [Work Supported by ONR and ARL:UT IR&D Program.]

11:25

2aUW9. Underwater implosion of re-enforced concrete piers. Paul Donavan (Illingworth & Rodkin, Inc., 429 E Cotati Ave., Cotati, CA 94931, pdonavan@illingworthrodkin.com) and Bruce Rymer (Noise Vib. and Hydroacoustics, Caltrans, Sacramento, CA)

From 2015 through 2018, 18 reinforced concrete piers formally supporting the east span of the San Francisco-Oakland Bay Bridge were removed using underwater demolition. The events ranged from one pier to up three piers at one time with the number of charges ranging from 92 to 688 and weights from 12 to 35 pounds. In all cases, a bubble curtain, blast attenuation system was used to reduce the blast pressures. Although the blast plans varied, more variation in the monitored peak sound pressure levels and sound exposure levels was observed than was expected from initial predictions. Potential causes of this variability were examined including varying charge confinement, directionality due to self-shielding of the structures as the individual blasts progressed, the effect of varying distance to the monitoring location for the individual charges, obscuration of the monitoring location due to intervening structures, and varying effectiveness of the blast attenuation system. Each of these potential causes are examined in this presentation.

Session 2pAA

Architectural Acoustics, Engineering Acoustics, ASA Committee on Standards, and Signal Processing in Acoustics: Current State-of-the-Art and Future Paradigms in Acoustic Measurement and Testing

Ronald Sauro, Chair
NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Spaced object absorbers, sound absorption coefficients, and the noise reduction coefficient rating in architecture. Eric P. Wolfram (Riverbank Acoust. Labs., 1512 S. Batavia Ave., Geneva, IL 60134, ewolfram@alionscience.com)

Spaced object arrays (e.g., vertical baffles) are a growing trend for reverberation time reduction in architectural spaces. Unfortunately, the acoustical considerations for these products are significantly more complex than traditional continuous ceiling treatments. Architects and designers are familiar with sound absorption coefficients (Sabins/ft^2) and the NRC Rating, but these are technically not applicable to spaced object arrays. Further, product manufacturers consider the correct Sabins/Object over 1/3 octave bands to be too technically complex for marketing purposes. The ASTM E33 committee is currently exploring ways to create a single number rating for spaced object arrays. A status update of proposed revisions to ASTM C423 will be shared. IP Units are substituted for SI Units in this abstract for the sake of clarity. [Eric Wolfram serves as Laboratory Manager for Riverbank Acoustical Laboratories and is currently Chair of the ASTM E33 Committee on Building and Environmental Acoustics. https://www.astm.org/COMMITTEE/E33.htm]

1:25

2pAA2. Reproducibility and repeatability: Why acoustics consultants, manufacturers, and academics should care. Evelyn Way (Res. & Development, Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, evelynway@gmail.com)

Trust in acoustic data from all sources is very low. The resulting crisis of confidence in end users, architects, contractors, and designers leads to overly conservative design standards or a sense that "nothing matters so why bother doing anything that might improve the acoustic performance. While this is partly because buildings are complex systems with many interrelated variables affecting the results, part of it is due to normal measurement variation. Study of measurement uncertainty and experimental design began in the late nineteenth and early twentieth centuries, and the fruits of statistical analysis can be found in the repeatability and reproducibility measurements that populate the ASTM standard Precision and Bias statements. Repeatibility and reproducibility can be used for three purposes: 1. to evaluate the quality of the standard, 2. to adjudicate when two testers' measurements disagree, and 3. to evaluate edge cases for compliance with specified values. Unfortunately, the reproducibility and repeatability data in the E33 Building and Environmental Acoustics standards are not typically statistically rigorous enough to provide useful information. By supporting and participating in round robins and interlaboratory studies, the uncertainty in our measurements can be quantified, and we as a community can be confident in our measurement data.

1:45

2pAA3. Improvements to the standards for the testing of noise protection equipment. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This paper will suggest improvements to the applicable testing standards for the testing of hearing protection units. The units can be over the ear or in the ear units. These suggestions will make the tests more objective and less about using people as test objects. We will discuss when sound is hazardous and what time has to do with the level of danger. We will discuss passive attenuation and active attenuation as well as amplification applications to aid in communications between users of such devices.

2:05

2pAA4. Acoustical design of the Maxxon acoustical laboratory. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

This presentation will discuss the acoustical design of the ASTM E492 and ASTM E90 sound testing laboratory for Maxxon, Inc. which was recently completed in Hamel, MN. The primary goal of the project was to create a world-class laboratory that would be able to test floor-ceiling assemblies with STC and IIC ratings as high as 80, with qualified measurements extending from 63 to 10 000 Hz. The presentation will show the results of pre-design site noise and ground vibration measurements as well as techniques that were used for environmental noise control. Reverberation room design and vibration isolation details that were employed to reduce ground-borne vibrations and control flanking will also be discussed. The presentation will also address the design, testing, and installation of low frequency sound absorption panels that were attached to the walls to control excess reverberation at low frequencies. Separate follow-up presentations will discuss the facility construction and the acoustical performance of the completed facility based on preliminary commissioning tests.

2:25

2pAA5. Construction of the Maxxon acoustical laboratory. Evelyn Way (Res. & Development, Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, evelynway@gmail.com) and Jerry G. Lilly (JGL Acoust., Inc., Issaquah, WA)

This presentation will discuss the construction of the Maxxon acoustical laboratory from the initial excavation to final completion, highlighting the problems and solutions that were met along the way. Special attention will be paid to the non-standard construction used for the lab including the acoustic isolation joint, managing services across the isolation joint, specialty equipment, doors, and paint. The selection and implementation of the test system of microphones and sound sources will also be discussed.

2:45–3:00 Break

3:00

2pAA6. Acoustical performance and operation of the Maxxon acoustical laboratory. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com) and Evelyn Way (Eng., Maxxon, Hamel, MN)

This third and final presentation will cover the acoustical performance of the Maxxon facility including: (1) background noise from exterior sources (e.g., truck traffic serving an adjacent business and train traffic from a nearby railroad) and HVAC systems serving the building, (2) sound isolation from nearby spaces including the adjacent work area and control room), (3) reverberation times with and without the low frequency absorbers, (4) ASTM room qualification with fixed microphones in both rooms, and (5) flanking noise limits of the facility. The operation of the laboratory will also be discussed including a description of the trolly mechanism that transports floor-ceiling test specimens from the adjacent work room and storage area through the double doors into and out of the upper reverberation room, test panel installation details, and managing measurement uncertainty.

Contributed Papers

3:20

2pAA7. Experimental evaluation of a sound diffuse field in a ordinary room. Sandrine T. Rakotonarivo (Aix-Marseille Université, CNRS, Centrale Marseille, LMA, LMA - UMR 7031, 4 Impasse Nikola Tesla, Marseille 13453, France, sandrine.rakotonarivo@univ-amu.fr), Vincent Roggerone, Antoine Caisson (Aix-Marseille Université, CNRS, Centrale Marseille, LMA, Marseille, France), and Alexis Bottero (Marine Physical Lab. of the Scripps Inst. of Oceanography, Univ. of California San Diego, La Jolla, CA)

This paper presents a methodology based on analysis of spatial correlation functions for assessing diffusivity of a sound field in reverberant and ordinary reflective rooms. A set of sound field measurements using a microphone array is performed in a room excited by a broadband signal and for many positions of loudspeakers. Each recorded signal is then broken down with a filter bank and the spatial correlation of each obtained band-limited signal is computed. Theoretical predictions of the spatial correlation function and the measured ones show very good agreement when spatial averaging and frequency conditions are met for producing a diffuse field. Evolution of the spatial correlation functions versus frequency and number of spatial averaging, i.e., number of sources positions, are examined in a reflective room. Results show that the proposed technique allows to determine the minimum number of required source positions and the actual lower frequency bound for considering that the field is diffuse. The latter is analogous to the Schroeder frequency for reverberant room. Interest of the presented methodology is its ability to experimentally assess in any reflective ordinary room the actual frequency range over which the field can be considered as diffuse. [Work funded by the AMIDEX Foundation.]

3:35

2pAA8. Beyond reverberation time: Toward a new geometric approach to room acoustics. Zackery Belanger (Arcgeometer, 1111 Bellevue St., Ste. 360, Detroit, MI 48207, zb@arcgeometer.com)

The limitations of the most prevalent parameter in room acoustics—reverberation time—are numerous, and some arise from absorption coefficients measured in reverberation chambers and, more broadly, assumptions of a diffuse field. The result is that material performance metrics and reverberation time prediction—whether calculated or simulated—are reliable over a more limited range of room types than is often recognized in practice. In this work, a new approach that could widen this range is presented via a potential new geometric view of room design. Room geometry has the potential to seamlessly connect dry, diffuse, reverberant, and focusing spaces (e.g., domes, cylinders) into one range of continuous design possibility. Evidence for the efficacy of this approach will be provided via a scrutinization of the behavior of reverberation time equations, including Sabine's, and empirically via room measurements of a space with and without diffusive boundaries: Studio 2 at the Experimental Media and Performing Arts Center in Troy, NY. Insight will also be lent into strange laboratory results, such as variations in measured absorption coefficients based on splitting and spreading of test samples.

3:50

2pAA9. Advances in impact noise insulation testing. Wayland Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com), John Lo Verde, and Samantha Rawlings (Veneklasen Assoc., Santa Monica, CA)

Recent years have seen numerous changes and proposed changes to the measurement and ratings of impact noise insulation testing. New ASTM ratings have been published this year for evaluating low-frequency (E3207) and high-frequency (E3222) impact noise, based on the authors' research [J. Acoust. Soc. Am. **141**, 428-440 (2017)]. This paper will review on-going research in several areas of impact insulation testing, including a modified laboratory method for simultaneously testing multiple floor toppings. The authors have previously published that the existing Impact Insulation Class (IIC per E989) rating be modified to remove the rule limiting the deficiency in any third-octave band to 8 dB ["Reconsidering the 8 dB Rule in ASTM Impact and Airborne Ratings", Noise-Con 2020]. This 8 dB rule has origins in concerns about transmission loss, and most of the analysis on the issue historically has in terms of airborne noise insulation, but the authors show that the consequences for impact testing are more severe. The statistical effects of the rule are also discussed along with the effects on the design of floor-ceiling assemblies. Finally, the statistical uncertainty of laboratory

impact testing continues to be a subject of investigation, and recent results and reviewed. The authors discuss the need to continue to advance testing technology to provide reliable and quality information to those who evaluate, engineer and predict impact noise.

4:05

2pAA10. Comparison between sound insulation parameters obtained through interrupted noise method standard 16283 and new method from standard 18233 from international organization for standardization. Tiago Belletti Romero (Lab., Giner, Rua Joaquim Ferreira, 55, São Paulo, São Paulo 05033080, Brazil, tiago@giner.com.br) and José Carlos Giner (Lab., Giner, São Paulo, São Paulo, Brazil)

The grow of the population density is a reality present in large urban centers. Linked to this, the valorization of the square meter is a common process that leads to the verticalization of buildings and the tendency to smaller dwellings. Such trends are more susceptible to problems related to noise transmission. Therefore, it is essential that some criteria are established to provide a minimal sound insulation between residences. As well as reliable and standardized methods of measuring such parameters for a correct assessment. Thus, this work compares two methods of measuring the acoustic insulation of building systems in loco. Measurements from the interrupted noise method according to ISO 16283 will be compared with measurements obtained with the new method described by ISO 18233: 2006, which first obtains the impulsive response of the system to then obtain sound pressure levels and decay rates. Not only will they be compared for the results obtained, but also for the time taken for the measurements, ease of use and transport of the equipments. Some comparisons with simulated values will also be made using the Sonarchitect software.

4:20

2pAA11. A proposal for a new 1/3 octave band noise criteria. Christopher J. Struck (CJS Labs, 57 States St., San Francisco, CA 94114, cjs@cjs-labs.com)

A proposal for a new noise criteria (NC) method in 1/3 octave bands to supersede the legacy ASA/ANSI S12.2-2019 octave band metric is presented. 1/3 octave band NC curves from NC 65 to NC 5 are derived from the existing octave band curves, adjusted for bandwidth, fit to continuous functions, and redistributed progressively over this space. The proposed 1/3 octave band NC 0 contour is based upon the diffuse field hearing threshold at low frequencies and parallels the existing NC curves at high frequencies. This synthesis is described in detail. NC values are calculated using both the proposed and legacy methods for a number of different room noise spectra. The resulting values were found to be similar for both methods. NC ratings using the proposed method are particularly applicable to very low noise level critical listening environments such as recording studios, scoring stages, and cinema screening rooms, but are shown to also be applicable to higher noise level environments. The proposed method better tracks audibility of noise at low levels as well as the audibility of tonal noise components, while the legacy method as originally conceived generally emphasizes speech interference.

4:35

2pAA12. Modeling a representative room to evaluate sound power measurement methods for ASTM ISR testing. Sunit Girdhar (Mech. Eng. - Eng. Mech., Michigan Technol. Univ., 207 Vivian St., Hancock, MI 49930, sgirdhar@mtu.edu), Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technol. Univ., Houghton, MI), John Lo Verde (Veneklasen Assoc., Santa Monica, CA), and Wayland Dong (Veneklasen Assoc., Santa Monica, California, CA)

ASTM ISR (Impact Sound Rating) test method suffers from non-reproducibility problems, especially in lower frequency bands in smaller rooms where the sound field is non-diffuse. The problem arises from the fact that Sound Pressure Level is measured in these rooms, which is dependent on the room dimensions if the sound field is non-diffuse due to low modal density. Currently, this non-diffusivity is ignored. We propose to measure sound power radiated from the ceiling instead of volume-averaged sound pressure level, which is independent of room dimensions regardless of the type of sound field in the receiving room. This study evaluates sixteen different ideas to measure sound power levels in the receiving rooms through a simulation model of a representative room using Simcenter 3D. Each method's pros and cons in the simulation are used to downselect methods to test in the field.

Invited Paper

4:50

2pAA13. Visual acoustical performance analysis and storytelling. Erik Miller-Klein (Tenor Eng. Group, 113 Cherry St., PMB 52397, Seattle, WA 98104-2205, erik.mk@tenor-eng.com)

With photo-enabled sound intensity systems acoustical practitioners can investigate and understand the root cause of acoustical performance previously unavailable to our industry. This seminar will show examples of measured performances that show the opportunities, limitations, and opportunities for acoustical assessment with this unique technology. See case studies on rapid prototyping, performance validation, and challenges to common "rule of thumb approaches.

Contributed Papers

5:05

2pAA14. Early-stage acoustic optimization of absorptive concrete panels in buildings. Ian Self (Architectural Eng., The Penn State Univ., University Park, PA), Jonathan Broyles (Architectural Eng., The Penn State Univ., State College, PA), Daniel Butko (Architecture, The Univ. of Oklahoma, Norman, OK), and Nathan C. Brown (Architectural Eng., The Penn State Univ., Eng. Unit A, University Park, PA 16802, ncb5048@psu.edu)

Although architectural acoustic design solutions have existed for many centuries, modern advances in material science and construction technology have allowed new possibilities in terms of cost, environmental impact, or systems integration. One design approach with significant architectural potential is to use porous concrete panels to reduce the reverberation and loudness within a space, particularly for larger classrooms, lecture halls, and multipurpose rooms. While porous concrete panels have been previously shown to improve acoustic environments, modifications to mix designs and form geometry can have substantial effects on acoustic criteria such as sound absorption. This presentation begins with a low-resolution acoustic simulation procedure for analyzing concrete panels that is coupled with validating and comparing to previous experimental data. The process then uses multi-objective optimization to shape the concrete panels at the early design stage for the best acoustic performance with the least material, using a parametric model of curved, sawtooth concrete geometry. The low-resolution model validation utilizes raytracing and image sourcing analyses to obtain the Reverberation Time, from which approximate absorption coefficients and corresponding NRC ratings of the concrete panels can be calculated and

compared to experimentally derived data from physical concrete panels. Optimal panel shapes are then recommended for improved acoustic performance.

5:20

2pAA15. Adaptive thermal comfort factors in classroom acoustics. David M. Ogoli (Architecture, California Baptist Univ., 8432 Magnolia Ave., Riverside, CA 92504, DOgoli@calbaptist.edu)

Climate change is affecting the built environment with a growing need to balance adaptive comfort models by relating ambient room temperature and outdoor climatic factors, on the one hand, with adequate classroom acoustics, on the other hand. The adaptive approach allows occupants of a space to adjust their clothing, activity and posture in a space to minimize thermal discomfort while strategizing for adequate reverberant energy and speech clarity. A parametric digital classroom model was built and simulated to various thermal comfort conditions while simultaneously observing the reverberation times (RT) to determine how quickly sound decayed in the space. Building materials were tested parametrically with variable thermal transmittance (U-values) analyzed with corresponding changes in reverberation times. Observed outdoor environmental variables included mean monthly outdoor temperature and relative humidity. A classroom with thirty University students were observed. Measured and simulated data were analyzed for the parametric model applying scatter plot of utilizing the ASHRAE seven-point predicted mean vote (PMV), percentage people dissatisfied (PPD) and reverberation time measurements. Initial findings assessed using *ANSI/ASHRAE Standard 55* indicate that occupants of a space tend to reach adaptive thermal comfort in a wider range of conditions than commonly noted but inadvertently negatively affect good classroom acoustics.

TUESDAY AFTERNOON, 30 NOVEMBER 2021 — 602, 1:00 P.M. TO 2:35 P.M.

Session 2pAB

Animal Bioacoustics and ASA Committee on Standards: Classifying and Quantifying Natural Soundscapes II

Michael Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Invited Paper

1:00

2pAB1. Underwater noise produced by anthropogenic activities on Vancouver Island, British Columbia. Kelsie Murchy (Biology, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3N5, Canada, kmurchy@uvic.ca), Svein Vagle (Inst. of Ocean Sci., Fisheries and Oceans Canada, Sidney, BC, Canada), and Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada)

In recent decades shipping traffic has been increasing, leading to elevated ambient underwater noise. Extensive research has been conducted on the changes to ambient noise levels of moving ships, but little is known for ships at anchor. Vancouver Island, British Columbia has many anchorage locations where freighters stop prior to entering the Port of Vancouver. Additionally, Vancouver Island has logging activities that occur in estuaries and surrounding waters. These human activities raise concern about what impacts they might be having on the soundscape and marine organisms that inhabit these key locations. Cowichan Bay, BC is an industrialized bay and a key migration corridor for Pacific Salmon (*Oncorhynchus spp.*). To understand changes to the ambient noise levels in Cowichan Bay, with different anthropogenic activities, seven stationary hydrophones were deployed during Fall 2019 and 2020. Results show substantial changes in the soundscape with both anchored freighters and logging activities for the duration of their presence in the bay, with elevated SPL detected throughout the bay for anchored freighters. Our results demonstrate the impact anchored freighters and logging activities have on underwater soundscapes and are the first step in understanding the impact these activities have on marine organisms and important ecosystems.

1:20

2pAB2. The relationship between male humpback whale song chorusing and whale abundance on the Hawaiian breeding ground. Anke Kügler (Marine Biology Graduate Program, Univ. of Hawai'i at Mānoa, 2525 Correa Rd., HIG 131A, Honolulu, HI 96822, akuegler@hawaii.edu), Marc Lammers, Eden J. Zang (Hawaiian Islands Humpback Whales National Marine Sanctuary, Kihei, HI), and Adam Pack (Departments of Psych. and Biology, Univ. of Hawai'i at Hilo, Hilo, HI)

Passive acoustic monitoring (PAM) with autonomous bottom-moored recorders is widely used to study cetacean occurrence, distribution, and behaviors, as it is less constrained by factors that often limit other traditional visual observation methods, such as weather and accessibility. During the breeding season, male humpback whales produce an elaborate acoustic display known as "song." The typical asynchronous chorusing of numerous singing males at any one time can provide challenges for monitoring abundance using PAM. Chorusing becomes the dominant source of low frequency (0–1.5 kHz) noise in the marine soundscape in Hawai'i and seasonal levels mirror the whales' migratory patterns. However, the relationship between chorusing levels and overall whale numbers, including non-singing whales (e.g., mother-calf pairs and juveniles), has remained poorly defined. We combined long-term PAM conducted between 2014/15 and 2020/21 off West Maui with concurrent visual land- and vessel-based observations. We found that daily median root-mean-squared sound pressure levels (RMS SPLs) correlate strongly with whale numbers (land: $0.71 \leq R^2 \leq 0.76$, vessel: $0.81 \leq R^2 \leq 0.85$ for three different PAM locations). Applying these results, we were able to use PAM to document significant population fluctuations between 2015 and 2021, as well as study habitat use patterns off West Maui.

1:35

2pAB3. A comparison of transient killer whale (Orcinus orca) whistles in Alaska. Kelly A. Newman (Univ. of Alaska, CFOS, 2150 Koyukuk Dr., 215 O'Neill Bldg., Fairbanks, AK 99709, knewman@uaf.edu)

Several species of dolphins in both isolated and continuous regions exhibit geographic variation in whistles. Although whistles are thought to be learned, possible structural components are exhibited across communities. Whistles from three transient killer whale (*Orcinus Orca*) groups, Prince William Sound, the Pribilof Islands, and False Pass were compared for spectral and temporal properties. Of 395 randomly selected frequency contours of whistles, there were 32 shape categories of whistles. Overall, whistle shapes from different areas reflect homogeneity, with four whistle shapes comprising 50% of all repertoires. The most common whistle contour shapes were up, down, flat, and oscillatory. Whistles from at St. Paul Island, described here for the first time, had higher fundamental frequencies and shorter durations than whistles from Prince William Sound and False Pass. Surface whistles from St. Paul Island had higher mean frequencies than those recorded at depth with passive acoustics. One identical whistle type was found at both St. Paul Island and False Pass, which suggests the same whales travel between the two areas. The longest whistles and most variable whistles were recorded at False Pass.

1:50

2pAB4. Acoustic scene modeling for echolocation in bottlenose dolphin. YeonJoon Cheong (Mech. Eng., Univ. of Michigan, Ann Arbor, 2350 Hayward St., Ann Arbor, MI 48109, yjcheong@umich.edu), K. Alex Shorter, and Bogdan-Ioan Popa (Mech. Eng., Univ. of Michigan, Ann Arbor, Ann Arbor, MI)

The biosonar of bottlenose dolphins has been shown to have excellent target discrimination performance in cluttered environments, but how the animals process the flow of information used in their adaptive searching process is still an open question. In this work, we present a physics-based model of the process echolocating dolphins may use during target discrimination tasks and validate the model with experimental results. Assuming that dolphins emit biosonar clicks and continuously compare the return echoes with learned representations of echoes from familiar objects, we define a likelihood parameter, a metric that quantifies this comparison. We show that the dolphins' adaptive search behavior during echolocation can be interpreted as an effort to maximize the likelihood parameter. Our model is validated experimentally using target discrimination tasks in which a dolphin correctly/incorrectly located a given target in the presence of distraction objects in an open lagoon while it was wearing eyecups. The results show that the dolphin positioned itself where the likelihood parameter was maximized demonstrating that our model explains well the dolphin's behavior during the tasks. [Work supported by the Department of the Navy Grant No. N00014-18-1-2069.]

2:05–2:35
Panel Discussion

Session 2pAO

Acoustical Oceanography and Signal Processing in Acoustics: Acoustical Oceanography Using Ocean Observatory Systems III

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

Andone C. Lavery, Cochair
AOPE, Woods Hole Oceanographic Institution, 98 Water Street, Woods Hole, MA 02543

Felix Schwock, Cochair
Electrical and Computer Engineering, University of Washington, 185 Stevens Way, Paul Allen Center – Room AE100R, Seattle, WA 98195

Contributed Papers

1:00

2pAO1. Remote and autonomous platforms for measuring broadband backscatter. Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA, alavery@whoi.edu), Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, Seattle, NJ), and Robert Petitt (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

The use of broadband acoustic sonars in acoustical oceanography studies in a broad range of environments, particularly for ecological studies, has rapidly overtaken the use of traditional narrowband sonar systems. Almost as importantly, and often underrecognized, is the diversity of platforms that have been used to support these measurements. In this talk, we present broadband acoustic backscatter collected from a number of platforms, including AUVs, a variety of towed and profiling systems, and a bioacoustics mooring. The primary focus of this talk will be on the data collected from REMUS 600 and REMUS 100 AUVs, and the types of science questions that these platforms, outfitted with broadband scientific echosounders, best support. This work was supported by the Ocean Acoustics and Task Force Ocean Programs at the Office of Naval Research.

1:15

2pAO2. The cabled observatory vent imaging sonar deployed on the Ocean Observatories initiative's cabled array. Guangyu Xu (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, guangyux@uw.edu), Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Karen Bemis (Marine and Coastal Sci., Rutgers Univ., NB, NJ), and Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The Cabled Observatory Vent Imaging Sonar (COVIS) was installed on the Ocean Observatories Initiative's Cabled Array observatory at ASHES hydrothermal vent field on Axial Seamount in July 2018. Diffuse hydrothermal flows are identified by maps made using the standard deviation of the phase change between pings separated in time by fractions of 1 s. The results demonstrate significant influences of ocean tides and bottom currents on diffuse hydrothermal discharge. The same data are used to estimate diffuse heat flux, providing two-dimensional maps of heat flux density at a rate of one per day. Sonar data are also used to generate three-dimensional backscatter images of the buoyant plumes above major sulfide structures. These backscatter images show substantial changes in plume bending in the presence of ambient currents and potentially the variations of outflow fluxes. The intensity of acoustic backscatter decreases significantly for highly bent plumes as compared to nearly vertical plumes, reflecting enhanced mixing of plume fluids with seawater driven by ambient currents. A forward model of acoustic backscatter from a buoyancy driven plume yields a reasonable match with the observation, paving the way for inversely estimating the source heat flux of a hydrothermal plume. [Work sponsored by NSF.]

1:30

2pAO3. Quantification of deep sound scattering layers using a moored upward-looking broadband split-beam echosounder. Zhaozhong Zhuang (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS 16, Woods Hole, MA 02543, zhuangz@mit.edu) and Andone C. Lavery (Woods Hole Oceanographic Inst., Woods Hole, MA)

Distinct deep sound scattering layers are prevalent throughout the ocean mesopelagic zones (200–1000 m depths). These deep sound scattering layers are often characterized by daily vertical migration, in which many of the inhabitants of the ocean mesopelagic communities migrate to shallow waters during the night to feed, and descend during the day to avoid predators. And yet little is known about the impact of mesoscale physical oceanography, such as warm core rings or Gulf stream meanders, on the patterns of daily vertical migration or on the small-scale spatial distribution, or patchiness, of the deep sound scattering layers. To address these questions, an upward-looking broadband split-beam echosounder (36 kHz–44 kHz) has been moored at 580 m depth in the slope waters south of the New England shelf break in a location in which warm core rings are a prevalent feature. Data from a 3-day test deployment are analyzed in terms of target density, spectral characterization, patchiness and timing of the migrations. The calibrated broadband spectra of individual targets are used to determine the number of swimbladdered fish in the deep sound scattering layers. Meanwhile, two other methods, including echo-counting and echo-statistics, are also applied to give target density estimation.

1:45

2pAO4. Summarizing long-term changes in daily echogram patterns observed by moored echosounders in the U.S. Ocean Observatories Initiative network. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu) and Valentina Staneva (Univ. of Washington, Seattle, WA)

Long-term deployments of autonomous echosounders deliver rich data embedded with information about mid-trophic level organisms of the marine food web. As a remote sensing tool, moored echosounders have played an important role in observing temporal changes of animal distributions in the water column over large temporal scales, ranging from months, seasons to

years. Taking advantage of the power of matrix decomposition techniques in exploiting regularity in the data to automatically discover low-dimensional structures in large data sets, we apply Principal Component Pursuit (PCP) and temporally smooth Nonnegative Matrix Factorization (tsNMF) in a time-windowed fashion to analyze data from a network of moored, upward-looking echosounders deployed by the U.S. Ocean Observatories Initiative (OOI) from 2015 to 2018. We show that these echosounder time series are low-rank in nature and can be reproduced by a time-varying linear combination (activation) of a small number of distinct daily movement patterns (components). The components and the associated activation jointly provide a compact representation that is suitable for visualization and systematic analysis of mid-trophic animal activities in response to environmental changes, such as those from seasonal changes and extreme weather events.

2:00

2pAO5. Abstract withdrawn.

2:15

2pAO6. Passive acoustic monitoring of hydrothermal vents at the endeavour hydrothermal vent field. Brendan Smith (Oceanogr., Dalhousie Univ., Halifax, NS B3H 4R2, Canada, brendan.smith@dal.ca) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Passive acoustics may provide a method for detection and long-term monitoring of hydrothermal vents. Direct measurement of vent activity can be challenging as vent plumes are often high temperature (i>300°C) and acidic. The discovery of new vent sites can also be challenging as detection of vent plumes or high-resolution seafloor mapping is required. For these reasons, remote detection and monitoring with passive acoustics can be advantageous. In this presentation, long-term passive acoustic data recorded in 2019 from an Ocean Networks Canada (ONC) bottom-mounted hydrophone in the Endeavour Hydrothermal Vent field are presented. A 6-month time window from April through October 2019 was selected for analysis, when fin whale calls were less prevalent. Statistical noise level metrics (e.g., the median) were used to reduce the influence of transient, infrequent events such as fin whale calls and ship noise, as the vent signal was expected to be relatively constant compared to these sources. Spectral analysis of the resulting time series across multiple frequency bins shows evidence of tidal-period variations in power spectral density which may be related to hydrothermal vent dynamics. Comparison is made to other nearby ONC sensors such as vent temperature, bottom pressure, and acoustic doppler current profiler measurements.

2:30–2:45 Break

2:45

2pAO7. Boundary Pass underwater listening station cabled hydrophone array system. David E. Hannay (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC, Canada, david.hannay@jasco.com), Art Cole, and Jack Hennessey (JASCO Appl. Sci., Dartmouth, NS, Canada)

An advanced cabled real-time underwater listening station was deployed in May 2020 in Boundary Pass, British Columbia, Canada, to measure underwater radiated noise of commercial vessels and to detect and track vocalizing marine mammals. This system was sponsored by Transport Canada and is operated jointly by Vancouver Fraser Port Authority and JASCO Applied Sciences (Canada) Ltd. The system consists of two synchronized tetrahedral hydrophone arrays, each with 1.65 m hydrophone spacings, deployed 300 m apart on the seabed in 190 m water depth. The hydrophone frames are connected to shore by two 2.8 km fiber-optic cables, and also support ADCP and CTD sensors, sound projectors for calibrations, and video cameras. The Boundary Pass location was chosen because of its relatively deep water and because the major inbound and outbound commercial shipping lanes leading to Vancouver become narrow and adjacent here, allowing acoustic measurements of ships passing in both directions. In this presentation we will discuss the system technical design and sensor specifications. We will also discuss its deployment and the data processing and presentation systems, with a brief overview of the substantial measurements obtained in the first full year of operation.

3:00

2pAO8. OOIPy: A Python toolbox for accessing and analyzing sata from the Ocean Observatories Initiative. Felix Schwock (Elec. and Comput. Eng., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, fschwock@uw.edu), John Ragland (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA), and Shima Abadi (Univ. of Washington, Seattle, WA)

The recent advent of long-term, large-scale, and publicly available underwater acoustic datasets can unlock new discoveries and fuel a deeper understanding of the ocean environment. One such dataset is provided by the Ocean Observatories Initiative (OOI), who maintain multiple research arrays distributed across the Pacific and Atlantic Ocean that feature a variety of instruments and platforms from hydrophones to water column profilers and surface buoys. However, analyzing the OOI data can be challenging, as tools for accessing the data are either completely lacking, or require a significant amount of additional programming effort to prepare data for analysis. With the development of the Python package OOIPy we aim to address these problems for the OOI data products. OOIPy combines low level functions for accessing OOI data in Python with higher level functions for data processing, analysis, and visualization. Currently, data products from low-frequency and broadband hydrophones, which are only available through the OOI raw data server, as well as conductivity, temperature, depths sensors in the northeast Pacific Ocean are supported. Our vision for the future is to expand OOIPy and allow scientists to focus on the analysis, rather than the accessing and preprocessing, of OOI data products. [Work supported by ONR.]

3:15

2pAO9. Tagging ocean soundscapes with machine learning and topological approaches. Kayla Thilges (Appl. Res., Acoust. LLC, 1500 Westlake Ave. N, Ste 001, Seattle, WA 98109, kayla.thilges@ariacoustics.com), Meredith Plumley (Appl. Res., Acoust. LLC, Seattle, WA), Justin McMillan (Appl. Res., Acoust. LLC, Madison, VA), and Jason E. Summers (Appl. Res., Acoust. LLC, Madison, VA)

With a multitude of sources simultaneously contributing to the ambient ocean soundscape, it is important to disaggregate noise source contributions for better monitoring and assessing noise impacts. Our work adopts a modular set of neural networks and topological pipelines to analyze multiple representations of time-series samples of the ambient ocean noise. Specifically, a convolutional neural network (CNN) is used to broadly tag the presence of local shipping and marine mammal vocalizations from spectrogram representations. Multiple approaches are investigated using topological data analysis to predict a class label of the ship hull type or specific marine mammal species using the audio data as input. The datasets used in training, testing, and validation comprise hydrophone recordings across varying sensors and ocean environments to address the wide variation in ambient background noise and propagation paths. Algorithm performance is characterized by classification accuracy on labeled data. This approach demonstrates a generalized tagging cabilitility of the presence of marine mammals and local shipping activity.

3:30

2pAO10. Reconstruction of sparse ocean noise fields with generative neural networks. Kayla Thilges (Appl. Res., Acoust. LLC, Seattle, WA), Meredith Plumley (Appl. Res., Acoust. LLC, 1500 Westlake Ave. N, Ste 001, Seattle, WA 98109, meredith.plumley@ariacoustics.com), Zaki Zuberi, Sam Delmerico, Craig Einstein, Adith Ramamurti, and Jason E. Summers (Appl. Res. in Acoust. LLC, Madison, VA)

Modeling the underwater noise field offers important insights for underwater navigation, assessing noise impacts, and establishing noise budgets. *In situ* measurements of the noise field are often sparse with limited information about spatially distributed underwater acoustic fields. In this work, we demonstrate the use of a generative neural network for estimating noise fields over regions not directly measured. An ARiA developed ambient noise model is used to generate noise fields for various ocean noise-source scenarios. The model output includes vertical directionality, depth dependence, and incorporates multiple sources of sound, including anthropogenic, biologic, and environmental. We train generative models on these simulated

spatial noise distributions a priori. These generative models are able to take sparsely sampled noise measurements and construct the spatial distribution of the noise field. The generative model learns the spatial distribution by minimizing the point-to-point difference between the generated fields and the true fields at each step while also maximizing the fidelity of the general distribution of the generated noise fields to the true fields. Through a nested generative model approach, along with auxiliary sound-speed-profile information, the local field is used to extrapolate the noise field to larger spatial scales.

3:45

2pAO11. Ship detection from passive underwater acoustic recordings using machine learning. Alejandro Alvaro (Ocean Eng., Florida Atlantic Univ., 7461 Plantation Rd., Plantation, FL 33317, aalvaro2018@fau.edu), Felix Schwock, John Ragland (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA), and Shima Abadi (Univ. of Washington, Seattle, WA)

The Ocean Observatories Initiative (OOI) hydrophone data contains a rich collection of ship passage events that can be leveraged for passive ship detection. This study uses 6 years of OOI acoustic data collected by the hydrophone located at the southern foot of the Axial Seamount volcano with a sampling rate of 200 Hz. Ship location data from the Automatic Identification System (AIS) is used to predict the presence or absence of ships within an effective radius of the hydrophone, estimated to be 15 km. After, identifying ships with similar acoustic profiles, such as commercial carrier, tanker, cargo, container, and supply ships, a balanced dataset is created with an even distribution of ships within and outside of the estimated radius. The final dataset is separated into a training (70%), validation (10%), and test (20%) dataset, and then used to create k-nearest neighbors (KNN) and logistic regression models using the scikit-learn Python library. The KNN model achieves the best performance and classifies the ships with an accuracy of 98.04%. This shows that it is possible to determine the presence of ships from passive underwater acoustic recordings with high accuracy. [Work supported by the SURIEA program and ONR.]

4:00

2pAO12. The neural adjoint method and its application to sound speed learning. Jihui Jin (Elec. and Comput. Eng., Georgia Inst. of Technol., 756 W Peachtree St. NW, Atlanta, GA 30308, jihui@gatech.edu), Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Priyabrata Saha, Justin Romberg, Saibal Mukhopadhyay (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ray-based acoustical tomography typically provides estimates of local ocean sound speed profile (SSP) fluctuations from precise measurements of acoustic travel times (AT) fluctuations (with respect to a reference environment) between multiple sources and receiver arrays along with a model of the ray propagation in the reference (i.e., fluctuation-free) environment. Classically, inverting the forward model (if available) yielding SSPs from ATs can be done by computing the pseudo-inverse (expensive) or using an iterative-method (e.g., gradient descent); however, including any acoustic forward model (e.g., ray-tracing) in an optimization loop is non-trivial because the nonlinear mapping between SPP and AT is computationally expensive to differentiate. Instead, we use a Neural Adjoint (NA) approach [Ren *et al.*, NeurIPS (2020)] to circumvent this problem by replacing the physics-based forward model with a deep neural network approximation, allowing for an inexpensive way to compute a gradient through backpropagation. The iterative nature of the NA method permits thorough exploration of the solution space depending on the initialization, as opposed to outputting a single point estimate. Here, we continue the discussion of data-driven methods presented in the companion presentation by Saha *et al.* and show that NA has the potential to further refine existing data-driven SSP estimation techniques.

Session 2pBAa

Biomedical Acoustics, Signal Processing in Acoustics and Engineering Acoustics: Nonlinear Acoustic Propagation: Theory, Simulations, Experiment, and Applications II

Vera A. Khokhlova, Cochair
Physics Faculty, University of Washington/Moscow State University, Moscow 119991, Russian Federation

Mark F. Hamilton, Cochair
Walker Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1591

Contributed Papers

1:00

2pBAa1. Topological deviation of membrane waves as a model of gravitational lens effect. Stefan Catheline (INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, stefan.catheline@inserm.fr), Victor Delattre, Gabrielle Laloy-Borgna (INSERM U1032, Lyon, France), Frederic Faure (Institut Fourier, Univ. of Grenoble, Grenoble, France), and Mathias Fink (Institut Langevin, espci, Paris, France)

Elastic membranes are often used as didactic demonstration of gravitation from the general relativity perspective. Indeed, trajectories of rolling spheres such as billiard balls influence each other through the deformation their mass print within the membrane tissue as would the space-time curvature of gravity. The analogy is pushed here using membrane waves. Indeed, rules of topology apply to masses and waves. We show through experiments and curved manifold simulations that wave propagation due to topological deviation of a two-dimensional flat fabric membrane, justifies classical analogy with gravitational lenses. This allows revisiting through membrane waves, the famous 1919 Eddington experiment that demonstrated light deviations of stars in the vicinity of the sun. It is our hope that the experiment described here will provide new insight for the testing of geometry effects on wave propagation. Also, inspired by acoustic metamaterials, it is our hope to use a more correctly constructed membrane to observe quantified exchanges of geometric waves with their elementary mesh.

1:15

2pBAa2. Determination of nonlinearity parameter B/A of liquids by comparison with solutions of the three-dimensional Westervelt equation. Lonnie Chien (Eng., Swarthmore College, 500 College Ave., Singer Hall, Swarthmore, PA 19081, lchien1@swarthmore.edu), John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Dept. of Medicine, Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA), E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA), and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Though nonlinear propagation depends upon the nonlinearity parameter B/A of the medium, it is often difficult to measure accurately. Thermodynamic methods often require high temperature and pressure excursions that can damage fragile molecules. Finite-amplitude methods often require unrealistic assumptions of ideal piston sources to account for diffraction effects. An alternative, described here, is to use the numerical solution of the three-dimensional Westervelt equation for weakly nonlinear fields (considering up to 10 s of harmonics of the fundamental frequency), and to match measured harmonic generation with that for simulated media. In particular, we follow Jafarzadeh *et al.* [*Ultrasound Med. Biol.* **47**(3), 809–819 (2021)] to find that the nonlinearity parameter of a liquid can be recovered by determining the best match between measurements and simulations of media with different B/A values. Ideal piston sources with radial symmetry are not required, only a high-spatial-resolution hydrophone scan of the near-field source plane. Other techniques to determine the nonlinearity parameter by comparing propagation models with experiments, such as Richard *et al.* [*New J. Phys.* **22**, 063021 (2020)], may improve on earlier measurement methods.

1:30

2pBAa3. Non-linearities under highly focused high-intensity ultrasound fields. Pradosh Pritam Dash (School of 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)

High-intensity focused ultrasound (HIFU) has seen widespread clinical adoption as a therapeutic tool, as it may be targeted noninvasively and without ionizing radiation. The converging sound waves induce thermal (e.g., ablation) and mechanical (e.g., radiation force) effects that may be localized to manipulate or destroy tissue. In addition to these primary effects, finite-amplitude acoustic influences (e.g., second harmonic generation) become relevant due to the high-pressure levels near the focal region. One of the ways these nonlinear effects occur is when a pressure wave develops at the difference frequency due to the nonlinearity of the medium when an acoustic beam is driven by a signal that contains two high but slightly different frequencies. This is termed as "scattering of sound by sound," an implication of finite-amplitude propagation is the existence of sum and difference frequencies. Using highly focused ultrasound beams as carrier waves at high amplitudes, we have observed that these nonlinear effects can be localized at scales below its wavelength with sufficient amplitudes. These effects were observed for a water reference medium, and thus are likely to be even more prominent in tissue. Exploring these higher-order acoustic effects for the diagnosis and treatment of human diseases is warranted.

1:45

2pBAa4. Examining influence of primary frequency ratio on distortion product otoacoustic emissions generation using an alternating frequency/time domain cochlear model. Haiqi Wen (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30313, hwen@gatech.edu), Julien Meaud (Georgia Inst. of Technol., Atlanta, GA), Wei Dong, and Sebastiaan Meenderink (Loma Linda Univ., Loma Linda, CA)

Distortion product otoacoustic emissions (DPOAEs) are sounds measured in the ear canal in response to a two-tone stimulus. Because DPOAEs are measured non-invasively, they have been intensively studied and widely applied for the clinical assessment of hearing function. In this work, we combine a three-dimensional computational model with mechanical-electrical-acoustic coupling with *in vivo* measurements of the pressure in scala tympani and the extracellular electrical outer hair cells (OHCs) response to

study distortion product (DP) generation by OHCs in the gerbil cochlea. An efficiency non-linear frequency domain algorithm is used to simulation the nonlinear response of the cochlea in response to a two-tone stimulus. The influence of primary frequency ratio (on IDP generation and propagation is examined. The results show the IDPs are broadly generated by OHCs but their contribution to OAEs are spatially restricted. This work not only improves our understanding of the role of the OHC electromotility in DP generation, but will also benefit clinical noninvasive hearing assessment based on otoacoustic emissions.

2:00

2pBAa5. Generation of a focused nonlinear difference frequency using high-amplitude time reversal focusing of airborne ultrasound. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N245 ESC, Provo, UT 84602, bea@byu.edu) and Carla Wallace (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) focusing of high-amplitude airborne ultrasound can be used to create a difference frequency. Methods are employed to increase the amplitude of the focus. These methods include creating a miniature reverberation chamber, using multiple sources, and using the clipping processing method. The use of a beam blocker to make the sources more omnidirectional is also examined, and it is found that for most source/microphone orientations, the use of a beam blocker increases the amplitude of the focus. A high-amplitude focus of 134 dB peak re 20 μPa sound pressure level with a center frequency of about 38 kHz is generated using eight sources. Then, using four sources centered at 36.1 kHz and another four sources centered a 39.6 kHz, nonlinear difference frequency content centered at 3.5 kHz is observed in the focus signal. The difference frequency amplitude grows quadratically with increasing primary frequency amplitude. When using beam blockers, the difference frequency content propagates away from the focal location with higher amplitude than when beam blockers are not used. This is likely due to the differences in the directionality of the converging waves during TR focusing.

2:15

2pBAa6. Experimental measurement for assessing pseudo-sound effect in parametric array. Jiyoung Song (Korea Maritime and Ocean Univ., 727 Taejong-ro, Busan 49112, Republic of Korea, jysong.ualab@gmail.com), Donghwan Jung, J. S. Kim (Korea Maritime and Ocean Univ., Busan, Republic of Korea), and Jaehyuk Lee (Daewoo Shipbuilding and Marine Eng., Geoje, Republic of Korea)

The parametric array that generates narrow beams at low frequencies with apertures shorter than the wavelength and has been widely investigated when measuring the acoustic characteristics of materials, such as sound transmission and reflection properties. Because the parametric array utilizes nonlinear interaction across the frequency components, nonlinear phenomena arise in both the acoustic medium and the hydrophone. The unwanted nonlinear sound generated in the hydrophone and instrumentation is termed pseudo-sound. This work devised an experiment to measure the pseudo-sound for the purpose of isolating the "real" nonlinear difference frequency sound from the total nonlinear sound. Experimental results were validated by comparing its results with the pseudo-sound measured with a truncator (i.e., acoustic filter) to identify hydrophone nonlinearity.

2:30

2pBAa7. Experimental investigation of parametric array for low frequency measurement system. Donghwan Jung (Korea Maritime & Ocean Univ., Busan, Republic of Korea, wjdehdghkss@gmail.com), Jiyoung Song, J. S. Kim (Underwater Acoust. Lab., Korea Maritime and Ocean Univ., Busan, Republic of Korea), and Jaehyuk Lee (Daewoo Shipbuilding & Marine Eng. CO., LTD, Geoje, Republic of Korea)

To evaluate the performance of an equipment or object in underwater acoustics, the acoustic properties of acoustic materials should be accurately predicted and measured. The measurement of acoustic properties via panel testing in water tank require a narrow beam due to unwanted signals such as reflection from walls and diffraction signal. However, in the case of low frequency sound source, it generally has broad beam, and in order to have narrow beam, it is difficult to operation and physical limitation. These limitations and problems can be overcome by using a small size of parametric array capable of generating high-directional low-frequency signals. In order to use the parametric array as a sound source for an acoustic measurement system, it is necessary to understand the characteristics of the parametric array in near field. So this study was performed an experimental investigation on parametric arrays and study to construct an optimal system. Numerous factors such as the effect of pseudo sounds and the optimal distance between the source and receiver were considered to construct the system, and the results of experimental investigations, such as an efficient method to remove the effects of pseudo sounds and effective length, were confirmed.

Session 2pBAb

Biomedical Acoustics, Engineering Acoustics and Physical Acoustics: Cavitation Nuclei: Bubbles, Droplets, and More III

Stuart Ibsen, Cochair
Biomedical Engineering, Oregon Health and Science University, 3181 S.W. Sam Jackson Park Rd., KCRB 5001.41, Portland, OR 97239

Michaelann Tartis, Cochair
Chemical Engineering, New Mexico Institute of Mining and Technology, 801 Leroy, Socorro, NM 87801

Contributed Papers

2:55

2pBAb1. Recondensation of nano-sized perfluorocarbon droplets after acoustic droplet vaporization for blinking ultrasound localization microscopy. Mark Burgess (Riverside Res., 156 William St., 9th Fl., New York, NY 10038, mark.t.b42@gmail.com), Mitra Aliabouzar, Christian Aguilar, Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI), and Jeffrey Ketterling (Riverside Res., New York, NY)

Ultrasound localization microscopy (ULM) is a promising super-resolution imaging technique based on the build-up of subwavelength localizations of gas-filled ultrasound contrast agents (UCAs) within the microvascular space. Slow flow regions can reduce localization rates and diminish the isolation of microbubble signals from tissue. This work compares the blinking contrast dynamics of nano-sized perfluorocarbon droplets to the slowly changing contrast dynamics of traditional gas-filled UCAs. A 16-MHz linear array connected to a Verasonics research ultrasound system was used to capture high-frame rate acquisitions of contrast agents flowing through an *in vitro-* flow phantom. Nano-sized droplets produced an on and off contrast signal due to vaporization/recondensation events which enhanced microbubble separation from tissue. This allows for higher localization rates for droplets at slow flow speeds, since a new population of bubbles was localized with each ultrasound pulse and was independent of flow speed. This could improve ULM at slow flow speeds, where UCAs require long acquisition times for adequate microbubble movement via flow to produce sufficient localization rates. Preliminary *in-vivo* experiments comparing UCAs to nano-sized droplets for contrast-enhanced imaging of a mouse kidney will also be discussed. [This work was supported by NIH grant R21 HD097485.]

3:10

2pBAb2. Impact of phase-change contrast agent size during activation using ultrasound. Dominique James (Bioengineering, Univ. of Texas at Dallas, 800 W Campbell Rd. BSB, Richardson, TX 75080, dominique.james@utdallas.edu), Darrah Merillat, Shashank Sirsi, and Kenneth Hoyt (Bioengineering, Univ. of Texas at Dallas, Richardson, TX)

Phase-change contrast agents (PCCAs) are liquid nanodroplets (ND) that transition into gas microbubbles (MB) when exposed to pulsed ultrasound (US) . The purpose of this study was to investigate the activation threshold of size-isolated PCCAs under physiologically relevant hydrostatic pressures. Size-isolated PFB NDs were prepared using an extrusion method and filter sizes of 100 or 400 nm. Using a programmable US scanner and linear array transducer, a custom scan sequence was implemented that interleaved pulsed US transmissions for both PCCA activation and MB detection. An automated US pressure sweep was performed (3 to 6 MPa, $N = 200$ discrete intervals), and grayscale US images were acquired at each increment. PCCAs were circulated through a flow phantom at 37 deg. Hydrostatic pressures applied to the PCCAs was controlled by constriction of flow phantom tubing. Reference measures were recorded by a calibrated pressure catheter. Activation thresholds were quantified using custom MATLAB software. The US-detected PCCA activation threshold increased with increased hydrostatic pressure in the range of 0 to 75 mmHg. The 100 nm size-isolated PCCAs activated at a higher US pressure as compared to 400 nm agents (4.3 ± 0.2 mmHg versus 3.7 ± 0.2 mmHg, $p < 0.001$). A positive correlation was found between the PCCA activation threshold and applied hydrostatic pressure for both 100 and 400 nm PCCAs ($R^2 > 0.95$, $p < 0.001$). This strong linear relationship could be exploited for noninvasive pressure estimation using US and PCCAs.

3:25

2pBAb3. Sonopermeation to deliver topotecan using lipid-prodrugs, liposomes, and microbubbles. Mendi G. Marquez (Chemical Eng., New Mexico Inst. of Mining and Technol., 801 Leroy Pl., Socorro, NM 87801, mendi.marquez@student.nmt.edu), Paula Gomez (Surgery Dept., The Univ. of Chicago Medicine and Biological Sci., Chicago, IL), Mohammadaref Ghaderi (Bioengineering, UT Dallas, Dallas, TX), Shashank Sirsi (Bioengineering, UT Dallas, Richardson, TX), Sonia Hernandez (Surgery Dept., The Univ. of Chicago Medicine and Biological Sci., Chicago, IL), Liliya Frolova (Chemistry, Purdue Univ., Fort Wayne, IN), and Michaelann Tartis (Chemical Eng., New Mexico Inst. of Mining and Technol., Socorro, NM)

Sonopermeation using microbubbles is an emerging extracorporeal drug delivery technique for impenetrable tumors. Utilizing the sonopermeation effect while enhancing the bioavailability of clinical chemotherapeutics using nanoparticle strategies is a potentially powerful combination. Clinically available topotecan, a chemotherapeutic, has limited bioavailability and therapeutic efficacy, which is attributed to poor solubility, rapid clearance, and inactivation under physiologic conditions. To overcome this, topotecan has been conjugated with phospholipids converting it into a lipid-prodrug (2T-T) for use in lipid-based carriers, such as liposomes, for sonopermeation. In this presentation, we present the characterization results of topotecan lipid-prodrug liposomes, preliminary *in vitro* efficacy in several neuroblastoma cell lines, and *in vivo* proof-of-concept in neuroblastoma xenograft tumors. Lipid-prodrug characterization demonstrates stable, high-drug loading liposomes are attainable while maintaining cytotoxicity. Liposome formulations, analyzed using ultraviolet-visible spectrometry and dynamic light scattering, indicate peak incorporation limits are attained for 100 nm extruded 60 mol. % 2T-T liposomes with no change in size over a 2 month period. *In vivo* xenograft neuroblastoma studies co-injected with lipid-prodrug liposomes (40 mol% 2T-T, 10 mg/kg) and microbubbles, for sonopermeation, were performed to determine the extent of 2T-T delivery with sonoporation over liposomal delivery alone at 24 hours post-treatment.

3:40

2pBAb4. Bubble nuclei in polyacrylamide hydrogels. Ferdousi Sabera Rawnaque (Penn State Univ., 210 East Hamilton Ave., Apt. 31, State College, PA 16801, fmr5186@psu.edu) and Julianna Simon (Penn State Univ., University Park, PA)

Bubble nuclei have been studied extensively in water for over 50 years with nuclei categorized as homogeneous or heterogeneous. However, it is unclear how those nuclei identified in water translate to viscoelastic hydrogels or tissues. In this study, bubble nuclei were evaluated in 17.5%, 20%, and 22.5% v/v% polyacrylamide (PA) hydrogels ($n = 3$ each). A 1.5 MHz focused ultrasound transducer was used to induce cavitation using 10-ms pulses with pressures ranging up to $p_+ = 8 = 89$ MPa, and $p_- = 2 = 26$ MPa and -6 dB focal dimensions of 9.4×1.15 mm^2 (p_-). Cavitation size and location were monitored with high-speed photography. When the concentration of PA increased from 17.5% to 20%, the area occupied by bubbles at 0.07 ms decreased from 0.14 mm^2 to 0.06 mm^2. The location of acoustic cavitation for replicate exposures in the 17.5% gel at 0.07 ms became more consistent as the acoustic pressure increased with no bubble overlap at $p_- = 2 = 21$MPa and 18% bubble overlap at $p_- = 2 = 26$MPa. These results suggest acoustic cavitation in PA hydrogels is dependent on the availability of bubble nuclei at each driving pressure. Future work includes investigating the distribution of bubble nuclei in tissues. [Work supported by NSF CAREER 1943937 and PSU Riess Fellowship].

3:55

2pBAb5. C-mode passive cavitation images for predicting large volume FUS-mediated drug delivery outcome. Yan Gong (Biomedical Eng., Washington Univ. in St. Louis, 4511 Forest Park Ave., St. Louis, MO 63108, yan.gong@wustl.edu), Chih-Yen Chien, Dezhuang Ye, and Hong Chen (Biomedical Eng., Washington Univ. in St. Louis, St. Louis, MO)

Passive cavitation detection has been commonly used to monitor focused ultrasound (FUS)-mediated blood-brain barrier disruption (FUS-BBBD). Previous research by our team has shown the feasibility to correlate passive cavitation imaging (PCI) with drug delivery outcome via FUS-BBBD at a single target. This study proposed to perform c-mode PCI for predicting the spatial distribution of aPD-L1 delivered by FUSBBBD. A single-element FUS transducer was used to perform sonication iteratively in a 3×3 grid at the whole brainstem of wild-type mice. PCI were acquired using an ultrasound imaging probe co-aligned with the FUS transducer. Reconstruction was performed to obtain c-mode PCI for the whole sonicated area in the plane normal to the PCI imaging plane (axial plane of FUS transducer). Fluorescence-labeled aPD-L1 was intravenously injected after FUS sonication and the delivery outcome was quantified using *ex vivo* fluorescence imaging. A high correlation ($R^2 = 0.79$) was obtained between the fluorescence intensity and the cavitation dosage calculated from the c-mode PCI. No significant difference was found between 3dB area of the fluroescence images (6.93 ± 2.24mm^2) and the 3 dB area of the c-mode PCI images (7.20 ± 0.74mm^2). This study demonstrated that the c-mode PCI had the potential to predict drug delivery outcome by large volume FUS-BBBD.

4:10

2pBAb6. Proton dosimetry using superheated droplets: Quantification with ultrasound. Gonzalo Collado-Lara (Dept. of Cardiology, Erasmus MC, Wytemaweg 80, Rotterdam, Zuid-Holland 3015CN, The Netherlands, g.colladolara@erasmusmc.nl), Sophie V. Heymans (KU Leuven, Kortrijk, Belgium), Marta Rovituso (Holland PTC, Delft, The Netherlands), Hendrik Vos (Dept. of Cardiology, Erasmus MC, Rotterdam, The Netherlands), Jan D'Hooge (KU Leuven, Leuven, Belgium), Nico de Jong (Dept. of Cardiology, Erasmus MC, Rotterdam, The Netherlands), Koen Van Den Abeele (KU Leuven, Kortrijk, Belgium), and Verya Daeichin (TU Delft, Delft, The Netherlands)

Superheated nanodroplets were proposed for dosimetry in proton therapy. Proton energy deposition within the droplet core leads to droplet vaporization. Vaporizations can be localized through acoustic pulse-echo, and used to measure the proton stopping distribution. Since protons can only lead to vaporizations as they stop, the probability of vaporization is estimated as the probability of a proton stopping within a droplet. Assuming that the number of vaporizations follows a binomial distribution, conversion from vaporizations to number of protons requires knowledge of: droplet size distribution and concentration, and acoustic FoV volume. Here we address the latter. We propose an experimental and numerical method to characterize the 2-way acoustic elevational width of a linear array. First, the impulse response and 1-way field of the acoustic array was measured using a hydrophone. Then, the 1-way field was fitted to field-ii simulations using a least-squares algorithm to estimate the optimal parameters. Finally, the 2-way acoustic field was simulated. The effective elevational width depends on the scattering cross section of the resulting microbubbles, and was determined from the experimental range of backscattering intensities. The method allowed to predict the vaporization counts of experimental measurements where three different size-sorted droplet populations were used.

Session 2pBAc

Biomedical Acoustics and Signal Processing in Acoustics: Ultrasound for Brain Stimulation

Yun Jing, Cochair
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Qifa Zhou, Cochair

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

Chair's Introduction—4:35

Invited Paper

4:40

2pBAc1. Noninvasive and cell-type specific neuromodulation by integrating ultrasound and genetics. Yaoheng Yang, Christopher P. Pacia, Dezhuang Ye, Lifei Zhu, Yimei Yue, Jinyun Yuan, Mark Miller, Jianmin Cui, Culver Joseph (Washington Univ. in St. Louis, St. Louis, MO), Michael Bruchas (Univ. of Washington, Seattle, WA), and Hong Chen (Washington Univ. in St. Louis, 6338 Washington Ave., University City, MO 63130, chenhongxjtu@gmail.com)

The investigation of brain functions and treatment of brain disorders requires modulation of selected neurons in a noninvasive manner in the deep brain. To achieve this goal, sonogenetics has been developed that involves the use of focused ultrasound (FUS) to selectively control a specific type of neurons that have been genetically modified to express ultrasound-sensitive ion channels. Existing sonogenetic techniques utilize ultrasound mechanical effect to activate mechanosensitive ion channels. Different from existing approaches, we aimed to develop sonothermogenetics for noninvasive, deep-penetrating, and cell-type-specific neuromodulation by combining a thermosensitive ion channel TRPV1 with FUS-induced brief, non-noxious thermal effect. FUS sonication at the mouse brain *in vivo* selectively activated neurons that were genetically modified to express TRPV1. Temporally precise activation of TRPV1-expressing neurons was achieved with its success rate linearly correlated with the peak average temperature within the FUS-targeted brain region as measured by *in vivo* magnetic resonance thermometry. FUS stimulation of TRPV1-expressing neurons of the Parkinsonian circuit at the striatum repeatedly evoked rotating locomotor behavior in freely moving mice. FUS sonication was confirmed to be safe based on inspection of neuronal integrity, inflammation, and apoptosis markers. In summary, our study demonstrated that sonothermogenetics is a noninvasive and cell-type-specific neuromodulation technique.

Contributed Papers

5:00

2pBAc2. Affordable stereotactic-guided focused ultrasound device for tunable drug delivery to the mouse brain. Zhongtao Hu (Washington Univ. in St. Louis, Radiation Oncology, St. Louis, MO 63108, zhongtaohu@wustl.edu), Si Chen, Yaoheng Yang, Yan Gong, and Hong Chen (Washington Univ. in St. Louis, St. Louis, MO)

Focused ultrasound combined with microbubble (FUS+MB)-mediated blood brain barrier (BBB) opening is not only a promising technique for clinical applications but also a powerful tool for preclinical neuroscience and neuro-oncology research. However, existing FUS systems are expensive and lack the flexibility of modulating BBB opening volume, which prevents a broader research community from adopting the FUS+MB technique in preclinical studies. To address the challenge, we developed a low cost (~\$100), mini FUS transducer with the capability to modulate BBB opening size that can be readily integrated with a standard stereotaxic frame for mouse. Specifically, we manufactured in-house mini FUS transducers with three frequencies (1.5, 3.0, and 6.0 MHz), and quantified BBB opening and targeting accuracy at varying pressures (0.20 to 0.81 MPa). The volume of FUS+MB-induced BBB opening was evaluated using both contrast-enhanced MRI and Evans blue extravasation. Our results showed that we can achieve varying amount of Evans blue delivery and BBB opening size by modulating the transducer frequency and acoustic pressure. Additionally, the amount of Evans blue delivery and BBB opening had a significant linear correlation with the cavitation index (defined by the ratio between acoustic pressure and frequency). We believe that this stereotaxic-guided FUS system will lower the barrier for adopting the FUS technique in a broader research community.

5:15

2pBAc3. Focused ultrasound immunomodulation on mononuclear phagocytes in the brain. Tao Sun (Radiology, Focused Ultrasound Lab., Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., EBRC 514, Boston, MA 02115, taosun@bwh.harvard.edu) and Nathan J. McDannold (Radiology, Brigham and Women's Hospital, Harvard Med. School, Boston, MA)

Focused ultrasound (FUS)-mediated blood–brain barrier opening is an attractive method of treating neurological diseases, such as glioblastoma (GBM) and Alzheimer's disease (AD). As an established drug delivery approach, FUS has also been associated with the stimulation of neuroinflammation, including the morphological activation of resident microglia. Here we report FUS immunomodulation effects may also include the enhanced recruitment and activation of other mononuclear phagocytes in murine models of GBM (GL261) and AD (APP/PS1dE9) in tests with or without monoclonal antibody-based passive immunotherapy. Immunoprofiling via flow cytometry and immunofluorescent staining were performed in the harvested brains. In the GBM model, our results demonstrated that FUS enhanced antigen presentation behaviors of tumor-associated macrophages without affecting the microglia. FUS also reprogrammed the macrophages locally towards the anti-cancer phenotype. In the AD model, plaque-associated Ly6G+ phagocytes were only present in FUS-treated areas, while microglia activation was found in both antibody-treated groups. Taken together, our results offer new evidence in FUS immunomodulation on the myeloid compartment of the brains in GBM and AD-like mouse models.

TUESDAY AFTERNOON, 30 NOVEMBER 2021 306 (L)/307 (O), 1:20 P.M. TO 4:30 P.M.

Session 2pCA

Computational Acoustics, Physical Acoustics, Noise, and Musical Acoustics: Computational Aeroacoustics

S. Hales Swift, Cochair
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Z. Charlie Zeng, Cochair
Mechanical and Aerospace Engineer, Univ. of Kansas, Lawrence, KS

Chair's Introduction—1:20

Invited Papers

1:25

2pCA1. A modified acoustic-turbulent scattering model for long-range propagation with wind tunnel validation. Tianshu Zhang (Mech. and Aerosp. Eng., Univ. of Florida, 939 Ctr. Dr., Gainesville, FL 32611, zhang.tianshu@ufl.edu) and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

We present a newly developed 'bridging' model to quantify turbulent effects on long-range acoustic propagation in the atmosphere. The bridging model is incorporated into an existing validated Burgers' equation solver. The new model combines the scattering model of Ostashev and Wilson and a traditional refraction model for acoustic-turbulent interaction. A new attenuation coefficient is obtained that is consistent with the atmospheric attenuation coefficient via integration of energy loss over all scattering angles. To validate and quantify the effect of the new model, we conduct a series of propagation experiments in the National Science Foundation funded University of Florida Boundary Layer Wind Tunnel. A synthetic noise signal propagates through the tunnel, which has controllable boundary layer turbulent statistics via fan speeds and hundreds of individually controlled roughness elements. We compare predictions of the bridging model, Ostashev and Wilson's model, and the experiments of the wind tunnel. Also, we perform example long-range predictions using the new model and Ostashev and Wilson's model with a synthetic tornadic infrasound signal. A sensitivity test of model parameters is performed. We show that for particular atmospheric conditions and turbulent statistics, the bridging model has particular advantages with minimal additional computational expense.

1:45

2pCA2. High-fidelity fan-noise simulations based on improved delayed detached eddy simulation. Takao Suzuki (The Boeing Co., 9705 24th Pl. W, Everett, WA 98204, takao.suzuki@boeing.com), Michael L. Shur, Michael K. Strelets, Andrey K. Travin (St. Petersburg Polytechnic Univ., St. Petersburg, Russian Federation), and Philippe R. Spalart (The Boeing Co., Woodinville, WA)

We have developed high-fidelity CFD/CAA capability for the prediction of fan tone and broadband noise from an aero-engine. With a zonal URANS—wall-modeled LES approach, referred to as the Improved Delayed Detached Eddy Simulation (IDDES), the entire geometry of NASA's Source Diagnostic Test (SDT) fan rig including the nacelle is solved at approach, cut-back and take-off conditions. Multi-block grids with high-order structured finite-volume schemes are designed to resolve turbulence in the wakes and vortices as well as sound propagation in the bypass duct through the blade rows. By applying the Ffowcs-Williams and Hawkings technique with permeable surfaces, far-field sound-power spectra are computed, and by incorporating with the duct-mode-extraction technique, the radiated sound from the inlet and exhaust is decomposed into duct modes. These predicted fan-noise characteristics as well as aerodynamic performances are validated against NASA's wind-tunnel test data. The fan-noise associated with the rotor-stator interaction is very well characterized: The tone power levels up to the second blade passing frequencies from both inlet and exhaust are predicted well across the engine speed, and the broadband power spectra are also predicted except for those from the inlet only at supersonic tip Mach numbers.

2:05

2pCA3. Acoustics of rotor ingesting an axisymmetric turbulent boundary layer. Meng Wang (Aerosp. and Mech. Eng., Univ. of Notre Dame, 105 Hessert Lab., Notre Dame, IN 46556, m.wang@nd.edu), Di Zhou, and Kan Wang (Aerosp. and Mech. Eng., Univ. of Notre Dame, Notre Dame, IN)

The noise generation by a five-bladed rotor ingesting an axisymmetric turbulent boundary layer at the tail end of a body of revolution (BOR) is investigated numerically at Reynolds number of 1.9×10^6 based on the BOR length and free-stream Mach number of 0.059. The turbulent boundary layer on the nose and midsection of the BOR is computed using wall-modeled LES whereas that in the acoustically important tail-cone section under a strong adverse pressure gradient is wall resolved. The radiated acoustic field is calculated based on the Ffowcs Williams-Hawkings equation. The computed flow statistics and sound pressure spectra agree well with the experimental measurements at Virginia Tech. In addition to broadband turbulence-ingestion noise, haystacking spectral peaks near the blade passing frequency and its harmonics are well captured and related to correlated blade interactions with coherent turbulence structures. The accuracy of the classical Sears theory for rotor-noise prediction is evaluated using the upwash velocity field from the LES. The theory is shown to predict well the sound pressure levels at low to mid-frequencies including the haystacking peaks but less accurately at higher frequencies. The spatial and frequency characteristics of blade unsteady-loading dipoles and their correlations are investigated, and the appropriate Mach number scaling for the ingestion noise is identified.

2:25

2pCA4. Acoustical holography-based sound power decomposition of a highly heated, large eddy-simulated supersonic jet. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu), Kevin M. Leete (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Junhui Liu (Naval Res. Lab., Washington, DC), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

In recent years, large-eddy simulation (LES) has proved a powerful tool for modeling increasingly heated, supersonic jets and analyzing their source characteristics. This paper describes an investigation of the acoustic source characteristics from an LES of an underexpanded jet with total temperature ratio of seven. Using frequency-domain pressures and particle velocities at, and generated from, the Ffowcs Williams Hawkings surface (FWHS), holography and intensity techniques are used to perform an axial decomposition of radiated sound power from the jet, whereas a far-field arc produces a narrow, nonphysical source distribution, the intensity from the near-field array and FWHS surface yield sound power decompositions that approximate the nozzle lipline squared-pressure's spatial distribution. For this LES, the peak sound power is radiated just upstream of the potential core tip, and over 90% of power is radiated upstream of the supersonic core tip.

2:45–3:00 Break

Contributed Papers

3:00

2pCA5. Spectral proper orthogonal decomposition analysis of flowfield for supersonic nozzle jet noise reduction. Zhongquan C. Zheng (Mech. and Aerosp. Eng., Univ. of Kansas, 4130 Old Main Hill, Logan, UT 84322, zzheng@usu.edu), Meihua Zhang (Mech. and Aerosp. Eng., Utah State Univ., Logan, UT), and Zhijian Wang (Aerosp. Eng., Univ. of Kansas, Lawrence, KS)

Spectral proper orthogonal decompositions (SPOD) is the frequency domain form of proper orthogonal decomposition (POD). SPOD is shown to be directly applicable to jet noise studies. A computational SPOD analysis is performed on the Large-Eddy Simulation (LES) computational data for both the baseline nozzle flow and a modified nozzle flow. The mode energy analysis from SPOD shows that for the baseline nozzle, the low-rank behavior is found at frequencies at which a large separation occurs. The low-rank behavior disappears for the modified nozzle flow where no significant peaks are detected. The coherent structures of the low frequency show the Kelvin–Helmholtz type instability. It is also shown that the reconstructed flow field using the first decades of SPOD modes can capture the major shock-expansion based interactions that are present within the original flow field. This suggests that the method of SPOD-based reconstruction can be used to carry out additional modal and acoustic analyses of jet flows.

3:15

2pCA6. On the use of numerical Green's functions in acoustic analogies for edge noise prediction at low Mach number. Nicolas Trafny (Naval Group, 199 Ave. Pierre-Gilles de Gennes, Ollioules 83190, France, nicolas@trafny.net), Benjamin Cotté (IMSIA, ENSTA Paris, CNRS, CEA, EDF, Institut Polytechnique de Paris, Palaiseau, France), Gilles Serre (Naval Group, Ollioules, France), and Jean-François Mercier (POEMS, ENSTA Paris, CNRS, INRIA, Institut Polytechnique de Paris, Palaiseau, France)

The interaction between a turbulent flow and a rigid obstacle leads to a far-field broadband acoustic radiation which can have a significant impact in many industrial applications. Depending mainly on the Mach number, different prediction methods have been developed. At low Mach number, direct noise computation methods are too expensive. Other approaches must be chosen, based on acoustic analogies which consist in separating noise generation and noise propagation mechanisms. In this study, we focus on the Lighthill's wave equation solved using both analytical and numerical tailored Green's functions. The latter is computed using an accelerated Boundary Element Method. Also, a statistical model for the velocity fluctuations in the turbulent flow is used to estimate the far-field radiated noise spectrum. It can be built from either a simulation of the mean flow or an estimation of the boundary layer parameters. To validate numerical predictions for the far-field acoustic pressure spectrum, a NACA 0012 airfoil is considered. Both leading edge and trailing edge predictions are validated against experimental data. Moreover, the impact of both the inflow velocity and the angle of attack on spectra and directivity diagrams is investigated.

3:30

2pCA7. A comparative study of passive compliant coatings in mitigating trailing-edge noise. Rohith Giridhar (Aerosp. Eng., The Univ. of Kansas, 2132 Learned Hall, 1530 W 15th St., Lawrence, KS 66045, rohith.giridhar91@ku.edu), Saeed Farokhi, and Ray Taghavi (Aerosp. Eng., The Univ. of Kansas, Lawrence, KS)

With growing dependence on wind energy as a clean source of electricity, it is imperative to address the shortcomings in modern day wind turbines and ensure their widespread application. One such area to address is the trailing-edge noise from their rotor blades. The present computational study investigates the ability of passive compliant coatings to favorably modulate flow boundary layer and mitigate trailing-edge noise. A flat-plate in fully turbulent flow is chosen to demonstrate this technique. Computational Aeroacoustics Analysis is performed on a flat-plate to predict trailing-edge noise using Ffowcs Williams and Hawkings acoustic analogy for a chord-based Reynolds number of $Re_c = 460{,}000$. Predicted noise accurately follows the magnitude and trend of measurements for the frequency range of 750—7000 Hz (i.e., Strouhal number $St = 5.14$—40.00). Accurate prediction of farfield noise demonstrates validity of this approach. Experiments have shown that compliant coatings reduce the amplitude of Tollmien-Schlichting (TS) waves, and enable drag reduction. Two such coatings with different stiffness characteristics are applied on the flat-plate to investigate their ability to offer constructive coupling with flow boundary layer and mitigate trailing-edge noise. *Fluid Structure Interaction* is employed to simulate coating deformation.

3:45

2pCA8. Sonic boom propagation in a non-homogeneous atmosphere using a stratified ray tracing technique. Kimberly A. Riegel (Phys., Queensborough Community College, 652 Timpson St., Pelham, NY 10803, kriegel@qcc.cuny.edu), William Costa (None, Albany, NY), George Seaton (New York Inst. of Technol., Westbury, NY), and Christian Gomez (SUNY Binghampton, Binghampton, NY)

The propagation of sonic booms in and around large urban areas needs to be well understood in order to determine the impact of the sound will have on the population. Previously a combined ray tracing/radiocity method was developed to model sonic booms around large buildings. While the overall propagation was acceptable, the model had some limitations. One of these limitations was the assumption of a constant speed of sound over the entire propagation area. This updated model uses an atmosphere stratified in the z direction to include the effects of changing temperature in the vertical direction. Another limitation of the previous model is high computation times. As the environment becomes larger and more complicated the computation time becomes very large. To improve computation time and allow for larger, more complex environments, the stratification of the atmosphere was exploited and applied to the environment as well. Only the sections of the building that are within the striation are checked for collisions. This significantly reduces the required computations and simplifies each step of the ray tracing even for complex building configurations. The computation times and pressure using the updated model are presented for several urban environments. The previous model results will be shown as a comparison.

4:00

2pCA9. Atmospheric sound transmission loss uncertainties induced by sea roughness. Andrea Vecchiotti (Mech. Eng., The Catholic Univ. of America, Viale dei Gerani 11, Senigallia (AN) 60019, Italy, a-vecchiotti@libero.it), Teresa J. Ryan, Faith A. Cobb (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 conducted on atmospheric sound propagation over sea. In particular, it focuses on the sound pressure level prediction uncertainties induced by the water surface roughness. To quantify these uncertainties, the generalized terrain parabolic equation (GTPE) is used to model sound propagation above water surfaces at different sea states. Water roughness is pseudo-randomly generated using an ocean wave spectrum. The GTPE predictions are compared with those obtained using the Crank-Nicholson parabolic equation (CNPE) solver. When using the CNPE the sea surface is flat and has a surface impedance equivalent to that of a rough surface. The use of the GTPE is less computationally efficient but provides insight on the detectability of an acoustic source at sea. This work presents relationships between fully developed sea states (up to sea state 4) and the uncertainties on sound pressure level predictions at distances up to 500 m from the source. These relationships are presented for typical diurnal and nocturnal thermal gradients and for different elevations from the water surface.

4:15

2pCA10. A mixed Fourier transform for impedance boundary conditions of acoustic waves at grazing incidence. Alexander N. Carr (Mech. and Aerosp. Eng., Univ. of Florida, PO Box 116250, 939 Sweetwater Dr., Gainesville, FL 32611, alexcarr.1721@gmail.com), Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, Hampton, VA), and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

Acoustic propagation in the atmosphere near the Earth's surface at grazing incidence is influenced by ground impedance. Consequently, numerical simulations must use an impedance ground boundary condition in order to accurately predict ground effects associated with grazing incidence. A mixed Fourier transform method for impedance boundary conditions has been developed for parabolic approaches by the electromagnetics community. Here, the mixed Fourier transform method is adapted and validated for acoustic wave propagation. The one-way solution of the Helmholtz equation is computed with a modified angular spectrum approach to account for a locally reacting surface using a mixed Fourier transform in the vertical direction. Numerical simulations are presented that focus on sonic boom propagation near the ground in the shadow zone. The simulations demonstrate the utility of the mixed Fourier transform method to account for ground impedance effects that are ignored by ray theory-based propagation codes. Consequently, the method extends sonic boom prediction to the lateral cutoff and shadow zone regions. [This research is supported by the Commercial Supersonic Technology Project of the National Aeronautics and Space Administration under Grant No. 80NSSC19K1685.]

Session 2pID

Interdisciplinary and Student Council: Introduction to Technical Committees

Mallory Morgan, Cochair
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Kieren McCord, Cochair
Arizona State University, 660 S. College Avenue, Tempe, AZ 85281

Chair's Introduction—3:00

Invited Papers

3:05

2pID1. An Introduction to the Technical Committee on Animal Bioacoustics. Laura Kloepper (Saint Mary's College, Notre Dame, IN) and Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

Animal Bioacoustics as a field of research involves the study of sound in non-human animals. The range of this field is wide and includes all aspects of sound production and reception, communication and associated behaviors, acoustic ecology and effects of noise/sound, and passive and active acoustic methods for monitoring individuals, populations, habitats and ecosystems. Members of the Technical Committee on Animal Bioacoustics (TCAB) come from diverse backgrounds, including those with training in biology, ecology, engineering, mathematics, oceanography, physics, and psychology. Many TCAB members also participate in the activities of other ASA Technical Committees including Acoustical Oceanography, Engineering Acoustics, Noise, Psychological and Physiological Acoustics, Signal Processing, and Underwater Acoustics, reflecting the interdisciplinary nature of the field. This talk will highlight some of the popular and emerging areas of research in Animal Bioacoustics.

3:15

2pID2. Signal Processing: Data analysis, machine learning, and communications. Geoffrey F. Edelmann (U. S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil)

Whether it is communicating to an autonomous underwater vehicle, finding a defect in an airplane's wing, focusing sound to destroy kidney stones, or learning how bats track their food, advanced signal processing is key to successful research. The Technical Committee on Signal Processing in Acoustics was formed to foster interdisciplinary interaction among the Technical Committees. This broad discipline covers such unique topics as detection, tracking, classification, communications, machine learning, environmental analysis, focalization, robotics, and their application to ships, animals, speech, earthquakes, medicine, environmental knowledge, and unmanned vehicles, as well as future problems yet undiscovered. This presentation will highlight some of the exciting techniques being used in signal processing and in the process give an appreciation for the breadth of the applications of the techniques being developed.

3:25

2pID3. Speech Communication: Talkers, listeners and signals. Rajka Smiljanic (Univ. of Texas at Austin, 305 E. 23rd St., B5100, Austin, TX 78712, rajka@austin.utexas.edu)

Speech communication research examines how spoken language is produced, transmitted and perceived. This involves a number of different disciplines, from linguistics and experimental psychology to speech and hearing sciences, and electrical engineering. The field covers a wide range of physiological, psychological, acoustic, and linguistic phenomena. In this presentation, I will highlight some ongoing research in the Speech Communication Technical Committee including work that examines how different challenges to the clarity of the speech signal, such as background noise, interlocutors who are not native speakers of the target language, or hearing loss, affect speech communication. This research has implications for improving communicative effectiveness for various talker and listener groups and human-machine interactions.

3:35

2pID4. An introduction to Psychological and Physiological acoustics. Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu)

The technical committee on Psychological and Physiological acoustics (P&P) encompasses a wide and multidisciplinary range of topics. It is concerned with questions of what happens to sound once it enters the auditory system, and how sound is processed to facilitate communication and navigation. Topics include the biomechanics of the middle and inner ear; the neuroscience of the auditory nerve,

brainstem, and cortex; and behavioral studies of auditory perception and cognition. Within these topics, research covers both normal function and effects of hearing disorders, as well as applied questions related to hearing aids, cochlear implants, and audio technology. The scope of P&P overlaps heavily with that of several other technical committees in ASA, especially Architectural Acoustics, Animal Bioacoustics, Noise, and Speech Communication. This presentation will provide an overview and some examples of currently active areas of investigation.

3:45

2pID5. Introduction to the Musical Acoustics technical area. Andrew A. Piacsek (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

The field of musical acoustics encompasses all aspects of the design, construction, and performance of musical instruments (including the voice), as well as the physical and cultural environment in which they are played, with the goal of understanding how these elements contribute, separately or together, to the acoustic characteristics. While the general acoustic behavior of most musical instruments can be explained in terms of fundamental principles governing waves in strings, pipes, and plates, there is a high degree of complexity in the mechanical motion and the resulting sound that enables a rich variety of tonal character and musical expression. This presentation will provide an overview of current research areas that address the complexity and variety of musical instruments and the listener experience, including nonlinear effects, flow-structure interaction, computational modeling and machine learning, measurement techniques, performer-instrument interaction, and recording/playback technology. Along the way, the relevance of other technical areas of acoustics will be noted, including physical acoustics, structural acoustics, signal processing, computational acoustics, and psychological and physiological acoustics.

3:55

2pID6. Technical Committee on Noise: Researchers and practitioners. Alexandra Loubeau (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

The Technical Committee on Noise (TCNS) includes researchers and industry practitioners interested in understanding noise generation from a variety of sources, its propagation, and its impact on structures, objects, and people. The goal of reducing the impact of this noise, i.e., unwanted sound, has led to innovative model development, measurement techniques, mitigation strategies, and input towards the establishment of regulations. Community noise definition and abatement is of particular interest and motivates the advancement of research on a variety of topics, such as transportation noise, hearing protection, jet and rocket noise, building systems noise and vibration, atmospheric sound propagation, soundscapes, and low-frequency sound. The diverse technical background of TCNS members is essential, given the interdisciplinary nature of this research, and it is mirrored in the topics of special sessions typically held jointly with other ASA technical committees.

4:05

2pID7. Architectural Acoustics: How does that building sound? Ana M. Jaramillo (Ahnert Feistel Media Group, 8717 Humboldt Ave. N., Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

Architectural acoustics is the area that studies sound in and around buildings. Sometimes the requirements call for good acoustic quality for music performance and sometimes for clear speech intelligibility. Additionally, architectural acoustics also deals with issues of unwanted sound in buildings and speech privacy. TCAA very often crosses interests with TCN, and sometimes with other technical areas in the ASA. This presentation will cover typical and current topics, as well as details about the Committee membership and activities.

4:15

2pID8. Introducing the Structural Acoustics and Vibration Technical Committee. Christina J. Naify (Appl. Res. Labs, The Univ. of Texas, Austin, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify@gmail.com)

The Structural Acoustics and Vibrations Technical Committee (TCSA) includes scientific study of vibrating structures, excited using either elastic or acoustic waves, and radiated acoustic fields from those structures. The committee members study a diverse range of physics relating to these basic phenomena, from damping and isolation, to active control, to modal response and a variety of techniques are used, including numerical modeling, analytical techniques, and experimental measurements. Due to the wide range of applications in which vibrating structures are found, TCSA is multidisciplinary, with ASA meeting sessions often co-chaired with Physical Acoustics, Engineering Acoustics, Signal Processing to name a few. Additionally, research fields span a range of disciplines in their practice, including industry, academia and government. This talk will provide a brief overview into TCSA including historical highlights and future directions as well as summaries of some of the recent special sessions presented at Acoustical Society meetings.

4:25

2pID9. The Technical Committee on Physical Acoustics: Exploration across the acoustic spectrum. Joel Mobley (Phys. and Astronomy, Univ. of MS, PO Box 1848, 108 Lewis Hall, University, MS 38677, jmobley@olemiss.edu)

The Technical Committee on Physical Acoustics (TCPA) is focused on the physics of wave propagation and related phenomena across the entire frequency range of acoustics—from infrasound through ultrasound. Our TC includes a broad range of scientists and engineers involved all in aspects of the study of acoustical phenomena from probing the properties of matter to exploring the physical effects of sound. The TCPA looks to advance not only the science of acoustics but also the methods of investigation whether theoretical, computational or experimental. Due to our broad interests in the underlying physics of wave phenomena, we have a diverse membership who are also involved with a number of other TC's in the Society. This presentation will discuss all facets of TCPA, and its intersection with other ASA technical areas.

4:35

2pID10. Babies, bubbles, and brain disease: An introduction to the Biomedical Acoustics Technical Committee. Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu)

For most folks, ultrasound in medicine is associated with fetal imaging, and it remains among the most widely available and frequently used imaging modalities in the clinic. Recent advances have expanded the capacity of ultrasound imaging to include the ability to quantify mechanical properties of tissue (elastography), microvascular structure (super resolution imaging), perfusion (contrast imaging), and molecular expression such as oxygenation (photo-acoustic imaging). Effective diagnostic information is now possible for targets traditionally avoided by ultrasound imaging, such as lung for COVID-19 diagnoses. Ultrasound energy can also be harnessed for therapeutic purposes. Focused sources enable us to ablate malignancies through thermal or mechanical-based mechanisms. A strong tumor immune response has been observed for histotripsy, a bubble-based ablation method, potentiating a means to treat multi-focal disease. Ultrasound-mediated bubble activity enhances drug delivery to improve treatment for thrombo occlusive disease (e.g., heart attack, stroke, venous thrombosis), cancer, and tissue regeneration. Improved drug delivery is also possible in the brain via ultrasound-mediated opening of the blood brain barrier. Separate from drug delivery, ultrasound can induce neuromodulation for the treatment of essential tremor, neuropathic pain, and Parkinson's disease. Diagnostic and therapeutic ultrasound continues to advance the field of medicine, and are integral for improving the outcomes for debilitating diseases.

4:45

2pID11. Introduction to the Technical Committee on Engineering Acoustics. Michael R. Haberman (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

This talk will introduce ongoing work in the Technical Committee on Engineering Acoustics (TCEA) of the Acoustical Society of America, which is one of the most diverse Technical Committees of the Society. Engineering Acoustics encompasses the theory and practice of creating tools to generate and investigate acoustical phenomena and then to apply the knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, arrays, and transduction systems in all media and frequency ranges. It is also concerned with the design of acoustical instrumentation, metrology, and the calibration of those systems. It further considers all aspects of measurement, fabrication, and computational techniques as they relate to acoustical phenomena and their utility. The talk will provide an introduction of a broad range of research topics in TCEA, with specific highlights on exciting new areas of research.

4:55

2pID12. Introduction to the Computational Acoustics Technical Specialty Group. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

In acoustics, as in other fields, computation makes a seemingly ever-growing contribution to scientific and engineering progress. The Computational Acoustics Technical Specialty Group (CA TSG), which first met officially at the Fall 2018 ASA meeting, provides a forum for computational acoustics research that had often been discussed independently within many of the TCs. The CA TSG thus facilitates exchanges of recent advances and experiences with computational methods having broad applications across acoustics. Topics considered by the CA TSG include: (1) numerical methods for wave propagation, scattering, structural interactions; (2) optimization, parallelization, and acceleration of computational algorithms; (3) validation, benchmarking, and uncertainty analysis; (4) computational learning methods; (5) data analytics and visualization; and (6) practical utilization of acoustical computations for engineering and noise control.

5:05

2pID13. An introduction to the acoustical oceanography technical committee. Grant B. Deane (Scripps Inst. of Oceanogr., Code 0238, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu)

Underwater acoustics signals, both anthropogenic and natural, are rich sources of information about the ocean interior, the sea surface, and the seafloor. The acoustical oceanography technical committee (AOTC) seeks to foster a broad range of work in pure and applied acoustics aimed at gaining new and fundamental understanding of physical, biological, geophysical, and chemical processes in the ocean and at its boundaries. With this focus, the AOTC is necessarily strongly interdisciplinary, having close ties to other technical committees such as animal bioacoustics, physical acoustics, signal processing, and underwater acoustics. This talk will describe some of the approaches of our members and highlight recent advances in Acoustical Oceanography.

5:15

2pID14. An introduction to Technical Committee on Underwater Acoustics. Jie Yang (Appl. Phys. Lab, Univ. of Washington, 1015 NE 40th St., Seattle, WA 98105, jieyang@uw.edu) and D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, Monterey, CA)

The Acoustical Society of America Technical Committee on Underwater Acoustics investigates sound propagation in underwater environments such as oceans, lakes, rivers and tanks. Sound waves can travel thousands of kilometers in water and therefore, have been widely used as an effective tool to study the ocean for both economic and defense purposes. It is an interdisciplinary, observation-driven field of study that involves physics, oceanography, marine biology, engineering, and signal processing. In this talk, several representative research areas are presented, including measurement and modeling of sound propagation and reverberation, and ocean ambient sound field monitoring and characterization. A number of large, integrated field experiments are also presented, to illustrate the manner in which observations are designed to pursue scientific questions.

Session 2pNS

Noise, Architectural Acoustics, Structural Acoustics and Vibration, and Engineering Acoustics: Building Systems Noise and Vibration Control: Beyond ASHRAE Chapter 49 II

Brandon Cudequest, Cochair
Threshold Acoustics, 141 W Jackson Blvd Suite 2080, Chicago, IL 60604

Joseph Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, Ste. 209, Woodbridge, NJ 07095

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. Evaluating noise data from VRF system manufacturers. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Variable refrigerant flow (VRF) systems are becoming increasingly popular because of improved energy efficiency and the requirement for less HVAC ductwork in the building. This presentation will examine the current acoustic test methods and published sound data for both interior and exterior components of Variable Refrigerant Flow (VRF) systems. It will be shown that the current laboratory test standards vary significantly around the world and often do not provide adequate information for accurately predicting noise levels inside buildings.

1:25

2pNS2. Case study of variable refrigerant flow fan noise. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

This paper presents a case study of multiple VRF units measured during building commissioning and the field data is compared to the manufacturer data provided.

1:45

2pNS3. VRF noise: From data sheets to field measurements. Tyler Tracy (Acentech, Inc., 201 Sherman St., Apt. B, Cambridge, MA 02140, tyler_tracy@ymail.com)

In the last 20 years, Variable Refrigerant Flow (VRF) systems have gained prominence in many sectors of U.S. construction. These devices move conditioned refrigerant from outdoor units through pipes to indoor units, serving areas that require heating or cooling. Advantages of a VRF system include eliminating the need to carry conditioned air through long runs of ductwork, allowing more environmental control in each space or zone served, and offering quieter operation levels. Indoor VRF system noise levels provided by manufacturers are often measured in anechoic conditions close to the unit, which is in contrast to the installed conditions in many buildings using VRF systems. We seek to explore the existing literature documenting VRF sound levels, while we compare and contrast these sound levels to field-installed VRF system measurements.

2:05

2pNS4. You keep using that term, I do not think it means what you think it means: variable-air-volume terminal box catalog noise criteria data and what AHRI Standard 885 really says. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Logan Pippitt (DLR Group, New York, NY)

Mechanical Engineers often include Noise Criteria data on equipment schedules for variable-air-volume terminal boxes. This can lead to confusion when acousticians recommend noise control interventions, as the catalog data leads the Engineer to believe they have selected appropriately quiet equipment. This session will review how Noise Criteria catalog ratings are determined according to AHRI Standard 885 and compare the Standard assumptions versus real world designs. Multiple design cases will be presented to show the pitfalls of relying on equipment catalog Noise Criteria ratings.

2:25–2:40 Break

2:40

2pNS5. Challenges and conundrums. David A. Conant (McKay Conant Hoover, 5655 Lindero Canyon Rd., McKay Conant Hoover, Ste. 325, Westlake Village, CA 91362, dconant@mchinc.com) and K. Anthony Hoover (McKay Conant Hoover, Westlake Village, CA)

Various noise and vibration control experiences will be described, with data presented where available. Project types range from performing arts to offices, large to small, near and far. Some solutions were surprisingly straightforward, others were simply surprising, and others seemed fairly inventive. Some resisted predictive approaches or were hampered by questionable "data." Several noise issues turned out to be unrelated to the MEP system, and for one or two conundrums we will welcome your insights. Often, such experiences can often offer useful approaches where conventional wisdom fails, and these war stories are the most fun to share.

3:00

2pNS6. Mechanical noise control case studies. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, Ste. 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

A few case studies of mechanical system noise are presented. These address before/after background sound levels associated with room and/or mechanical system modifications, comparison of predicted mechanical system noise to measured values, our approach to noise control recommendations, and lessons learned. Interesting cases include a college theater renovation, office noise control involving implementation of a fan array and a VFD, and a challenging hunt for the source of an exterior pharmaceutical campus noise source generating complaints in a nearby neighborhood.

3:20

2pNS7. Noise control in laboratories. William Rosentel (TEECOM, 50 California St., San Francisco, CA 94111, william.rosentel@teecom.com) and Peter Holst (TEECOM, Oakland, CA)

ASHRAE Ch. 49 is a leading industry reference in noise control, providing extensive guidance on background noise levels, as well as practical noise control methods. The design guideline for background noise criteria in laboratories ranges from NC 35 to 50 depending on the expected activity. Noise control guidance for laboratories predominantly addresses fume hood design. In practice, recent workplace projects have included laboratory spaces without fume hoods; and background noise levels that are not necessarily controlled by the HVAC systems. The modern definition of "lab" space can include machining and fabrication (makerspaces), workplace server labs (server / chip design), and R&D testing suites for consumer electronics with various microphones/speakers. Accurate estimate of the overall background sound level in the lab is critical in determining appropriate criteria and noise control and avoiding unnecessary cost additions to a project. In some cases (such as makerspaces), ambient noise levels of idling equipment may exceed general ASHRAE guidance and still be acceptable to the occupants, even for a teaching application. In other applications (such as R&D testing), appropriate noise criteria may be significantly lower than ASHRAE's guidance. This presentation will discuss recent observation and design experiences in evaluating noise levels and noise sensitivity of makerspace, server chip design, and R&D laboratory spaces.

Session 2pPA

Physical Acoustics, Structural Acoustics and Vibration and Signal Processing in Acoustics: The Impact of Logan Hargrove on Physical Acoustics and Beyond

Veerle M. Keppens, Cochair
The University of Tennessee, 414 Ferris Hall, Knoxville, TN 37996

Ralph T. Muehleisen, Cochair
Energy Systems, Argonne National Laboratory, 9700 S. Cass Ave., Bldg 362, Lemont, IL 60439-4801

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—2:30

Invited Papers

2:35

2pPA1. Some physical acoustics research supported by Logan E. Hargrove. Julian D. Maynard (Phys., Penn State Univ., 104 Davey Lab., Box 231, University Park, PA 16802, maynard@phys.psu.edu)

As a Scientific Officer for the Office of Naval Research, Logan Hargrove would support research which addressed Navy problems, such as making ships quieter. Logan recognized that this problem could be tackled with nearfield acoustical holography (NAH), which, unlike conventional holography, could image sources with a resolution not limited by the wavelength of the radiation. NAH was cited as one the Navy's 35 big achievements in 75 years. A related problem was sound radiation from a rib-stiffened plate; in this case it was conjectured that a condensed matter physics concept, Anderson localization (cited in a Nobel prize), could be used to reduce noise radiation from a plate. Logan supported this research, and extended it to include research with other condensed matter concepts merged with acoustics. This led to an acoustic model of a quasicrystal formed with 150 musical tuning forks coupled in a Penrose tile pattern. While the Navy was indeed interested in novel materials such as alloy quasicrystals, Logan was pleased that part of his supported research, the tuning fork quasicrystal, was featured in the New York Times (September 5, 1989, Science Section).

2:55

2pPA2. Logan Hargrove and thermoacoustics. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg 362, Lemont, IL 60439-4801, rmuehleisen@anl.gov)

In his time as a Scientific Officer for the Office of Naval Research (ONR), Logan Hargrove supported a number of research areas in physical acoustics including non-linear acoustics, resonant ultrasound spectroscopy, sonoluminescence, and thermoacoustic heat engines and refrigerators. In the 1980s, 1990s, and 2000s, Logan provided support in thermoacoustics for dozens of students, postdocs, faculty, and researchers throughout the US resulting in dozens of papers, reports, and patents and spurring the creation of several small businesses. This presentation provides an overview of some of the thermoacoustics related projects and the lasting impact Logan's thermoacoustic funding has made.

3:15

2pPA3. Logan Hargrove and nonlinear acoustics. Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, hamilton@mail.utexas.edu)

Logan Hargrove became involved in nonlinear acoustics as a master's student at Oklahoma State University under George Thurston, research that led to the 1957 JASA paper "Nonlinear Properties of Circular Orifices." His involvement continued as a doctoral student under Egon Hiedemann at Michigan State University, where he published the 1960 JASA letter "Fourier Series for the Finite Amplitude Sound Waveform in a Dissipationless Medium." The letter was part of his research on the use of optical methods to characterize sound fields, which culminated in Hargrove and Achyuthan's 1965 Physical Acoustics series chapter "Use of Light Diffraction in Measuring the Parameter of Nonlinearity of Liquids and the Photoelastic Constants of Solids." Following his subsequent employment at Bell Laboratories, where he worked primarily in optics and reported the first demonstration of a mode-locked laser, Logan became a scientific officer at the Office of Naval Research and began funding basic research in nonlinear acoustics. This presentation provides an overview of

the research projects and graduate students in nonlinear acoustics that were funded by Logan at UT Austin for nearly three decades. Part of Logan's legacy is that many students he funded perform research in nonlinear acoustics to this day.

3:35

2pPA4. Illuminating finite-amplitude wave propagation: Hargrove's use of recurrence relations. Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu)

A classic problem in the theory of finite-amplitude wave propagation is that of the spectral evolution of a plane wave. An initial sinusoidal wave becomes triangular, with a vertical face at the shock distance. The wave in the pre-shock regime was represented by a Fourier series in the seminal work by Fubini, i.e., Fubini Ghiron [Alta Frequenza, 4, 530 (1935)]. Independently, Hargrove performed a similar derivation, also invoking recurrence relations for cylindrical Bessel functions, but presenting this with exceptional heuristic clarity and brevity [*J. Acoust. Soc. Am.* **32**, 511 (1960)]. This was coincident with a third paper, by Keck and Beyer [*Phys. Fluids* **3**, 346 (1960)], which knowingly followed Fubini. All of this work has facilitated understanding, providing cogent reminders of the power of recurrence relations when dealing with special functions.

3:55–4:10 Break

4:10

2pPA5. Selected research in physical, structural, and underwater acoustics at WSU associated with Logan Hargrove's ONR scientific program. Philip L. Marston (Phys. & Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Logan Hargrove, during his service as an ONR scientific and program officer, facilitated basic research at Washington State University for two decades. The Hargrove era investigations included research into: acoustic scattering physics and holographic imaging for elastic objects in water, high-order caustic wave-fields, generation and holographic-verification of acoustic vortex beams, light scattering by bubbles and drops, acoustical and optical radiation forces, nonlinear wave interactions in heterogeneous media, and cavitation induced by shock-wave reflection. The scattering research progressed from symmetric objects to objects of greater complexity. While the bulk of the funds provided went to support graduate researchers, some laboratory improvements of lasting value were also facilitated through the ONR support. The impact of that support on ONR's research interests and on the development of careers of several students who went on to make long term contributions to those interests will be summarized. While the emphasis is on the aforementioned two decades, some subsequent developments dependent on the facility improvements are also noteworthy. Recollections of some personal encouragement provided by Logan prior to receiving ONR support will also be noted. [Research supported by the Office of Naval Research.]

4:30

2pPA6. The role of Logan Hargrove in the creation of the National Center for Physical Acoustics. Lawrence A. Crum (Appl. Phys. Lab, Univ. of Washington, 4662 175th Ave. SE, Bellievue, WA 98006, lacuw@uw.edu)

Shortly after I arrived at the University of Mississippi (UM), Hank Bass proposed that any member of the Ole Miss faculty with an interest in acoustics join the Physical Acoustic Research Group at the University of Mississippi (PARGUM), and, informally constituted, we all agreed to devote 10% of our time to the promotion of this entity. One of the first items of business was to increase our available laboratory space. On one of Logan's visits, Logan agreed to speak to the Chancellor and said that, within his power, he would continue to provide ONR support for PARGUM. This meeting with the Chancellor was instrumental in the assignment of the Ole Band Building to PARGUM, but more importantly, the Chancellor became supportive of Hank's initiative to seek the engagement of Mississippi's powerful congressional delegation in creating a national center for acoustics. With the enthusiastic support of the Chancellor, NCPA was created by an Act of Congress in 1986.

4:50

2pPA7. Logan Hargrove's greatest contribution to the University of Mississippi. Richard Raspet (NCPA, Univ. of MS, 145 Hill Dr., University, MS 38677, raspet@olemiss.edu) and Craig J. Hickey (NCPA, Univ. of MS, University, MS)

We believe that Logan always felt that the people and community of Physical Acoustics were the most important product of his ONR research program. In particular, the PhD students who were able to receive degrees and continue to work in Physical Acoustics after graduation were dear to him. A characteristic of his funding was that he always provided sufficient funds for tuition, salary for the student, summer salary for the PI, and travel to ASA meetings to present the work and to interact with others PI's and students in Physical Acoustics. We are very thankful for his support and will talk about the students supported by Logan at the University of Mississippi over the last four decades.

5:10

2pPA8. The critical role of Logan Hargrove in establishing the physical acoustics summer school. Joseph Gladden (NCPA / Office of Res., Univ. of MS, PO Box 1848, University, MS 38677, jgladden@olemiss.edu)

First established in 1988, the Physical Acoustics Summer School (PASS) was patterned off of a long tradition in European scientific communities of carving out a week or more for established scientists and graduate students to come together in a remote location in order to focus on the state of the science and build community between existing and emerging generations of thought leaders. PASS has convened every two years from that time until a global pandemic disrupted that in 2020. During those early years, Logan Hargrove was an enthusiastic and energetic supporter of physical acoustics research, but had a vision to build a program to support the field beyond funding individual research projects through ONR—he wanted to support a vibrant and long-term physical acoustics community. Logan's vision is now a well-established reality and still guides the planning and organization of PASS today. In this talk, I will present some of the history of PASS and its evolution over the years with a focus on Logan's critical role all along the way.

Session 2pPPa

Psychological and Physiological Acoustics and Animal Bioacoustics: Auditory Perception of Stationary and Moving Sounds II

Michaela Warnecke, Cochair
Audio Team, Facebook Reality, 9845 Willows Road NE, Redmond, WA 98052

Cynthia F. Moss, Cochair
Psychological and Brain Sciences, Johns Hopkins University, 3400 N. Charles Street, Ames 200B, Baltimore, MD 21218

Contributed Papers

1:00

2pPPa1. Echolocating bats modify biosonar emissions when avoiding obstacles during difficult navigation tasks. Andrea M. Simmons (Cognit., Linguist., & Psychol. Sci., Brown Univ., 190 Thayer St., Box 1821, Providence, RI 02912-9067, Andrea_Simmons@brown.edu), Charlotte R. Thorson, Madeline McLaughlin, Pedro Polanco (Neurosci., Brown Univ., Providence, RI), Amaro Tuninetti (Cognit., Linguist., & Psychol. Sci., Brown Univ., Providence, RI), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

Big brown bats modulate their biosonar emissions in both time and frequency when flying through acoustically cluttered environments. We challenged single bats or bats in pairs to fly along a narrow U-shaped corridor 7 m in path length formed by rows of vertical plastic chains, while also avoiding additional interspersed horizontal chains. Bats emitted calls in complex sonar sound groups in all flight conditions. When no added horizontal obstacles were present, each bat's flight success rates averaged 80%; introducing horizontal obstacles decreased success rates to 45-65%, with little improvement over several days. When two bats flew simultaneously from opposite ends of the U-shaped path, they rarely collided, instead swerving closely past or keeping farther apart in elevation. They increased call rate and call amplitude, while decreasing terminal FM broadcast frequency, as they approached near the middle of the U-shaped path. Increases in call amplitude persisted throughout the remainder of the flight while call rate and terminal frequency returned to baseline levels. These data show the limits of call modifications during difficult maneuvers in dynamic conditions. They demonstrate the bat's effective awareness not only of the continuously distributed clutter but also of another, moving bat. [Work supported by ONR.]

1:15

2pPPa2. Egyptian fruit bats exhibit precise 3D spatial memory that eclipses sensory information. Xiaoyan Yin (Psychol. and Brain Sci., Johns Hopkins Univ., 460 Grindall St., Baltimore, MD 21230-4128, xyin17@jhu.edu) and Cynthia Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Egyptian fruit bats (*Rousettus aegyptiacus*) have well-developed vision and lingual echolocation, which provide complementary sensory information to execute natural behaviors, such as obstacle avoidance and landing. However, spatial memory can reduce the computational load in natural tasks and there is evidence that bats sometimes fail to use sensory information when they operate in familiar surroundings. Here we investigate the precision of this bat species' spatial memory for a landing location when it has access to vision and sonar echoes and when it has access to sonar echoes alone. Animals were trained for about 10 days to find a 5 cm diameter landing platform at a fixed location in a 2.5×6 m^2 flight corridor and later tested when the landing platform was shifted in azimuth and elevation by 5–30 cm. The bat's 3D flight trajectory was tracked with high-speed video cameras and its sonar inspection of objects was quantified with an ultrasound microphone array that permits reconstruction of the sonar beam aim. Our data suggest that spatial memory in Egyptian fruit bats is highly precise and eclipses multimodal sensory information. This bat species' robust reliance on 3D spatial memory may support long-distance navigation and forage in the natural environment.

1:30

2pPPa3. Comparison of discrete and continuous auditory detection for real and simulated over-flight signals. Frank S. Mobley (711 Human Performance Wing, Air Force Research Lab., Wright-Patterson AFB, OH 45433, frank.mobley.1@us.af.mil), Eric R. Thompson (711th Human Performance Wing, Human Systems Directorate, Air Force Res. Lab, Wright-Patterson AFB, OH), Margaret H. Ugolini (Ball Aerosp. Technologies Corp., Wright Patterson Air Force Base, OH), Elizabeth L. Fox (U.S. Air Force, Air Force Res. Labs, Wright-Patterson Air Force Base, OH), and Hilary Gallagher (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

Many auditory detection tasks employ discrete stimulus samples, presented during well-defined, brief observation intervals at predictable times. However, real-world listening is a continuous process without clearly defined intervals possessing indicators of when to listen and respond. The current study was designed to investigate the differences between a discrete, interval-based method and a continuous, free-response method. Increasing understanding of differences in methodology leads to a model predicting real-world perception from interval-based laboratory detection data. Real and simulated aircraft over-flight signals were presented to human subjects in a continuous, free-response task in which the subjects indicated when they could detect the signal. The same signals were also divided into 1-s audio files, which were presented to the same subjects in different sequences using a single-interval, forced-choice paradigm. The results indicated that the presentation sequence does not have an effect on human performance. Comparison of psychometric function derived from interval-based and free-response data indicated a shift to higher signal-to-noise ratios on approach when no clear response interval is provided, but the same shift was not observed on departure.

1:45

2pPPa4. Virtual sound source distance evaluation in acoustically and visually incongruent contexts. Vincent Martin (UMR 9912 STMS-CNRS-UPMC, IRCAM, 1 Pl. Igor Stravinsky, IRCAM, Paris 75004, France, vmartin@ircam.fr), Olivier Warusfel, and Isabelle Viaud-Delmon (UMR 9912 STMS-CNRS-UPMC, IRCAM, Paris, France)

Visual and acoustic environment may influence the perception of auditory distance. In the context of Audio-only augmented reality (AAR), the coherence of the perceived virtual sound sources with the apparent room

geometry and acoustics cannot always be guaranteed. The perceptual consequences of these incoherences are not well known. We conducted two online perceptual studies with a sound distance rendering model based on measured spatial room impulse responses (SRIR). A first study evaluated the perceptual performances of the model in incongruent visual contexts. The incongruent environment-related visual cues (spatial visual boundary and room volume) demonstrated a significant effect on the auditory distance perception (ADP) of virtual sound sources, through a calibration effect. A second study evaluated the impact of acoustical incongruence. Virtual sound sources distances were judged after the participant listened to distracting sound sources conveying distance cues relative to a different acoustical environment. When this distracting sound sources corresponded to a larger room than the one reproduced by the model, a higher compression effect was observed on the ADP of virtual sound sources. However, when the intensity cue conveyed by the distracting sound sources were coherent with the acoustical environment simulated by the model, their distracting effect were negligible.

2:00

2pPPa5. The quantification of head-related transfer function's dependency on anthropometric features. Ganesh Kailas (Indian Inst. of Technol. Kanpur, IIT Kanpur, Kanpur, Uttar Pradesh 690501, India, ganeshgk@iitk.ac.in), Voona Satish Kumar, and Nachiketa Tiwari (Indian Inst. of Technol. Kanpur, Kanpur, India)

The work quantifies the anthropometric dependency of the head-related transfer functions(HRTFs). The human body modifies the soundwaves while traveling to the ear canal from different spatial locations, and the HRTF defines the transformation. The peaks and notches in the HRTF magnitude spectrum, generated by the complex asymmetrical shape of the outer ear (pinna), are essential for sound localization in the elevation plane. However, the anthropometric dependency of HRTF is not well quantified in the literature. The huge experimental cost of conventional HRTF measurement methods is a hurdle in these studies. To overcome this, distance-dependent HRTFs have been evaluated using the finite element-based computational tool on surface meshes of the human upper body with different pinnae. The size and interaural position of the pinna were altered, and various HRTF simulations were carried out in the study. The frequency shift of the spectral peaks and notches in the magnitude spectrum was examined to understand its anthropometric dependency. The dependency of HRTF on human anthropometric features was appraised in the proposed work. It can be beneficial in predicting personalized HRTF according to the size and interaural position of the subject's pinna.

2:15

2pPPa6. Measurements and models for tone localization by moving listeners. William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu) and Eric Macaulay (Michigan State Univ., East Lansing, MI)

Listeners were given the challenging task of localizing a steady-state sine tone in a moderately reverberant room. Listeners attempted to optimize performance by moving during the nine-second tone interval. To discover listener localization strategies, the experiment acquired complete records of listener head motion (Polhemus head tracker) and the signals in the ear canals (Etymotic probe tubes). Experiment data were used to test two models of sound localization by moving listeners: According to the "Relative Null model, listeners search for nulls in relevant interaural differences and use the momentary nose orientations where these nulls occur as indicators of source azimuth. According to the "Inferred Azimuth model, listeners acquire a continuous, long-term record of inferred source location based on momentary head orientation combined with corresponding interaural differences. Models include different cue weighting assumptions and are evaluated by their ability to predict listener responses. The Relative Null model proves to be more accurate, but sometimes fails to make a prediction. The Inferred Azimuth model always makes multiple predictions, and often leads to further insights. These models are the most straightforward of numerous possibilities. Alternatives that weight the time course of listener cues, should be considered.

Session 2pPPb

Psychological and Physiological Acoustics: Psychoacoustics and Neural Processing (Poster Session)

Ellen Peng, Chair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68131

All posters will be on display from 2:45 p.m. to 4:45 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:45 p.m. to 3:15 p.m. and authors of even-numbered papers will be at their posters from 3:15 p.m. to 4:45 p.m.

Contributed Papers

2pPPb1. Counting the steps for a loudness scale. Lance Nizami (Independent Res. Scholar, 5302 Avalon Dr., Bedford, MA 01730, lancenizamiphd@comcast.net) and Claire S. Barnes (Independent Res. Scholar, Bedford, MA)

It is a century-old desire to construct a scale for loudness. Such a scale's intervals would, on face value, be just-noticeable changes in loudness. Those changes correspond to empirical just-noticeable-differences (jnds) in RMS sound-pressure-level. Hence, a putative loudness scale is the cumulative number of jnds, counted upward from an auditory stimulus's absolute detection threshold (i.e., the just-perceptible stimulus). Jnd-counting has been intermittently pursued for over a century. The historical method is mathematical integration, but integration inherently assumes infinitely fine jnds, an absurdity. A novel, discrete method of jnd-counting was pioneered for hearing by Dallos and Carhart, after a step-counting method developed for genetics by the polymath J. B. S. Haldane. Nonetheless, one element of the integration method remains necessary: an equation for jnd versus stimulus level. The literature offers such equations in hearing, vision, and the other senses. Unfortunately, Dallos and Carhart omitted crucial details from their exposition, and employed a unique approximation for the jnd, expressing it versus decibels above absolute detection threshold, rather than versus RMS sound-pressure-level. Perhaps consequently, the method lacked acceptance. Here, the method is clarified by generalizing through full details, and then applied to a variety of jnd equations. Non-constant loudness intervals can be accommodated, if desired.

2pPPb2. Forward masking of amplitude modulation detection includes a large perceptual effect. Christopher Conroy (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cwconroy@bu.edu), Andrew J. Byrne, and Gerald Kidd (Boston Univ., Boston, MA)

Forward masking of amplitude modulation (AM) detection is tuned: target sinusoidal AM (SAM) is masked more easily by masker SAM with a similar, rather than dissimilar, rate. This finding has been interpreted either as a sensory effect related to adapting neural units tuned to the target's AM rate, or as a perceptual effect related to the perceived similarity of the acoustically similar AM. This study investigated these two alternative possibilities. The minimum modulation depth required to detect 96-Hz target SAM was measured following exposure to suprathreshold masker AM applied to the same broadband-noise carrier. Three forward masker modulators were tested: (1) 32-Hz SAM; (2) a 32-Hz square wave; (3) and 96-Hz SAM with a modulation index equivalent to that of the third harmonic of the square wave. The 96-Hz masker SAM produced substantial amounts of forward masking; the 32-Hz sine- and square-waves, however, did not. These findings were inconsistent with the hypothesis that detection of the target SAM was mediated by adapting neural units tuned to the target's AM rate, suggesting, instead, that forward masking of AM detection includes a large perceptual effect.

2pPPb3. Differentiating perceptual, procedural, and task learning for an auditory temporal discrimination task. Leslie Zhen (Commun. Sci. and Disord., Univ. of Pittsburgh, 4028 Forbes Tower, Rm. 5011, Pittsburgh, PA 15213, lez35@pitt.edu) and Sheila Pratt (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Perceptual or stimulus learning (PL) reflects experience-driven improvements in the ability to detect changes in stimulus characteristics. The time course for PL overlaps with that for procedural learning (acquiring general skills and strategies) and task learning (learning the perceptual judgment specific to the task), making it difficult to isolate their individual effects. This study was conducted to examine the impact of exposure to stimulus, procedure, and task information on learning for auditory temporal-interval discrimination. Eighty-three listeners completed multiple sessions requiring auditory temporal-interval discrimination (target learning task), but before these sessions listeners were exposed to one of three conditions that differed by the information conveyed about the target task's stimulus, procedure, or task characteristics. Significant learning occurred but an exposure effect was not found when examined across sessions. However, percentage improvement scores (initial to final) revealed that listeners exposed to elements of the target task demonstrated larger percentage improvements on that task compared to naïve listeners without any prior experience in a manner that suggested that stimulus and procedural learning were possible drivers of temporal learning. These findings could clarify the influence of experience on temporal PL and inform future research aimed at developing training paradigms that optimize perceptual improvements.

2pPPb4. Performance evaluation on three-forced choice pure tone audiometry in open space environment. Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), Fiqiyah U. Azmi, Naomi Ashilah, Bagus A. Herlambang (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya, Indonesia), and Nyilo Purnami (Faculty of Medicine, Airlangga Univ., Surabaya, Indonesia)

In this report, we investigated the performance comparison of Three-Forced Choice (3AFC) with a pair of Bluetooth-connected active noise cancelling headphones and Two-Forced Choice (2AFC) in an audiological booth for audiometry examination. The procedure was essentially to generate mono pure tone on six different frequencies with adaptive loudness level to both left and right ear, respectively. The subject presented the pure tone, then he or she pressed a button. If it is right than the loudness level will be lowered until it is inaudible. This level was marked as the hearing threshold of the subject. We recruited 38 normal-hearing subjects who underwent to an audiology clinic for audiometry examination (using the gold standard, 2AFC technique). The recorded responses were statistically analyzed using non-parametric Wilcoxon test with Bonferroni correction. The audiometry results using the outdoor 3AFC method on 2000 Hz and 4000 Hz have mean-differences closer to ISO 226 accuracy reference for all 38

participants. These results suggested that the proposed technique may replace the gold standard.

2pPPb5. Detection of aircraft targets varies as a function of target and masker temporal complexity, similarity, and realism. Margaret H. Ugolini (Ball Aerosp. Technologies Corp., 2610 Seventh St., Bldg 441, Wright Patterson Air Force Base, OH 45433, margaret.ugolini.ctr@us.af.mil), Frank S. Mobley (711 Human Performance Wing, Air Force Reseach Lab., Wright-Patterson AFB, OH), and Eric R. Thompson (711th Human Performance Wing, Human Systems Directorate, Air Force Res. Lab, Wright-Patterson AFB, OH)

The complexity of real-world auditory scenes makes it difficult to disentangle the factors that drive detectability of sounds within those scenes. However, many real-world applications of auditory perception research require the disentanglement of these factors – for example, to predict the effects of aircraft noise on the public depending on the ambient sound environment and the aircraft itself. The present study investigates effects of masker temporal complexity and similarity to targets on the detectability of helicopter target sounds. We investigate three levels of masker temporal complexity – urban sound scenes (high); synthetic versions of each scene that preserve overall statistics (medium); and noise of the same spectrum as the real scene (low). Target similarity was probed in a way that would occur in the world—as a function of the presence of other engine sounds. Each real masking sound could be comprised of only engine noise, engine sounds embedded in an urban scene, or a scene without engines. This provides maskers with three levels of target similarity. Combination with the three levels of temporal complexity allows us to probe interactions between temporal complexity and engine-related signal energy. Effects of target signal characteristics and interactions with signal to noise ratio will be discussed.

2pPPb6. Dissociating sensitivity from bias in the mini profile of music perception skills. Kelly L. Whiteford (Psych., Univ. of Minnesota, N218 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55414, whit1945@umn.edu), PuiYii Goh (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Kara Stevens (Psychiatry & Behavioral Sci., Univ. of Minnesota, Minneapolis, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

The Mini Profile of Music Perception Skills (Mini-PROMS) is a widely used objective measure of musical competence that has been validated for online data collection [Law and Zentner, PLoS One **7**, e52508 (2012); Zentner and Strauss, Ann. NY. Acad. Sci. **1400**, 33–45 (2017)]. This measure involves determining whether a comparison melody is the same or different from a standard on a 5-point scale ("definitely different", "probably different", "I don't know", "probably same", or "definitely same"). The traditional PROMS scoring method is to calculate a composite score, which differentially weights correct responses based on participant confidence. In this method, a high score can only be achieved with a confident response bias. The present study employed signal detection theory to calculate the participants' sensitivity (d'), which is a bias-free measure of sensitivity if the signal and noise distributions (1) are normally distributed and (2) have roughly equal variance. 74 participants were tested on the Mini-PROMS with the purpose of calculating empirical ROC curves to test whether the assumptions of normality and variance are met. The study findings will provide important insight into how using different analysis methods might improve the validity of the Mini-PROMS. [Work supported by NIH grant R01 DC005216.]

2pPPb7. Effects of modulator shape at thresholds and supra-thresholds spectral modulation perception. Sittiprapa Isarangura (Commun. Sci. and Disord., Faculty of Medicine Ramathibodi Hospital, Mahidol Univ., 270 Thanon Rama VI, Thung Phaya Thai Subdistrict, Ratchathewi District, Bangkok 10400, Thailand, sittiprapa.isa@mahidol.edu), Ann C. Eddins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), Aaron Seitz (Psych., Univ. of California, Riverside, Riverside, CA), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL)

Spectral modulation detection is common method to assess spectral envelope perception in research and clinical settings. Prior investigations into spectral modulation (SM) detection have used two different modulator shapes: a sinusoidal envelope on a logarithmic amplitude scale or a sinusoidal envelope on a linear amplitude scale. The former creates a sinusoidally shaped excitation pattern, while the latter creates an excitation pattern with broad peaks and sharp valleys. What impact, if any, modulator shape has on SM perception at detection threshold or at supra-threshold levels is unknown. To investigate potential perceptual differences between waveform shapes: (1) SM detection was measured for modulation frequencies ranging from 0.5 to 8.0 cycles/octave; (2) Spectral modulation masking was measured for SM frequencies of 0.5 and 2.0 cycles/octave. The SM phase of the masker was manipulated (0, 90, or 180 degrees) so that maskers had a peak, a midpoint, or a valley at 1147 Hz. Tone detection (1147 Hz) was measured for the logarithmic and linear sinusoidal SM envelope shapes for each of the three phases. Results revealed significant differences between the two spectral envelope shapes for both spectral modulation detection and for spectral modulation masking. [Work supported by NIH R01DC015051].

2pPPb8. Attention effects on cortical neural processing in spatial listening. Fan-Yin Cheng (Speech, Lang., and Hearing Sci., Univ. of Texas at Austin, 2504A Whitis Ave. (A1100), Austin, TX 78712-0114, fanyin.cheng@utexas.edu), Can Xu, Heather Goodall, Miah Elise Ornelas, Lisa Gold, and Spencer Smith (Speech, Lang., and Hearing Sci., Univ. of Texas at Austin, Austin, TX)

Speech understanding in complex acoustic scenes relies on the formation of and attention to auditory objects. Spatial information is an essential cue in the formation of auditory objects and may facilitate attention-based neural enhancement and suppression of target and non-target speech, respectively. Previous studies have used cortical auditory evoked potentials (CAEP) to measure how attention modulates neural representation of spatially separated auditory objects. The current study further extends this work to investigate how expected and unexpected shifts in spatial location of speech signals affect neural processing of these signals. In the experiment, listeners were directed where to listen with a spatial cue presented from one of five locations (±90, ±45, or 0 deg azimuth); the following target sequence was either presented in the cued location (congruent trials) or from a non-cued location (incongruent trials). We hypothesized that neural responses to target sequences would be enhanced when presented from the cued location, whereas neural responses to incongruent sequences would initially be suppressed before spatial attention was shifted toward the target. Further, we expected the size of this suppression to be proportional to the size of the relative shift between cue and incongruent target.

2pPPb9. Neural correlates of bilateral summation between natural and vocoded speech. Francis X. Smith (Otolaryngol., Univ. of Iowa, W311 Seashore Hall, Iowa City, IA 52242, francis-smith@uiowa.edu), Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), Ruth Y. Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI), and Inyong Choi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

The normal auditory system integrates redundant information across ears, which aids listeners to detect weak sounds. For patients with single sided deafness (SSD), cochlear implants (CI) have been attempted expecting improved speech recognition in challenging listening scenarios. However, it is not yet known whether CI recovers the bilateral summation for weak sound detection in SSD. The binaural benefit of soft-speech recognition was tested in six SSD CI patients and thirty normal hearing (NH) listeners. The

task was identifying quietly presented (i.e., RMS ∼6 dB SL) monosyllabic words. CI patients were tested in three conditions: acoustic-ear only (A1), electric-ear only (E1), or both ears (A+E). NH participants were tested in four conditions: monaural natural speech (A1), monaural vocoded speech (E1), dichotic natural and vocoded speech (A+E), or dichotic natural speech (A2). Electroencephalography was recorded during the task. Bilateral conditions (A+E and A2), even when one ear was vocoded, resulted better accuracy than monaural conditions (A1 and E1). However, A+E condition exhibited smaller early auditory cortical responses (i.e., N1P2 complex) than any natural-speech conditions (A2 and even A1). This result may imply that bilateral summation benefits in SSD-CI users are achieved by later cognitive processing rather than immediate early integration.

2pPPb10. Energetic/informational masking and listening effort, as measured by electroencephalography and pupillometry. Sarah Villard (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, svillard@bu.edu), Ayesha Alam, Tyler K. Perrachione, and Gerald Kidd (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Measuring the *listening effort* exerted by an individual to understand speech under auditory masking conditions can provide vital information that is not available from accuracy scores alone. The goal of this study was to compare the amount of listening effort elicited in young, normal-hearing subjects during a noise masking condition producing primarily energetic masking (due to spectrotemporal overlap between target and maskers) versus during an intelligible speech masking condition producing a high degree of informational masking (additional masking due to listener uncertainty; e.g., target-masker confusions). Listening effort was measured simultaneously using two physiological metrics: alpha power (via electroencephalography) and pupil dilation. The target speech comprised 5-word matrix-style sentences in both conditions. The target was presented at 0 deg azimuth, and the maskers were presented at ± 45 deg azimuth. In each condition, the target-to-masker ratio was set at each subject's 75% correct point on the psychometric function. Preliminary results suggest that the intelligible speech masking condition elicits a higher degree of listening effort than the noise condition. This finding is consistent with the view that greater effort is involved in ignoring acoustically and linguistically similar sources than highly dissimilar, low-information value sources. [Work supported by NIH K99DC018829.]

2pPPb11. An auditory-like region in the motor cortex. Christian Herrera Ortiz (Res. Services, Loma Linda Veteran's Assoc. for Res. and Educatoin, 11201 Benton St., Loma Linda, CA 92357, christian.herreraortiz@va.gov), Nicole Whittle (Res. Services, Loma Linda Veteran's Assoc. for Res. and Educatoin, Loma Linda, CA), Marjorie R. Leek (Res., Loma Linda VA Healthcare System, Loma Linda, CA), Samuel Barnes, Barbara Holshouser (Loma Linda Univ., Loma Linda, CA), Alex Yi, and Jonathan H. Venezia (Res., Loma Linda VA Healthcare System, Loma Linda, CA)

Left premotor cortex is activated during listening to speech, but whether this reflects a motor mechanism or a response to auditory features is debated. Previously, we showed that left dorsal premotor cortex (dPM) responded to vocal pitch in a degraded speech recognition task, but only when speech was rated as unintelligible. Crucially, vocal pitch was not relevant to the task. Here, we hypothesize that left dPM will respond to vocal pitch for increasingly intelligible speech in a competing speech task that emphasizes pitch for talker segregation. We use fMRI ($N = 25$) and apply spectrotemporal modulation distortion to modulate pitch in two-talker (male/female) mixtures across two conditions (Competing, Unison), only one of which requires pitch-based segregation (Competing). A Bayesian drift-diffusion model was used to predict speech recognition performance (3-AFC response times) from the pattern of spectrotemporal distortion imposed on each trial. The model's 'drift rate' parameter, a d'-like measure, was strongly associated with vocal pitch for Competing but not Unison. Trial-wise predictions of 'pitch-restricted' drift rate—i.e., that component of performance driven only by pitch – were positively associated with activation in left dPM for Competing but not Unison. These findings show that left dPM responds to auditory features during speech recognition.

2pPPb12. Dynamic functional connectivity in auditory attention task. Jordan Drew (Elec. and Comput. Eng., Univ. of Washington, 185 E Stevens Way NE, Seattle, WA 98195, jadrew43@uw.edu), Eric Larson (Speech & Hearing Sci., Univ. of Washington, Seattle, WA), Nicholas Foti, Rahul Nadkarni, Emily Fox (Comput. Sci. and Eng., Univ. of Washington, Seattle, WA), and Adrian KC Lee (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Navigating conversations in complex auditory environments requires dynamic control of attention—switching and maintaining attention across multiple auditory objects. Several brain regions, some traditionally considered auditory and others non-auditory, have been highlighted as part of the cortical networks that contribute to the ability to switch and maintain attention between multiple talkers. M/EEG data were collected during a dual-stream auditory attention task in which the listener was asked to attend to one of two speech streams that differed in location or pitch. In each trial, the listener was cued to either maintain their attention on one talker, or switch their attention to the second talker during a silent interval. In this study, we apply a state-space model to the brain imaging data to elucidate the differences in the dynamic cortical networks responsible for maintaining and switching attention across different auditory cues.

2pPPb13. Contralateral noise degrades frequency-coding accuracy in normal-hearing adults. Fuh-Cherng Jeng (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W231, Athens, OH 45701, jeng@ohio.edu), Kelley A. Stehura, Breanna N. Hart, and Allison T. Giordano (Commun. Sci. and Disord., Ohio Univ., Athens, OH)

Noise presents a constant challenge in daily communication. Previous research has demonstrated the effects of ipsilateral noise on speech processing in the human brain. In this study, we examined the effects of contralateral noise on the accuracy of frequency coding at the subcortical level. By using a speech stimulus that mimicked the English vowel /i/ with a rising frequency contour, we obtained scalp-recorded, frequency-following responses in nine normal-hearing adults. To determine the effects of contralateral noise, we performed two experimental conditions: with and without the presence of contralateral noise. Results indicated that the fidelity of frequency coding, as reflected through *Tracking Accuracy* and *Slope Error*, were significantly degraded when continuous white noise was added to the contralateral ear. These findings provide important information and help us better understand how frequency cues are processed in noisy environments.

2pPPb14. Feature dependence of neural mechanisms for divided auditory attention. Kayla Howerton (Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, kayla-howerton@uiowa.edu), Yuqi Deng (Biomedical Eng., Boston Univ., Boston, MA), Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA), and Inyong Choi (Univ. of Iowa, Iowa City, IA)

Multi-talker social situations require the listener to focus attention on a single talker, as well as monitor and divide attention to additional talkers, as needed. The neural mechanisms behind how one can efficiently divide their attention has yet to be thoroughly explored. The current study investigates these neural mechanisms via electroencephalography when participants listen to competing speech and focus or divide attention. Participants were tasked either to focus and maintain attention on an initial target or to divide attention, listening to the initial target but monitoring for a potential, later "supertarget." At the end of each trial, listeners recorded the target stimulus, defined either by location or talker identity (male or female), via a keypad. A pilot study found that focused, location-based attention modulates alpha oscillations in parietal cortex, thought to reflect inhibition of non-target locations. Here, we test the hypothesis that, during talker identity trials, attention enhances the neural representation of the target via talker identity features through enhanced beta oscillations in the frontal and middle temporal gyrus. Comparison of electroencephalography between divided and focused attention conditions suggests that using different acoustic features to focus or divide attention recruits completely different neural mechanisms.

2pPPb15. Hemispheric lateralization during spectral envelope perception is dynamic and individualized. Ann C. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, aeddins@usf.edu), Sittiprapa Isarangura (Commun. Sci. and Disord., Mahidol Univ., Ratchathewi District, Bangkok, Thailand), and Robert A. Lutfi (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL)

The spectral envelope is an important acoustic cue for vowel and consonant perception. Sensitivity to the spectral envelope is commonly studied with a spectral modulation detection task in which two co-occurring spectral cues (spectral modulation—SM, carrier frequency—CF) may be available. Neuroimaging studies indicate that spectral cues are often processed asymmetrically in the cortex, but it is unknown if such lateralization depends on perceptual weighting of the available cues. Here we simultaneously measured SM discrimination and electroencephalography (EEG) in young adult listeners in a single-interval, two-alternative task during passive and active attention conditions as well as in a separate perceptual decision-weights task. Using condition-on-a-single-stimulus (COSS) analyses, individual differences in perceptual weights of spectral cues (SM, CF) were obtained for each listener. EEG source localized results showed that hemispheric lateralization was dynamically modulated over time between left and right auditory cortices and that lateralization was dependent on individual perceptual weighting patterns for SM and CF. [Work supported by NIH/NIA P01-AG09524 (ACE) and NIH/NIDCD R01- DC001262 (RL, ACE).]

2pPPb16. Sensitivity to Schroeder phase arising from sensitivity to temporal patterns across nerve fibers. Hsin-Wei Lu (Neurosciences, KU Leuven, Leuven, Belgium), Philip H. Smith (Neurosci., UW-Madison, Madison, WI), and Philip X. Joris (Neurosciences, KU Leuven, Herestraat 49, Bus 1021, Leuven B-3000, Belgium, Philip.Joris@kuleuven.be)

The flat-envelope harmonic complexes devised by Schroeder (1970) have been extensively used in psychophysical studies. The stimuli contain a linear frequency sweep, whose direction and speed depend on phase curvature C. For a given characteristic frequency, basilar membrane and nerve responses show only a subtle dependence on the sign and magnitude of C. The question arises whether CNS neurons are sensitive to the temporal pattern across nerve fibers. We studied octopus cells in the cochlear nucleus, which receive a convergent excitatory input from nerve fibers and have been hypothesized to be monaural coincidence detectors. We studied both their spike output and nerve input using multiple recording techniques in anesthetized chinchillas. Octopus cells showed marked sensitivity to C, which was broadly stable across SPL and fundamental frequency. They were tuned to a wide frequency range, but with discrete frequency "hotspots" that generate major and minor inputs. The sensitivity to C arises not from coincidence detection, but from a sensitivity to the temporal sequence of activation of inputs tuned to different frequencies. We conclude that marked sensitivity to Schroeder phase is already present at the first CNS processing stage, based on temporal sequence detection across frequency channels rather than on coincidence detection.

2pPPb17. Neural correlates of top-down attention in auditory and visual contexts reflect individual differences in the population. Jasmine Kwasa (Neurosci. Inst., Carnegie Mellon Univ., 5000 Forbes Ave., Apt. 22, Pittsburgh, PA 15217, jkwasa@andrew.cmu.edu), Abigail Noyce, and Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA)

Individual variability in performance on selective attention tasks that require top-down control can be large. Recent neuroimaging and electroencephalography (EEG) experiments also suggest that auditory spatial selective attention engages the same fronto-parietal network used by visual spatial selective attention. We therefore hypothesized that individual differences in spatial selective attention would correlate for similar auditory and visual tasks and produce correlated changes in sensory responses in the two modalities. We calculated EEG-derived correlates of top-down, spatial attention during auditory and visual tasks that required subjects to sustain selective attention at one of two cued spatial locations (left or right) and report the auditory or visual sequence that appeared in that direction. Individual behavioral performance on the tasks correlated across sensory modalities. Spatial attention differentially modulated event-related potential N1 responses at the group level for both auditory and visual tasks. At the individual level, these attention-modulated N1 responses correlated with task performance for the auditory task; however, the ERP peaks were noisy and unreliable in the visual task. In contrast, the inter-trial coherence (ITC) in a time window encompassing the N1 correlated to performance for the visual, but not the auditory task. Our work highlights differences in how top-down attention modulates these representations across the two modalities.

Session 2pSA

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials II

Bogdan-Ioan Popa, Cochair
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Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin,

Alexey Titovich, Cochair
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Nathan Geib, Cochair
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Chair's Introduction—1:00

Contributed Papers

1:05

2pSA1. Surface manipulation of elastic half-space to suppress Rayleigh wave propagation: A step towards boundary condition-based meta-surface design. 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)

A boundary condition (BC)-based design methodology for metasurfaces was recently proposed to control low-frequency Lamb waves in a plate. This study highlights the significance of a type of Cauchy BCs, called Mindlin BCs, for controlling Rayleigh wave propagation and provides motivation for a BC-based metamaterial design for surface wave control. Analytical study on an elastic half-space with Mindlin BCs reveals no possible surface wave solutions. The frequency-domain and time-domain simulations by imposing Mindlin BCs on the surface in the path of Rayleigh wave propagation are consistent with analytical predictions exhibiting no surface wave transmission. The simulation results reveal mode conversions from Rayleigh wave to bulk waves directed into the half-space and a low amplitude Rayleigh wave reflection. For a finite-sized Mindlin BC patch, a portion of mode-converted bulk waves near the surface converts back to Rayleigh waves at the end of the BC patch suggesting the requirement of a minimum Mindlin BC patch length for effective suppression of surface waves. Finally, we show that frequency-dependent Mindlin BCs are imposed by an array of rod-like resonators resulting in a surface wave bandgap. These findings provide new insights into designing novel metasurfaces that provide the necessary coupling for desired surface wave control.

1:20

2pSA2. A finite element formulation to determine the complex wavenumber spectrum of an underwater leaky wave antenna. Craig W. Broadman (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, craigbroadman1@gmail.com), Benjamin M. Goldsberry (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Christina J. Naify (Appl. Res. Labs., The Univ. of Texas at Austin, Washington, DC), and Michael R. Haberman (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Underwater acoustic imaging traditionally utilizes phased arrays of electro-acoustic transducers to steer acoustic energy in specific directions. An alternative approach to steer acoustic beams is to use an acoustic leaky wave antenna (LWA). Acoustic LWAs consists of a single transducer attached to a dispersive antenna that enables frequency-dependent directionality of acoustic radiation and reception. Finite element analysis of an underwater LWA presents a unique computational challenge due to the presence of fluid loading from the surrounding water domain. Previous unit cell analyses performed on the underwater LWA approximated the surrounding fluid as a lumped mass on the surface of the LWA [Broadman *et al.*, J. Appl. Phys., **129**(19), 194902 (2021)]. In this work, we investigate LWA unit cell analysis that includes fluid-solid coupling and an exact absorbing boundary condition to enforce outgoing waves from the LWA. We solve for the complex wavenumbers of the propagating and leaky modes of the LWA given a real-valued frequency via a nonlinear eigenvalue problem, providing insight into the propagation and attenuation of elastic waves in the LWA. Finally, we compare the present approach with the lumped mass approximation and numerical results from a finite LWA. [Work supported by ONR.]

1:35

2pSA3. Thin Helmholtz resonators for a broadband acoustic metamaterial. Vidhya Rajendran (Architecture, ZGF Architects, 4348 9th Ave. NE, Apt. 210, Seattle, WA 98105, vidhya.rj28@gmail.com), Tomás I. Méndez Echenagucia (Dept. of Architecture, Univ. of Washington, Seattle, WA), and Andrew A. Piacsek (Phys., Central Washington Univ., Ellensburg, WA)

Acoustic absorption of the low to mid-frequency range of human speech (80 to 800 Hz) is important for open office environments and can be achieved with resonant absorbers. However, traditional Helmholtz Resonators (HRs) are not an efficient solution because of the large dimensions required and their limited acoustic performance. A heterogeneous array of novel HRs geometries that are compact and tuned to different frequencies function as a metamaterial that provides broadband absorption. A numerical study of the acoustic absorption of different resonator shapes is carried out to optimize performance in the target frequency. These geometries are compared based on their absorption coefficients, resonator dimensions, and their mass-productibility. The acoustic behavior of the HRs is modeled in COMSOL Multiphysics using the Finite Element Method and the efficient geometries were selected for designing the broadband acoustic metamaterials. The accuracy of the numerical model is tested experimentally in an impedance tube, using 20 cm $\times$ 20 cm panels fabricated with embedded resonator arrays.

1:50

2pSA4. Multiplexed acoustic holography using an iterative angular spectrum approach. Ian Meighan (Dept. of Mech. Eng., Rowan Univ., 201 Mullica Hill Rd., Glassboro, NJ 08028, meigha72@students.rowan.edu), Edward De Asis, and Chen Shen (Dept. of Mech. Eng., Rowan Univ., Glassboro, NJ)

Technological advancements have enhanced the capability of acoustic holography over the years. A major drawback for current techniques is the lack of ability of multiplexing, which has limited their potentials in ultrasound imaging and acoustofluidics. In this work, we propose an algorithm based on an iterative angular spectrum approach to overcome this limit. An optimized phase distribution at the source plane can be obtained which can project multiple acoustic projections at various distances and frequencies. As such, the acoustic fields at the image planes are encoded onto the same source plane and this facilitates the design of 3D printed acoustic kinoforms integrated on a single element transducer. The parameters including the frequency and projection distance to obtain the best performance are tested. The algorithm can be used in conjunction with other optimization strategies to fine tune pressure fields and further improve the image quality. The results are expected to be useful in medical ultrasound and acoustofluidics where complex acoustic field projection is needed.

2:05

2pSA5. Buckling-mediated phase transitions in nano-electromechanical phononic waveguides. Ali Kanj (Mech. Eng. Lab., Univ. of Illinois at Urbana-Champaign, 105 S. Mathews Ave., Urbana, IL 61801, alimk2@illinois.edu), Paolo F Ferrari (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), SunPhil Kim (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Berkeley, CA), Jonathan Bunyan (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Hillsboro, OR), Alexander F. Vakakis, Arend M. van der Zande, and Sameh Tawfick (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Acoustic signals in the megahertz to gigahertz regimes enable applications in phononic, electric and photonic circuitry. Hence, advancement in these multi-physics circuitries requires studying the transmission of elastic waves in micro/nano-electromechanical systems. Therefore, our work investigated the transmission of flexural waves in unidirectional phononic waveguides of coupled nano-electromechanical drumhead resonators. In particular, we studied the manipulation and tailoring of the transmission by thermal control of buckling within the waveguide. We showed—via experimental and computational evidence—that buckling tunes the frequency of transmission within the waveguide as determined by previous studies in literature. However, in our work, we demonstrated that excessive buckling stops the transmission in the fundamental (lowest frequency) passband of the waveguide, thus, inducing a phase transition in the waveguide leading to the absence of transmission in the respective frequency range. We computationally proved that this buckling-mediated phase transition necessitates the presence of weak inhomogeneities ($\sim$5% variations) within the waveguide, which are inevitable in any fabrication process. In fact, these inhomogeneities are amplified due to buckling-residual stresses leading to structural aperiodicity that only affects the fundamental passband by converting the extended shapes of the transmission modes into localized shapes that prevent signal propagating throughout the waveguide.

2:20

2pSA6. Wave propagation in a continuum nonlinear phononic material with soft nonlinear 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 (Mech. Sci. and Eng., Univ. of Illinois at Urbana Champaign, Urbana, IL)

This presentation discusses modeling the propagation of nonlinear waves in a continuum phononic material. This system is shown to exhibit amplitude-dependent response and energy transfer between frequencies, which are well studied phenomena of nonlinear discrete systems, but are relatively unexplored in fully elastic continuum models. The phononic material is composed of repeated layers of stiff and soft materials, and the nonlinearity is introduced through a nonlinear hyperelastic Gent model in the soft material. We analyze the dispersion of the linearized system using Bloch-wave analysis to identify band gaps in the system, and we use full-scale time-domain finite element simulations to analyze the dynamic behavior of the system. Generation of zero frequency and second harmonic frequency amplitudes as well as their accumulation with distance are observed, and the influence of phononic band gaps on nonlinear wave propagation are described. We explore the influence of material nonlinearity in soft materials in continuum phononic materials as a mechanism for control of nonlinear wave propagation.

2:35

2pSA7. Emergence of stegotons in phononic materials with strongly nonlinear rough contacts. Ganesh U. Patil (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, 144 Sidney Lu Mech. Eng. Bldg., MC-244, 1206 West Green St., Urbana, IL 61801, gupatil2@illinois.edu) and Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

The incorporation of nonlinearity in phononic materials enables complex wave interactions in both space and time enriching the dynamic response of the underlying linear media. In this talk, we discuss the strongly nonlinear wave response of continuum phononic material with periodic and discrete nonlinearity. The studied phononic material is a periodic architecture of contact interfaces with rough surfaces, connecting linear elastic layers. These contacts exhibit strong nonlinearity, stemming from variable contact areas under compressive loads and their inability to support the tensile loads. We reveal the evolution of propagating waves using finite element time-domain simulations. The interplay of strong nonlinearity, and dispersion, in the presence of elastic layers, generate traveling localized waves, referred to as "stegotons." Unlike classical solitons, these stagotons exhibit evolving spatial wave profiles and local variations in wave speed. Moreover, the elastodynamic effects arising from the nonlinearly coupled elastic layers enable strongly nonlinear energy transfer in the frequency domain by activating acoustic resonances of the layers. This study sheds light on the role of the strongly nonlinear coupling of linear media on the formation of traveling localized waves, and could open opportunities for enhanced dynamic response not possible with pure discrete or continuum phononic materials.

2:50–3:05 Break

3:05

2pSA8. Generating characteristic acoustic impedances with hydrogel based phononic crystals for use in ultrasonic transducer matching layers. Paul Daly (Univ. of Strathclyde, Rm. 319B, 204 George St., Glasgow G1 1XW, United Kingdom, paul.daly@strath.ac.uk), James Windmill, and Joseph Jackson (Univ. of Strathclyde, Glasgow, United Kingdom)

Quarter wavelength acoustic impedance matching layers (IMLs) are often attached to an ultrasonic transducer (UT) to enhance energy transmission into a load medium. Natural materials are rarely suitable as an IML but even composites with a suitable impedance can be excessively attenuating, or unsuitable for UT mounting. It would be beneficial then, if a non-absorptive IML with the specified impedance could be manufactured using common fabrication tools. Recent publications have shown that phononic crystal (PC) structures, made using modern printing and curing technology, can operate as IMLs and may satisfy these criteria. It is important to continue this research because it has the potential to improve UT performance, and simplify or surpass existing solutions. Here, finite element analysis (FEA) is used to validate a hydrogel-matrix / steel-inclusion PC IML that separates PZT and water domains. The PZT boundary is excited over a frequency range and the resultant in water pressure is averaged. Results are compared with an ideal, bulk IML and common, in practice materials. Comparison of spectra demonstrate that the PC increases in water pressure at the expected frequency, is an effective medium in the long wavelength regime, and is comparable to the ideal IML.

3:20

2pSA9. Magic phononic structures. Marupong Vongsiri, Maria Carrillo-Munoz (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)

To alter the dynamical response of a structure, engineers must alter its overall mass or stiffness, frequently at the expense of functional performance. Here, borrowing from the field of recreational mathematics, we propose the idea of magic structures—a new class of structures with invariant global mass and stiffness properties, but varying dynamical response. A magic square is a special array of numbers where all individual rows, columns, and diagonals sum up to the same constant value, called the magic number. Here, working within the context of phononic structures, we treat the total mass of the structure as its magic property. We study the dispersion behavior of an infinite, periodic spring-mass system with a 3×3 square unit cell with the magic mass property. While maintaining the magic mass, we study the effect of various mass distributions on the dispersion bands and phononic bandgaps. Our analysis shows that the frequency and width of the phononic bandgaps occurring in such structures can be altered while maintaining the total mass of the structure at a constant magic value. Finally, we present a practical application by altering the density distribution of a 2D plate to create low-frequency bandgaps without altering its global mass.

3:35

2pSA10. Engineering bandgaps in TPMS structures by breaking their underlying symmetries. Maria Carrillo-Munoz (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, mxcarrillomunoz@shockers.wichita.edu), Saranchana Keattitorn, and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Triply periodic minimal surface (TPMS) structures are an attractive choice for multifunctional applications—their surface curvature reduces stress concentration locations, they provide excellent stiffness- and strength-to-density scaling, and they are robust thermal and acoustic energy dissipators. Here, we study their elastic wave propagation properties and investigate the possibility of generating wave attenuation bandgaps by breaking their underlying symmetries. We limit our focus to the most used TPMS geometry: the gyroid lattice. We model three dimensional shell-based gyroid structures and calculate their dispersion properties using the finite element approach. Our analysis shows that a significant number of degeneracies exist within the gyroid lattice band structure. We then investigate the possibility of lifting these degeneracies by breaking the underlying symmetric architecture in two distinct ways. First, we disturb the symmetry by altering material properties across the line of symmetry; the effects of elastic and density perturbations are individually studied. Next, we break the symmetry by altering the lattice geometry by directionally stretching the lattice along the x, y, and z directions, individually. Our results show that while both material and geometrical perturbations help create additional directional and polarized bandgaps, both methods result in very different wave propagation and attenuation behaviors.

3:50

2pSA11. Acoustical properties of spinodoid porous 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)

Recently, Kumar *et al*. [1] have proposed a new class of non-periodic cellular structures called spinodoid (spinodal-like) metamaterials. Spinodal decomposition is a diffusion-driven phase transformation mechanism by which a multiphase solution separates into distinct phases, separated by surfaces with smooth, non-intersecting, and non-periodic topologies. Modifying the classical linear Cahn-Hillard equation—which describes this spinodal decomposition process—by biasing the sampled phase field in a controlled manner, Kumar *et al*. have provided a numerically efficient method for designing spinodal-like structures. Here, we use this method to generate porous structures with spinodoid topologies and investigate their acoustical properties. The designed spinodoid structures are fabricated using an additive manufacturing process and experimentally studied using a normal incidence impedance tube setup. We investigate the effect of various topological parameters on the acoustical properties, including the effects of through-thickness gradients with finite transition layers. Our results show that the sound absorption and transmission loss properties of such structures can be conveniently altered by altering their cellular anisotropy, making them a suitable candidate for multifunctional applications that require simultaneous high stiffness, high sound absorption performance. [1] S. Kumar, S. Tan, L., Zheng, and D. M. Kochmann, "Inverse-designed spinodoid metamaterials," npj Comput. Mater. **6**(1), 1–10 (2020).

4:05

2pSA12. Ultra-sparse near-perfect sound absorbers. Jun Ji (Graduate Program in Acoust. College of Eng., Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, jbj5431@psu.edu), Junfei Li, Steve Cummer (Duke Univ., Durham, NC), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Sparseness of an absorber array and its sound absorption performance is a trade-off, which has attracted much research interest. Here, we demonstrate near-perfect absorption (99% absorption) when the spatial period of monopole-dipole resonators is close to one working wavelength (95% of the wavelength). The condition for perfect absorption is to render degenerate monopole-dipole resonators critically coupled. Frequency domain simulations, Eigenfrequency simulations, and coupled mode theory are utilized to reveal the underlying physics and the acoustic performances, showing a good agreement with measurements. The sparse-resonator-based sound absorber could greatly benefit noise control while maintaining air flow and the results of this study could have implications for electromagnetic wave absorbers.

4:20

2pSA13. Acoustic black hole duct with damping material. Kayla Petrover (NSWC, 9500 MacArthur Blvd., Apt. 8B, Bethesda, MD 20817, kayla.petrover@gmail.com)

Acoustic black holes have a tapering thickness profile to create the idealized phenomenon of zero reflection and transmission. The phenomenon entails wave propagation in an ideal medium where the profile tapers according to the power law shortening the wavelength and decreasing the wave speed. Consequently, the wave's travel time tends to infinity so that the wave never reaches the end of the acoustic black hole and cannot be reflected or transmitted. For the application of a duct, thin annular rings of inner radii decrease with the power law to control pressure levels. An infinite number of rings and a final cross sectional area of zero is needed to create the ideal case and is impossible to manufacture. A common method to compensate for imperfections is to use damping material, such as in the experiment [El-Ouahabi *et al*, in International Conference *InterNoise 2015* (2015)], though their results did not show that adding damping to the termination made an improvement. This work aims to find which areas of the acoustic black hole duct are most beneficial to add damping to in order to reduce the reflection and transmission coefficients by measuring the properties of different damping configurations. [Approved for Public Release Distribution Statement A.]

4:35

2pSA14. Isolation of flexural waves in a wide and low frequency range using a phononic beam with acoustic black hole configuration. Seongmin Park (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291 Daehak-ro, Yuseong-gu, Daejeon 34141, Republic of Korea, kaistian13@kaist.ac.kr) and Wonju Jeon (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, Republic of Korea)

We propose a phononic beam to realize flexural wave bandgaps in wide and low frequency regimes. A unit cell of the phononic beam consists of two uniform parts and a tapered part with acoustic black hole configuration sandwiched between uniform parts. We identify the effects of geometrical parameters of the ABH-based phononic beam on band structures and determine those geometrical parameters to obtain a bandgap from 3.6 Hz to 237.9 Hz with a lattice constant of 1/33 wavelength. For experimental realization, the displacement transmission of the phononic beam is measured by impact hammer test, and the result shows that low transmission of –35 dB to i–10 dB could be achieved in the wide bandgap.

4:50

2pSA15. On the acoustical performance of plenum window installed with rigid circular cylinders at low frequency. Wai Kit LAM (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Rm. ZS801, Block Z, Hung Hom, Kowloon, Hong Kong, Hong Kong, 20039053r@connect.polyu.hk), Shiu Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Anton Krynkin, and Alexander Dell (Dept. of Mech. Eng., The Univ. of Sheffield, Sheffield, United Kingdom)

The low-frequency acoustical performance of plenum window with an array of rigid circular cylinders installed is analysed with the Finite Element Method (FEM) in the current study. This study includes the effect of vibrational glass boundaries, cavity resonance, the opening width of the window, and the arrangement of cylinders, on the sound transmission loss (TL) of the plenum window. The relationship between the above factors with TL is also investigated statistically and simple optimization of the geometry has been done based on the results. Results have shown that including the mechanical vibration of glass does not result in a significant difference with previous results, but an average decrease of 1 dB in TL is observed near the eigenfrequencies of the glass boundary. The effect of opening width on TL is not trivial but frequency-dependent due to the cavity resonance formed between the overlapping regions. Lastly, the introduction of rigid cylinders causes a shift of significant TL to lower frequency regardless of the opening width. The installation of them also seems to have destroyed the formation of cavity resonance, which results in much smaller TL.

5:05

2pSA16. Effective design of metamaterial to stop system generated harmonics during nonlinear ultrasonic testings. Pravinkumar R. Ghodake (Mech. Eng., Indian Inst. of Technol., Bombay, IIT Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

System-generated higher harmonics during nonlinear ultrasonic testing mask the actual higher harmonic response due to early-stage damages in solids. To block such unwanted higher harmonics, the periodic arrangement of different materials in phononic crystals shows features like bandgap formation resulting in creating stopbands and passbands. Trial-and-error methods are used by Mostavi *et al*. (2017) to design phononic crystals to stop second unwanted harmonics. They reduced the amplitude of second harmonics by 24 times lower. Here in this study, a more systematic size optimization problem is defined and solved to design metamaterial by considering more realistic boundary conditions and geometric nonlinearity of the phononic materials during finite element simulations using COMSOL. The Nelder-Mead algorithm is used to solve size optimization problems. The optimization problems are solved to block second harmonics due to monochromatic wave and one-way two-wave mixing by considering constraints such as keeping the shape of output pulse and without keeping the shape of output pulse exactly similar to the sinusoidal shape of the input pulse. The amplitude of second harmonics is reduced at maximum up to 50 and 80 times in cases such as with and without maintaining the shape of the output pulses, respectively.

Session 2pSC

Speech Communication: Child Speech Production and Perception (Poster Session)

Margaret Cychosz, Chair
Hearing and Speech Sciences, University of Maryland, 7251 Preinkert Dr., College Park, MD 20742

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

2pSC1. Age and sex effects in emotional prosody processing revealed in infants' mismatch responses but not in preferential looking time. Chieh Kao (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, kaoxx096@umn.edu) and Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Emotional prosody is crucial in early communication, but the developmental trend and the emergence of sex differences in infants' emotional processing remain unclear. This report compared 46 infants' behavioral and neurophysiological responses to four vocal emotions (happy, angry, sad, and neutral). We used a central fixation paradigm to measure infants' listening times to different emotions and a multi-feature oddball paradigm with electroencephalography (EEG) recordings to measure infants' mismatch responses (MMRs) to vocal emotional changes. We modified both paradigms by including non-repeating words within each emotion, enforcing infant listeners to respond to the overarching emotional prosodic category. We only observed age and sex effects in infants' MMRs. Older infants showed more negative MMRs to happy and sad voices than younger infants, but more positive MMRs to angry voices. Female infants showed more positive MMRs at a later time window than male infants across all emotional changes, indicating an emotion-general sex effect. The behavioral study showed longer listening times for happy and sad prosodies across age and sex. The findings indicate that neurophysiological measurement may be more sensitive in capturing age and sex effects in early emotional prosody perception when complex speech materials are used.

2pSC2. Distributional properties of voice onset time in child speech and adult listener judgments. 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)

Voice onset time (VOT) is well-recognized as the primary cue for perception of plosive voicing distinctions although perceptual judgments can also be affected by secondary cues, such as fundamental frequency (f0) and burst amplitude. Further, perceptual judgments may be influenced by distributional properties of the dataset, i.e., how VOT values are distributed along the continuum, and how many productions fall into each category in a listening test. This study assessed the impact of distributional characteristics of naturally produced child speech stimuli to determine their influence on listeners' labeling behavior. The dataset comprised bilabial and alveolar CV words produced by 2–3-year-old English-speaking children. Six exemplars were chosen from six children for four groups: short-lag /b d/ and /p t/, long-lag /b d/ and /p t/. Within-category VOT distributions per POA were bimodal, separated by a 5 ms gap. Listening data were obtained from 20 adults. Preliminary data are consistent with our past experiments: Listeners were highly accurate (i $>$ 90%) at labeling the production as intended by the child when the production matched the VOT category expected for the target word. Measures of f0 and burst amplitude suggest that secondary cues in the speech signal contributed to adults' perception of children's stops.

2pSC3. Disentangling acoustic measures from lexical statistics in child-directed speech. Margaret Cychosz (Hearing & Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, mcychosz@umd.edu), Jan R. Edwards, Nan Bernstein Ratner (Hearing & Speech Sci., Univ. of Maryland, College Park, MD), Catherine Torrington Eaton (School of Health Professions, Univ. of Texas at San Antonio, San Antonio, TX), and Rochelle S. Newman (Hearing & Speech Sci., Univ. of Maryland, College Park, MD)

Acoustically, North American child-directed speech is hyperarticulated: it is slower than adult-directed speech and the vowel space is more expanded. Lexically, child-directed speech is composed of shorter, more frequent words, from denser neighborhoods. The issue is that hyperarticulation, including vowel space expansion, varies according to word structure and statistics: speakers hyperarticulate low-frequency words and phonetically reduce words from denser neighborhoods (Bell *et al*., 2009; Gahl *et al*., 2012). Since the acoustic and lexical parameters of child-directed speech co-vary so tightly, it may be important to disentangle them when modeling the effects of child-directed speech for developmental outcomes. In this longitudinal study, we measured a battery of acoustic-lexical characteristics of child-directed speech from n = 84 caregivers speaking to their infants aged 7, 10, 18, and 24 months. We then fit a series of models to regress out how *each*characteristic independently predicted the children's vocabulary sizes and phonological processing at 24 months. Results confirmed that many characteristics, such as vowel space, coarticulation, and speaking rate, co-varied with the lexical statistics. We nevertheless found direct effects of coarticulation, as well as word frequency, length, and diversity, on the children's development, demonstrating the need to carefully disentangle these co-varying parameters going forward.

2pSC4. Quantifying phonetic variation: A large-scale corpus analysis of coronal segments in English infant-directed speech. Ekaterina A. Khlystova (Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, ekhlystova@ucla.edu), Adam J. Chong (Queen Mary Univ. of London, London, United Kingdom), and Megha Sundara (Linguist., UCLA, Los Angeles, CA)

Phonetic variation poses a challenge for language learners tasked with identifying the abstract sound categories (phonemes) of their target language(s). However, relatively little is known about the full extent of phonetic variability in naturalistic infant-directed speech (IDS), and how much of this variation can be explained by positional allophony. We phonetically annotated ~31 000 tokens of coronal segments (/t/, /d/, /s/, /z/, and /n/) in naturalistic English IDS from the Providence Corpus (Demuth *et al*., 2006)

in order to quantify the degree of phonetic variation present in the everyday speech directed to infants. We found that canonical variants are the most frequent variants for /d/, /s/, /z/, and /n/, but not for /t/. Additionally, every segment had more canonical instances in word-initial compared to word-final position. We are currently comparing the distribution of phonetic variants in specific phonological environments (e.g., tapping environments) in IDS with naturalistic adult-directed speech. We discuss the implications of these results for infants' ability to identify the canonical (phonemic) variant from a set of phones, and uncover when and where positionally governed phonetic variants must surface.

2pSC5. Interactions of culture and depression in infant-directed speech. Favour Bright-Agindotan (Dept. of Commun. Sci. and Disord., Univ. of Utah, 261 E 1st Ave. A, Salt Lake City, UT 84103, favbright@gmail.com) and Gordon Ramsay (Dept. of Pediatrics, Emory Univ., Atlanta, GA, Georgia)

Infant-directed speech (IDS) is a change in voice register adopted by adults when speaking to young children. Maternal depression is known to affect IDS, resulting in a lower fundamental frequency (f0) range that can in turn affect infant response. However, there is little information regarding the interaction of culture and language with depression. The goal of this study was to determine if significant differences in IDS exist between Spanish- and English-speaking mothers and to explore potential interactions with the severity of depression. Whole-day audio recordings were collected in the home from 14 mother-infant dyads, 7 English-speaking and 7 Spanish-speaking, all with infants 6 months of age. 10 examples of adult-directed speech and infant-directed speech were segmented from each recording and the f0 mean and range were measured for each speaker, normalizing infant-directed by adult-directed values to account for speaker variation. The Edinburgh Postnatal Depression Scale (EPDS) was used to determine maternal depression levels. A weak negative correlation was found between depression scores and f0, confirming that depression adversely impacts IDS, but there was no effect or interaction regarding language, suggesting that the mechanism may be common across cultures.

2pSC6. Does F3-F2 distance predict transcriptions of preschoolers' /r/ productions? Elizabeth Ancel (Dept. of Speech Lang. Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, 115 Shevlin Hall, Minneapolis, MN 55455-0279, ancel014@umn.edu), Michael L. Smith (Dept. of Speech Lang. Sci., Univ. of Minnesota, Minneapolis, MN), V. N. Vimal Rao (Educational Psych., Univ. of Minnesota, Minneapolis, MN), and Benjamin Munson (Dept. of Speech Lang. Sci., Univ. of Minnesota, Minneapolis, MN)

Children learning to speak American English are highly variable in early productions of /r/, often producing [w] as an error. One acoustic indicator of /r/ goodness is the difference between the second (F2) and third formant (F3), with smaller F3-F2 differences being associated with a 'better' /r/. This study analyzed the effectiveness of this acoustic measure in characterizing children's productions of the /r/-/w/ contrast on acoustic information extracted automatically from a large data set, where hand measurements are infeasible. LPC formant measures were extracted using Praat from productions of /r/- and /w/-initial words ($n = 19$) by 3.5- to 4.5-year old monolingual preschoolers ($n = 136$). These were rated by naive adult listeners ($n = 142$) on a visual-analog scale of phoneme goodness, and were narrowly transcribed by trained researchers into four categories: [r], [w], and two intermediate categories. Data visualization showed a weak relationship between F3-F2 measures, transcriptions, and ratings. Generalized linear mixed effects regressions found that the F3-F2 difference only modestly predicted transcription categories ($R^2 = 0.39$) and listener ratings ($R^2 = 0.34$). Hence, a description of preschoolers' mastery of the /r/-/w/ contrast based on automatic analyses should include additional measures beyond the F3-F2 measures.

2pSC7. Bottom-up and top-down effects on the perception of children's speech. Benjamin Munson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

This study examined adult's perception of spoken words produced by children and by other adults in audio-only (AO) and audiovisual (AV) contexts, to determine whether talker age affects the magnitude of AV benefit. One-syllable words produced by five 4-6 year old child talkers and five adult talkers were presented to 151 adult listeners at three signal-to-noise ratios. The SNRs were determined through pilot testing to be the ones at which adult and child talkers' productions were recognized equally accurately in the AO condition. The magnitude of AV benefit was statistically equivalent for the adult and child talkers. However, lexical factors affected the perception of children and adults' speech differently. Word frequency had a larger effect on the perception of children's speech, perhaps reflecting listeners' expectation that children would be especially unlikely to use a low frequency word. Lexical factors also affected the nature of listener errors: listeners were less likely to provide a low-frequency word when misperceiving a child than when misperceiving an adult. Listeners were also more likely to guess a word with a different CV structure than the target for children than for adults. Results suggest that listeners apply qualitatively different expectations when listening to adults and children.

2pSC8. Automated vowel space measurement of young children. Marisha Speights Atkins (Commun. Sci. and Disord., Northwestern Univ., Frances Searle Bldg., 2240 Campus Dr., Evanston, IL 60208, marisha.speights@northwestern.edu), Joel MacAuslan (Speech Technologies and Appl. Res. Corp., Lexington, MA), and Maame Agyarko (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Urbana-Champaign, IL)

Vowel formant frequencies are frequently reported as a measure of the articulatory workspace and have been used to study speech development and speech disorders across the lifespan. It is widely acknowledged that the measurement of F1/F2 frequencies for Vowel Space Area (VSA) estimation is a cumbersome task, wrought with challenges and uncertainties. Although there have been advancements in automated approaches, many of these tools continue to be challenged by child speech due to the anatomical differences and dynamic changes of the supraglottal, glottal, and subglottal regions of the airway during the years of speech development. This study examines automated VSA estimation in 4-year-old children using models that have been adapted to manage the higher fundamental frequencies, wider formant bandwidths, and higher amplitudes of subglottal resonances detected in child speech. The speech of six, native American English-speaking, four-year-old children with no history of hearing or speech disorders were included in this investigation. 48 words, including eight American English vowels, were analyzed using traditional manual methods and the SpeechMark MATLAB toolbox automated VSA estimator. The toolbox automates the generation of a convex hull enclosing the child VSA and computation of the area while accounting for the anatomical/acoustic differences that occur during early speech development.

2pSC9. Advancing towards automatic speech recognition of children in naturalistic preschool environment. Satwik Dutta (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, 800 West Campbell Rd., Richardson, TX 75080, satwik.dutta@utdallas.edu), Dwight Irvin (Juniper Gardens Children's Project, Univ. of Kansas, Kansas City, KS), and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

The preschool classroom is a viable space for capturing young children's interactions with teachers. Developing speech/language assessment tools will offer a non-invasive way to quantify child-teacher interactions, allowing teachers to make changes as needed and better support children's speech, language, and cognitive skills. Building Automatic Speech Recognition (ASR) systems for child speech has been challenging, since most studies have been conducted: (1) with older children, (2) in clean and controlled settings, and (3) generally are limited to prompted or read stimuli. To date, limited research has focused on developing ASR systems for spontaneous child speech in preschool settings. For our study, spontaneous child-teacher conversations were captured in preschool classrooms using the LENA

digital audio recorder. A 25-h corpus of child speech were used for experiments using an open-source ASR toolkit. Major challenges in this data for developing ASR models include noisy background environments, age and diversity of children. Our effort focuses on alternative time-delay neural networks (TDNN) for acoustic modelling to improve ASR, as well as trade-offs in language model development and lexicon/vocabulary partitioning. Results from this study across acoustic and language models will be presented using child preschool classroom data. [Work sponsored by NSF CyberLearning Grant #1918032.]

2pSC10. The challenges and uncertainties when hand measuring F1/F2 values in child speech: Towards an automated approach. Maame Agyarko (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Urbana-Champaign, IL, magyar3@illinois.edu) and Marisha Speights Atkins (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Vowels serve a critical role in understanding the acoustic properties of speech. Since they appear early in speech development, they can be used to assess important speech milestones. Prior literature provides the empirical foundation for understanding expected F1/F2 values for typically developing children. Although the traditional measurement of the first and second formant frequencies is a widely accepted practice, its clinical use is limited because of the challenges and uncertainties that are encountered during the process. Automated tools have been developed to address these limitations, however many of these tools have been reported to have inconsistencies when identifying F1/F2 values for child speech. This study explores the use of three semi-automated measurement methods: Wavesurfer Linear Predictive Coding, Praat formant tracking, and a lab-developed approach. The speech of six, native English-speaking, four-year-old children with no history of hearing or speech disorders was analyzed. 48 words, including eight American English vowels, were measured using the three approaches by two researchers, an undergraduate researcher, and an experienced research mentor. The challenges, procedural observations, and accuracy of the methods are discussed. This project was supported by the SURIEA summer research internship program.

2pSC11. A longitudinal study of gender-specific characteristics of children's vowels. Eugene Wong (Univ. of Minnesota, Shevlin Hall, Minneapolis, MN 55455, wong0703@umn.edu) and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

Cisgender adult men and women's vocal tracts differ in length, but the vocal tracts of children assigned male at birth (AMAB) and assigned female at birth (AFAB) do not. Despite this, a growing body of literature has shown that naive listeners robustly rate AMAB and AFAB children's speech to sound different from one another. Recently, Munson *et al*. (2019) found that this was true for children as young as 2.5 years old. However, it is unclear what gender-specific phonetic features exist in children's speech, and how children acquire them. The current study investigates the vowel-space size and shape for the 55 AMAB and 55 AFAB children examined in a longitudinal study by Munson *et al*. at two time-points: 2.5 to 3.5 years old, and 4.5 to 5.5 years old. We aim to assess whether the AMAB and AFAB children's vowel spaces differ in size and shape, and to determine whether these predict the ratings collected by Munson *et al*. As women's vowel spaces are typically more expanded than men's (Bradlow *et al*., 1996), we expect to find larger vowel spaces in AFAB children than AMAB children. Data analysis is ongoing. [Funded by NIH R01 DC002932.]

2pSC12. The Children's English and Spanish Speech Recognition Test (ChEgSS): Normative data. Lori Leibold (Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, lori.leibold@boystown.org), Lauren Calandruccio (Case Western Reserve Univ., Cleveland, OH), Ryan McCreery (Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Jacob Oleson (Univ. of Iowa, Iowa City, IA), Barbara Rodriguez (The Univ. of New Mexico, Albuquerque, NM), and Emily Buss (The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

The Children's English and Spanish Speech Recognition Test (ChEgSS) is a clinical tool for assessing children's English and Spanish word recognition in a noise or two-talker masker. The English and Spanish versions of the test are acoustically, phonetically, and linguistically similar. The goal of this study was to establish threshold norms for Spanish-English bilingual and English monolingual children, and to evaluate effects of age and language proficiency on performance. Participants were 81 Spanish-English bilingual and 89 English monolingual children (4–17 years) with normal hearing. Children completed word recognition testing using an adaptive, 4AFC procedure with a picture-pointing response. All children completed testing in both a noise and a two-talker masker. Bilingual children were tested in English and in Spanish, and monolingual children were tested in English. Language proficiency was assessed using standardized assessments of receptive vocabulary as well as parent report. Both groups of children performed more poorly in two-talker speech than in speech-shaped noise. Preliminary results indicate that age and receptive vocabulary were significant predictors of word recognition performance in both languages and for both maskers. Performance for balanced bilinguals (i.e., similar language proficiency in Spanish and English) was similar for Spanish and English stimuli.

2pSC13. Vowel acoustics in Greek-speaking children with cochlear implants. Areti Okalidou (Education and Social Policy, Univ. of Macedonia, Salonika, Greece, okalidou@uom.edu.gr), Laura Koenig (Haskins Labs, New Haven, NY), Thomas Moran (Commun. Sci. and Disord., Adelphi Univ., Garden City, NY), and George Kyriafinis (Aristotle Univ. Thessaloniki, AHEPA Hospital, Salonika, Greece)

Children who are deaf have been reported to have atypical, often reduced vowel spaces. Improved auditory input via a cochlear implant [CI] may ameliorate such issues, although wide variation among CI users has been observed. Much of the existing literature on speech development post-cochlear implantation assessed speakers of English, which has a rather large vowel inventory. Smaller vowel inventories may be associated with differences in the overall vowel space area as well as in the size of individual vowel areas. This study investigates children acquiring Greek, which has five phonemic vowels /i ɛ a ɔ u/. Seven children with CIs, ages 4–16 years, were age-matched with typically developing children. Vowels were elicited in disyllabic words matched for preceding (bilabial) and following (alveolar) phonetic context: *pita, peta, pata, puse, pote*. Ten productions of each word were elicited using a picture-naming paradigm. Analyses will compare overall vowel space sizes and the degree of variability of individual vowel categories between the two groups. Investigations of children learning phonologies with smaller vowel inventories will lead to a fuller picture of how deafness and subsequent cochlear implantation impact speech communication across languages. [Work supported by Fulbright Greece and iCARE.]

2pSC14. The role of cue weighting and contextual information in speech perception by younger and older adults. Ruoqian Cheng (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, rqcheng@ku.edu) and Allard Jongman (Dept. of Linguist., Univ. of Kansas, Lawrence, KS)

Normal-hearing older listeners are as accurate as younger listeners when perceiving native English words in quiet despite challenges in temporal processing. Older listeners may compensate for the declined use of fine-grained temporal cues by reducing the weight of temporal cues (VOT) and increase the reliance on other acoustic correlates (F0) of the sound contrast. In Experiment 1, younger (age 18–25) and older (age 55–65) normal-hearing listeners participate in an online 2AFC identification task with /d/-/t/ contrast varying in both VOT and F0. We predict that, while both younger and older listeners rely more on VOT than on F0, older listeners, because of their reduced temporal processing abilities, rely on F0 to a larger degree than younger listeners. Temporal processing not only involves local durational cues of the target segments, but also global contextual cues such as speaking rate. In Experiment 2, the same listeners complete another online 2AFC identification task with /dɑ/-/tɑ/ syllables that vary in VOT and vowel duration (short versus long). We predict that older listeners exhibit a smaller shift in the /d/-/t/ category boundary between the long and short vowel durations than younger listeners since older adults are less sensitive to contextual temporal information.

2pSC15. Abstract withdrawn.

2pSC16. Abstract withdrawn.

2pSC17. The effects of face masks on speech-in-speech recognition for children and adults. Mary M. Flaherty (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Briana Arzuaga (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, barzuag2@illinois.edu), and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

While face masks have been used in medical settings for decades, they have only recently become a part of everyday life due to the COVID-19 pandemic. The implications for how face masks impact speech communication in noise are not well understood, particularly in the pediatric population. Given that previous research has shown that speech recognition in noise is more challenging for children than for adults, it is critical that we examine the effects of face masks on children's speech understanding in these contexts. The present study explored the effects of different face masks on school-age children's speech-in-speech recognition. Word recognition was measured in a competing two-talker speech masker. Target words were recorded using a real talker in 5 mask conditions: no mask, surgical, N95, clear plastic, and face shield. Preliminary results indicate that performance was the most negatively impacted when the talker wore a face shield or clear plastic mask. There were no differences between the no-mask condition and either surgical or N95 mask. Acoustic analyses on the target speech were performed to explore causes of these effects. Findings suggest that optimal communication with children may be achieved using surgical or N95 masks as opposed to other mask types.

2pSC18. How do children combine visual speech with other auditory grouping cues? Destinee Halverson (Dept. of Otolaryngology-Head and Neck Surgery, Virginia Merrill Bloedel Hearing Res. Ctr., Univ. of Washington, 1701 NE Columbia Rd., CHDD Clinic Bldg., Seattle, WA 98195, dhalve@uw.edu) and Kaylah Lalonde (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Adults use visual speech to help perceptually group talkers into separate objects and to attend to the target talker. As a result, they benefit more from visual speech in the presence of a two-talker speech masker than in speech-spectrum noise and benefit less from visual speech when other cues are available to support auditory grouping. Recent studies demonstrated that children's visual speech benefit was similar for a noise masker and a two-talker speech masker, suggesting that children may not use visual speech as a grouping cue. The current study further examined whether children use visual speech as a grouping cue by testing whether 40 seven- to nine-year-old children's visual speech benefit decreased in the presence of other cues known to promote auditory grouping in children, namely carrier phrases and perceived spatial separation. Children completed an adaptive closed-set speech recognition task in a two-talker speech masker, in auditory-only and audiovisual conditions, and in the presence or absence of one of the auditory grouping cues. Preliminary analyses suggest that children's visual speech benefit did not decrease when other auditory grouping cues were present. These results provide converging evidence that children in this age range may not use visual speech as a grouping cue.

2pSC19. Predicting children's audiovisual speech recognition thresholds based on unimodal performance. Kaylah Lalonde (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE, kaylah.lalonde@boystown.org)

This study examined the extent to which individual differences in children's audiovisual speech recognition thresholds are explained by their performance in unimodal auditory and unimodal visual conditions. Auditory-only and audiovisual speech recognition thresholds were measured in fifty children between 7 and 9 years of age using an adaptive closed-set speech recognition task in a two-talker speech masker. Children heard color-number combinations and responded by choosing the color-number combination on a touch-screen monitor. Visual-only speech recognition accuracy scores were measured using the same stimuli and closed-set response format. Correlations and linear modeling were used to examine the extent to which visual-only speech recognition accuracy, auditory-only speech recognition thresholds, and age predict audiovisual speech recognition thresholds. Audiovisual speech recognition thresholds were positively correlated with auditory-only speech recognition thresholds and negatively correlated with visual-only accuracy. A linear model explained 38% of variance in audiovisual thresholds and indicated that children with better visual-only accuracy had lower audiovisual thresholds. Effects of auditory-only accuracy and age were non-significant. These results suggest that lipreading ability contributes to individual differences in 7- to 9-year-old children's audiovisual recognition of color-number stimuli in a two-talker speech masker.

Session 2pSP

Signal Processing in Acoustics: General Topics in Signal Processing I

Yongsung Park, Chair
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Contributed Papers

1:00

2pSP1. Autoregressive power law noise for testing of signal processing algorithms. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

Signal detection algorithms should often be tested in the context of noise with different "colors," or exponents, of power law noise (PLN). These noise colors range from "red" (when the frequency exponent is –2), through "white" (0), to "blue" (2). Noise synthesizers that use autoregressive (AR) functions are particularly useful because they can be parameterized to generate the full range of PLN "colors." The degree, or number of terms, in the AR generator for any color of PLN may be infinite. In practice, there are theoretical caps on the degree for some exponents, and diminishing returns for increasing the degree for others. The point of diminishing returns can be identified with two parameters: first, the size of the data vector being generated, and second, the user's tolerence for deviations between the realized power law and the desired exponent. The presentation will describe an open-source Python package, pypowerlawnoise, for generating PLN. This package demonstrates quantitative decision rules for choosing the degree for a discrete AR generator of PLN. A second Python package will be used to demonstrate how artificial PLN can be used to demonstrate the usefulness of a signal processing technique. The whifter package uses frequency-domain Fourier smoothing to isolate whistles and harmonics from background noise. It can also be used for source-filter analysis of speech. Using pypowerlawnoise to create test conditions, I quantify how whifter's efficacy changes across different background noise regimes.

1:15

2pSP2. Delayless polyphase implementation of filters in active noise control systems. Yongjie Zhuang (Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S, Russell St., West Lafayette, IN 47907, zhuang32@purdue.edu) and Yangfan Liu (Mech. Eng., Purdue Univ., West Lafayette, IN)

In real-time implementation of active noise control filters, high sampling rate is sometimes preferred to reduce the delay in analog-to-digital converters or to reduce the risk of aliasing effect. A multi-rate structure is usually implemented with a polyphase structure to maintain a low sampling rate for the control filters to avoid the increase of real-time computational load. However, the use of decimation or interpolation filter will introduce additional delay, which can negatively affect the noise control performance. In the current work, a delayless implementation of polyphase structure is proposed where the original active noise control filter is first constrained to have sufficient attenuation at high frequencies in its design phase and is then decomposed into two multiplicative sub-filters, each of which involves a sufficiently high frequency attenuation. These two decomposed sub-filters can perform both noise control and decimation/interpolation without introducing additional delay in the polyphase structure. The proposed delayless approach can thus reduce the delay of a traditional multi-rate active noise control system and, in principle, can provide better active noise control performance. Moreover, if a single rate system is implemented at a high sampling rate, the proposed polyphase structure in filter implementation can effectively reduce the real-time computational load.

1:30

2pSP3. Variational Bayesian inference for beamforming. Yongsung Park (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, yongsungpark@ucsd.edu), Florian Meyer, and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

A variational Bayesian method for beamforming is presented. The proposed method aims at estimating the time-varying directions of arrivals (DOAs) of source signals. Sequential estimation is performed based on a random process representation of the unknown DOAs. Since calculating the posterior probability density functions (pdfs) of the individual DOA variables is intractable, we utilize variational Bayesian inference to analytically approximate these posterior pdfs. A von Mises representation enables closed-form integration over approximate DOA pdfs. The proposed algorithm iteratively estimates DOAs, source amplitudes, number of sources, and model parameters. Furthermore, it is grid-less, promotes sparse solutions, and provides an uncertainty characterization for estimated DOAs. For an improved time-varying DOA tracking performance, inference results from previous time steps can be used as prior information for sequential processing. We evaluate the proposed method using simulated data and acoustic data from an underwater source localization experiment.

1:45

2pSP4. On underwater acoustic particle velocity channels: Some experimental results. Erjian Zhang (New Jersey Inst. of Technol., Newark, NJ, ez7@njit.edu) and Ali Abdi (New Jersey Inst. of Technol., Newark, NJ)

Studies have shown that acoustic particle velocity channels, in addition to the acoustic pressure channel, can be beneficial for underwater acoustic communication [A. Abdi and H. Guo, "A new compact multichannel receiver for underwater wireless communication networks," *IEEE Trans. Wireless Commun.* **8**, 3326–3329 (2009); A. Abdi, *et al.*, "An overview of underwater acoustic communication via particle velocity channels: Channel modeling and transceiver design," *Proc. Meetings Acoust.* **9**, 070002 (2010)]. In this paper, we use experimentally measured acoustic particle velocity and acoustic pressure channels, using transmitted linearly frequency modulated (LFM) signals, to investigate and compare various signal and noise characteristics of these channels. Examples include signal and noise powers and ratios, delay spreads of the channels, temporal and spatial auto and cross covariance functions for the particle velocity and acoustic pressure noise components, and frequency responses of the acoustic particle velocity and acoustic pressure channels. Understanding these signal and noise characteristics allows a system designer to design and implement reliable and efficient underwater acoustic communication systems that employ multiple acoustic particle velocity channels, together with the acoustic pressure channel. [The work is supported in part by the National Science Foundation (NSF), Grant IIP-1500123.]

2:00–2:15 Break

2:15

2pSP5. Experimental results on acoustic communication channels in pipelines. Erjian Zhang (New Jersey Inst. of Technol., Newark, NJ, ez7@njit.edu) and Ali Abdi (New Jersey Inst. of Technol., Newark, NJ)

Transmission of information and data through pipelines using acoustic waves is a useful and practical approach in remote sensing, telemetry and monitoring applications. The data to be transmitted is first modulated by an actuator on acoustic waves, to travel through the body of the pipeline. At the receiver side, strain sensors and triaxial accelerometers can be used to receive and demodulate the data [E. Zhang and A. Abdi, "Communication rate increase in drill strings of oil and gas wells using multiple actuators," *Sensors* **19** 1337 (2019)]. Understanding the characteristics of the strain and acceleration communication channels in pipelines is of interest, for effective communication and telemetry system design. In this paper, we use the data measured by the strain and triaxial accelerometer sensors, to study several essential communication characteristics of the strain and acceleration channels in pipes, such as frequency responses, possible correlations among different channels, and signal-to-noise ratios of different channels. Results of this study are useful for proper design and optimization of acoustic communication and telemetry systems operating through pipelines. [The work is supported in part by NSF.]

2:30

2pSP6. Some measured frequency domain characteristics of the underwater vector and scalar acoustic field components with applications to communication receivers. Rami Rashid (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., 323 Dr. Martin Luther King Jr. Blvd, Newark, NJ 07102, raa62@njit.edu), Erjian Zhang, and Ali Abdi (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ)

Vector and scalar components of the underwater acoustic field can be used for multichannel signal demodulation [A. Abdi, *et al*., "A new vector sensor receiver for underwater acoustic communication," in Proceedings of the Oceans (2007), pp. 1–10; A. Abdi and H. Guo, "A new compact multichannel receiver for underwater wireless communication networks," *IEEE Trans. Wireless Commun*. **8**, 3326–3329 (2009)]. In frequency domain-based communication methods such as orthogonal frequency division multiplexing (OFDM) using multiple collected signals, understanding the behavior of the channels in frequency domain is important for proper system design. In this paper, we examine the experimental data obtained by transmitting pilot tones for channel estimation, to estimate various channel frequency responses. More specifically, channel frequency responses of the X, Y and Z components of the underwater acoustic vector field are estimated, in addition to the scalar field component channel frequency response. Analysis of the measured frequency responses reveals differences in parameters such as condition numbers and eigenvalues of measured channel matrices. Knowledge of these parameters is needed for designing communication receivers that utilize the vector and scalar components of the acoustic field in frequency domain. [The work is supported in part by NSF, Grant IIP-1500123.]

2:45

2pSP7. Signal detection in time spreading underwater environments using vector and scalar sensors. Rami Rashid (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., 323 Dr. Martin Luther King Jr. Blvd, Newark, NJ 07102, raa62@njit.edu), Erjian Zhang (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ), Zoi-Heleni Michalopoulou (Mathematical Sci. Dept., New Jersey Inst. of Technol., Newark, NJ), and Ali Abdi (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ)

In this paper, we aim to examine the effects of the time-spreading signal distortion caused by multipath propagation in underwater environments, using experimental data. We use the Replica Correlation Integration detector [B. Friedlander and A. Zeira, "Detection of broadband signals in frequency and time dispersive channels," *IEEE Trans. Signal Process.* **44**(7), 1613–1622 (1996)], which is designed to take the multiple echoes of the transmitted signal into consideration. We apply this detector to measured data collected by underwater acoustic sensors. We use two types of sensors. The first sensor is a hydrophone that measures only the scalar component of the acoustic field, i.e., the acoustic pressure. The second sensor is a vector sensor that simultaneously measures the scalar and the three vector components of the acoustic field, i.e., the three acoustic particle velocity x, y, and z components. This sensor can perform as a multi-channel detector. Analysis of the measured data provides insights into the signal detection capabilities of the vector field components in time-spreading underwater environments. [The work is supported in part by NSF, Grant IIP-1500123, and ONR.]

Session 2pUW

Underwater Acoustics: General Topics in Underwater Acoustics: Inversions and Parameter Estimation

Martin Siderius, Chair
Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201

Contributed Papers

1:00

2pUW1. Trans-dimensional geoacoustic inversion on a range-dependent track: Using chirp subbottom survey data as prior information for seabed layering. Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gopu R. Potty, James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, RI), and John A. Goff (Jackson School of Geosciences, The Univ. of Texas at Austin, Austin, TX)

Geoacoustic inversion methods can be broadly divided into two categories. Fixed-dimensional methods are based on the assumption of a known environmental parametrization, including the number of seabed layers. Trans-dimensional (trans-D) methods estimate the environmental parametrization as part of the inverse problem. Although trans-D methods are very powerful, they have barely been applied in range-dependent geoacoustic inversion, and fully range-dependent trans-D inversion is a challenging problem. To mitigate this issue, we propose to use chirp subbottom survey data as prior information within a trans-D inversion method. To do so, the method considers the seabed as an unknown number of homogeneous sediment layers; the number of layers and geoacoustic properties within layers are constant over range, but layer interface depths are range dependent, with the lateral variation informed by the two-way travel-times to reflectors from the chirp survey. The method was successfully applied for single hydrophone geoacoustic inversion using data collected during the 2017 Seabed Characterization Experiment. [Work supported by the Office of Naval Research.]

1:15

2pUW2. Deep ocean sound speed profile estimation using matched autoproduct processing. Jay R. Johnson (Dept. of Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, jayrj@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Sound speed profile (SSP) estimation from acoustic measurements alone in ocean environments is often a highly undetermined problem. Many methods have been developed which match arrival times or acoustic pressure observations to estimate a reduced-order SSP described with shape functions or empirical orthogonal functions (EOFs). These EOFs are commonly derived from multiple *in situ* measurements. This presentation describes an alternative method that matches autoproduct fields to invert for deep ocean SSPs without knowledge of the source waveform. The inverted SSPs are parameterized with EOFs derived from synthetic SSPs extracted from HYCOM (hybrid coordinate ocean model). This method is developed and illustrated using simulated long-range deep-ocean propagation data for frequencies in the 100s of Hz. The technique is also assessed with measured propagation data from the PhilSea10 experiment. [Sponsored by ONR]

1:30

2pUW3. Matched-field geoacoustic inversion using Gaussian Processes for field prediction. Zoi-Heleni Michalopoulou (Dept. of Math. 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)

Matched-field geoacoustic inversion is the process of maximizing a correlation measure between received data at deployed hydrophones and replica fields calculated via an acoustic model for a set of values for environmental parameters, frequently along with source range and depth. The values that produce the maximum correlation form the geoacoustic parameter and source location estimates. Matched-field inversion performance improves as the number of receiving hydrophones increases and the water column is sampled in a dense manner. In order to achieve dense water column sampling when only a few phones are available, we predict the field at a large number of virtual depths within a given aperture using Gaussian Processes. Similarly to classical matched-field inversion, the predicted field, computed after the choice of a suitable kernel and its hyperparameters, is correlated with replicas calculated at the same phone depths. It is demonstrated with both synthetic and real data that the Gaussian Process matched-field inversion approach outperforms conventional processing. The effect is more pronounced as the Signal-to-Noise Ratio attains lower values. [Work supported by ONR.]

1:45

2pUW4. Applications of semi-supervised learning methods in underwater acoustics. Mason C. Acree (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mason7acree@gmail.com), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

Research on supervised deep learning methods have shown potential for ocean acoustic applications. Supervised methods require labeled data samples for training. However, ocean acoustics data often do not contain labels. In experimental situations it is possible to estimate source range and speed labels as inferred by GPS data, but the environmental labels may not be easily defined. Applications of machine learning in ocean acoustics will be greatly enhanced by developing ways to utilize semi-supervised learning. In other applications, semi-supervised learning has been accomplished via contrastive learning or self-attention methods. Our work applies semi-supervised learning by developing a contrastive learning framework for acoustic data. First the model does self-training using an augmentation policy on unlabeled data to initialize representations of key features in the data. Then a small set of cleanly labelled samples are given to a model for supervised learning to teach the desired prediction task. Our work in contrastive learning will be applied to transiting surface ship spectrograms to demonstrate the effectiveness for predicting source labels and seabed type. Preliminary work will be presented to illustrate the advantages of semi-supervised learning. [Work supported by ONR contract N00014-19-C-20001.]

2:00

2pUW5. Infering the characteristics of marine sediments from the acoustic response of a known object. Alexis Bottero (Marine Physical Lab., Scripps Inst. of Oceanogr., UC San Diego, 8820 Shellback Way, La Jolla, CA, abottero@ucsd.edu), Simone Sternini, and William Kuperman (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

The acoustic response of an object depends on the surrounding environment. We show here how to take advantage of a known elastic target located near the seabed to infer characteristics of marine sediment. The approach consists of several steps. First, the structural admittance matrix of the elastic object is determined (numerically or experimentally). Then, the Equivalent Source Method (ESM) coupled with a spectral representation of the Green's functions in stratified domains makes it relatively inexpensive in terms of computational time to predict the object acoustic signature in various environments and experimental configurations. The resulting solver takes into account all multiple scattering between target (buried or not), sea floor and sea surface and is not limited to short distances. After presenting the solution to the forward problem we then show several synthetic inversions for sediment characteristics.

2:15

2pUW6. Estimating sediment interval velocity using a monostatic sonar from a seabed with tilted layers. Cody Henderson (Elec. and Comput. Eng., Portland State Univ., 1825 SW Broadway, Portland, OR 97201, ch27@pdx.edu) and Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR)

Recently, it has been shown that sediment layer interval velocity can be obtained from a multibeam monostatic sonar. In that work, plane-parallel sediment layering was assumed. Since tilted sediment layers are common in the marine environment, the method is expanded to treat them explicitly. The in- and out-of-plane tilted layers affect the travel times in each layer, which complicates interval velocity estimation. This effect is mitigated by applying a dip moveout correction for each sediment layer. Testing is performed using simulated scattering data from rough, tilted, layered seabeds for dip angles up to a few degrees. The interval velocity estimation method is also applied to at-sea data (acquired by the 2–9 kHz SBP29 multibeam sub-bottom profiler) in areas with tilted layers. [Research supported by ONR Coastal Geosciences]

2:30

2pUW7. Long time-series measurements of high-frequency acoustic scattering from sea ice. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu)

Understanding relatively recent rapid changes in Arctic sea ice calls for extensive and continuous observations of the ice cover. Utilizing datasets obtained with 38, 70, and 200 kHz upward-looking transducers in the northeast Chukchi Sea, seasonal and year-to year changes in acoustic scattering were measured. Two complete annual cycles of scattering returns from the winters of 2017–2018 and 2018–2019 were analyzed. The initial effort in this research is constrained to the stable growth phase of sea ice in winter through the early melt onset the following spring. We also restricted our analysis to individual, minimally deformed ice floes to limit the variability caused by large-scale roughness. Results from this two-year analysis will be discussed along with their implications for future measurements and acoustic remote sensing of sea ice properties.

2:45–3:05 Break

3:05

2pUW8. Long-term measurements of high-frequency scattering from the seafloor. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu), Matthew Catoire (US Navy, Norfolk, VA), and Gabriel R. Venegas (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, Durham, NH)

The assumption often used for predicting the performance of many sonar systems is that the acoustic response of the seafloor is solely a function of seabed type and that this response has no temporal dependence. However, seafloor properties controlling acoustic scattered levels and statistics are variable as they are influenced by near-bottom hydrodynamics and/or biological activity. Variability in seafloor acoustic scattering can significantly increase uncertainty in predictions of the performance of object-detection and classification sonar systems as well as seabed-characterization systems. Using several datasets obtained during a series of experiments utilizing 38, 70, and 200 kHz transducers, variability in scattered levels over temporal scales of weeks to months was measured at two sites near Portsmouth, New Hampshire, USA. Results from these experiments will be discussed along with the mechanisms causing variability in scattering strength at the two sites and potential for modeling this temporal variability.

3:20

2pUW9. Acoustic reflection measurements from the bottom of the Challenger Deep. Ying-Tsong Lin (Woods Hole Oceanographic Inst., 86 Water St., Woods Hole, RI 02543, ytlin@whoi.edu), Victor Vescovo, Timothy Macdonald, Kate von Krusenstiern (Caladan Oceanic LLC, Dallas, TX), Robert P. Dziak, Haru Matsumoto, and Christian Meinig (NOAA/Pacific Marine Environ. Lab., Newport, OR)

An underwater sound propagation experiment was conducted in June 2020 at the Challenger Deep of the Mariana Trench on a scientific expedition cruise sponsored by Caladan Oceanic LLC. The primary objective of the experiment was to study ambient sound and acoustic signal propagation in the hadal zone, the deepest region of the ocean. State-of-the-art equipment was utilized to measure the seafloor bathymetry and water column temperature and salinity profiles. A full-ocean-depth hydrophone (FODH) was mounted on a scientific bottom lander deployed right at the bottom of the Challenger Deep (10 935 $\pm$ 6 m deep.) A low-frequency (115 Hz) sound source was towed by the expedition ship at a depth near the sea surface to transmit acoustic signals that were recorded by the FODH. High-fidelity numerical models of sound propagation were employed to aid acoustic data analyses, especially for characterizing the geoacoustic properties of the bottom sediments from the acoustic reflection measurements. [Worked supported by Caladan Oceanic LLC.]

3:35

2pUW10. Correlation between the fluctuations of underwater sound propagation and shelfbreak oceanography. Ying-Tsong Lin (Woods Hole Oceanographic Inst., 86 Water St., Woods Hole, RI 02543, ytlin@whoi.edu), Glen Gawarkiewicz, Andone C. Lavery, Weifeng G. Zhang (Woods Hole Oceanographic Inst., Woods Hole, MA), J Michael Jech (NEFSC, Woods Hole, MA), Martin Siderius (Portland State Univ., Portland, OR), Jason Chaytor (U.S. Geological Survey, Woods Hole Coastal and Marine Sci. Ctr., Woods Hole, MA), William L. Siegmann (Math Sci., RPI, Troy, NY), Emma Reeves Ozanich (Woods Hole Oceanographic Inst., Woods Hole, MA), Brendan J. DeCourcy (Woods Hole Oceanographic Inst., Falmouth, MA), Scott Loranger, Jacob Forsyth, Jennifer Johnson, Arthur E. Newhall, and Frank Bahr (Woods Hole Oceanographic Inst., Woods Hole, MA)

A network of sound sources, hydrophone arrays and physical oceanography and biological survey equipment was established at the southern edge of the New England Shelf in May 2021 to investigate how the oceanic processes at shelfbreak regions affect underwater sound propagation. The ocean processes of particular interest include shelfbreak fronts, shelf water streamers, thermohaline intrusions, and internal waves along with other significant marine geological features and biological factors, such as seafloor slopes, submarine canyons, variable seabed properties, and fish schooling and shoaling. With fixed propagation paths, acoustic fluctuations can be correlated with environmental variations between the network nodes. The temporal scale of the acoustic measurements ranges from minutes to weeks, and the spatial coverage is up to 20–30 km. Adaptive sampling and tracking of acoustic sensitivity "hot spots" was also conducted during the experiment to assist real-time joint ocean acoustics and circulation modeling. This presentation will review the design concept of this ocean acoustic network experiment and provide an overview the preliminary results. [Work supported by the Office of Naval Research.]

3:50

2pUW11. New England shelf break acoustic experiment (NESBA): Assimilation of acoustic pressure observations. Drew Wendeborn (NEAR-LAB ECE Dept. Portland State Univ., PO Box 751, Portland, OR 97207-0751, dwendeborn@pdx.edu), Bill Stevens (Portland State Univ., San Diego, CA), and Martin Siderius (Portland State Univ., Portland, OR)

The NESBA Signals and Noise experiment was conducted in April–May 2021. The goal of the experiment is to assess the potential for sonar prediction effectiveness gains given improved environmental awareness. This talk addresses the NESBA sub-goal of demonstrating that acoustic pressure observations could be assimilated into high-resolution ocean models (i.e., Regional NCOM) resulting in lower analysis and forecast errors for both the ocean environment variables and acoustic predictions. The steps involved in this experimental analysis include: (1) measurement of broadband signals received by drifting vertical arrays; (2) generation of empirical orthogonal function (EOF) representations of the local sound speed environment based on NOAA Regional NCOM forecasts; (3) use of acoustic simulation to generate simulated signals for general elements of the EOF state space; and (4) the use of optimization techniques to determine the EOF coefficients that best match each set of acoustic measurements. Multiple optimizations are to be conducted, each resulting in a synthetic CTD measurement at a specific location and time. These synthetic CTDs will be assimilated into Regional NCOM and comparisons will be made between baseline NCOM forecasts, NCOM forecasts including the synthetic CTD data, and CTD measurement data. [Work supported by the Office of Naval Research.]

4:05

2pUW12. Evaluating empirical and variational mode decomposition capabilities on Risso's and Pacific white-sided dolphin pulsed signals in a sparse and noisy dataset. Kerri D. Seger (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, kerri.seger@appliedoceansciences.com), Yue Liang, Mahdi H. Al-Badrawi, Christopher Foster, and Nicholas J. Kirsch (Univ. of New Hampshire, Durham, NH)

Tracking marine species is critical for conversation efforts, therefore automatic detectors and classifiers play a significant role to facilitate such efforts. Previous work demonstrated the efficacy of the Empirical Mode Decomposition (EMD) classification capabilities in differentiating marine mammal vocalizations. However, EMD failed to separate Risso's and Pacific white-sided dolphin pulsed signals due to their high spectral similarity and limitations of EMD filtering response. The Variational Mode Decomposition (VMD), a more advanced version of EMD, paired with an automated detector, provided a promising tool to tell such dolphins apart with accuracy up to 81.3% even under severe channel conditions. This level of accuracy on a sparse dataset that does not contain whistles is important for automatically classifying pulsed signals from dolphins that do not whistle, live in noisy environments, and/or are recorded in datasets with low duty cycles. Because many datasets collected to date in the Arctic Ocean are sparse due to conserving battery power over long deployments, the VMD method can help add to our ability to track dolphins using just their pulsed signals as they expand northward with warming waters.

4:20

2pUW13. Tidal pressure as a mechanism to explain the steep rise in ambient infrasound below 0.1 Hz at Eleuthera reported by Nichols in 1981. Michael S. McBeth (Naval Information Warfare Ctr. Atlantic, NASA Langley Res. Ctr., 8 North Dryden St., M.S. 473, Hampton, VA 23681, michael.s.mcbeth@navy.mil) and Umesh A. Korde (Johns Hopkins Environ. Health & Eng., Johns Hopkins Univ., Baltimore, MD)

R. H. Nichols reported measurements of ambient ocean noise made off Eleuthera Island [*J. Acoust. Soc. Am.* **69**, 974–981 (1981)]. Nichols was unable to identify a mechanism to explain the steep rise of 20 dB per octave from 0.02–0.1 Hz in spectrum levels obtained with hydrophones at depths of 300 and 1,200 m. Here we show that a tidal pressure signal modeled as a ramp pulse waveform through an octave bandpass filter results in spectrum levels increasing about 12 dB per octave with decreasing frequency below 0.1 Hz. If the infrasound noise levels continue to follow the 6 dB per octave increase with decreasing frequency down to 0.02 Hz as they do from 0.1–1 Hz, the tidal pressure signal will dominate the infrasound. After a crossover point the infrasound noise signal becomes dominant with increasing frequency. This proposed tidal pressure mechanism could be tested by correlation of observed spectrum levels with the tides just as measurements of surface wind speeds have been shown to correlate with increased infrasound spectrum levels.

4:35

2pUW14. Peak-time sensitivity kernels for noise cross-correlation patterns. Bruce Cornuelle (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA) and Emmanuel Skarsoulis (Inst. of Appl. and Computational Mathematics, Foundation for Res. and Technol.—Hellas, N. Plastira 100, Heraklion, Crete GR-70013, Greece, eskars@iacm.forth.gr)

Sensitivity kernels for travel-time observables defined on the noise cross-correlation pattern – the envelope of the cross-correlation function of the underwater noise field between two locations – with respect to changes in the sound-speed field in refractive underwater environments will be addressed. A wave-theoretic scheme allowing for finite-frequency calculations in 2 and 3 dimensions combined with the Born approximation for perturbations of the Green's function and the peak-arrival approach is used to obtain peak-time sensitivity kernels (PTSKs) with respect to environmental (sound-speed) changes. PTSK results for different propagation conditions and noise-source distributions, ranging from spatial distributions of uncorrelated noise sources to point sources such as individual ships, will be presented. The PTSKs can be a useful tool for obtaining environmental information from noise measurements. [Work supported by the Office of Naval Research.]

4:50

2pUW15. Quantifying acoustic backscatter of seafloor using multibeam echosounder. Henry Manik (IPB Univ., Kampus IPB Dramaga, Bogor 16680, Indonesia, henrymanik@apps.ipb.ac.id)

Multibeam echosounder is one of the most important technology for carrying out hydrographic surveys, benthic habitat mapping, marine fisheries, and seafloor surveys. The advantage of multibeam echosounder were produced simultaneously bathymetric and acoustic backscatter. The purpose of this research was to process and to analyze multibeam echosounder data for mapping the seafloor and quantifying acoustic backscatter. Bathymetric data was processed used the Combined Uncertainty and Bathymetry Estimator (CUBE) method. The analysis of seabed type used Angular Response Analysis (ARA) and Sediment Analysis Tool (SAT). Seawater depth in the survey area was measured from 11.2 m to 45.3 m. Analysis of seafloor classification consists of clay, silt, sand, and rock bottom. The highest acoustic backscatter was rock bottom and the lowest was clay bottom.

5:05

2pUW16. Simultaneous velocity and concentration profiling of nuclear waste suspensions in pipe flow, using ultrasonic Doppler and backscatter analysis. Serish T. Hussain (School of Chemical and Process Eng., Univ. of Leeds, 211 Clarendon Rd., Leeds, West Yorkshire ls29jt, United Kingdom, pm14sth@leeds.ac.uk), Martyn Barnes (Sellafield Ltd., Cumbria, United Kingdom), Jeffrey Peakall (School of Earth and Environment, Univ. of Leeds, Leeds, United Kingdom), and Timothy Hunter (School of Chemical and Process Eng., Univ. of Leeds, Leeds, United Kingdom)

Complex magnox ponds at Sellafield retain hydroxide-based sludge with unknown physicochemical properties. The sludge is a corroded form of magnesium-based fuel cladding used in first generation magnox reactors. Long term storage and open-air aqueous conditions are major contributors to corrosion. Waste needs to be transported to interim storage using engineered pipelines. Remote *in situ* characterisation of sludge was achieved using a non-invasive ultrasonic velocity profiler (UVP). The UVP uses raw velocity and echo amplitude to extract velocity and acoustic profiles. Profiles are used to understand effect of particle size and concentration on attenuation. Spherical glass particles with two size distributions of known acoustic behaviour were analysed to establish acoustic profiles. 4MHz frequency transducers were mounted in-line on engineered pipelines, 90 deg and 135 deg to flow. Data were extracted from *in situ* transducers for validation and non-contact probes were utilised to prove data could be collected safely without exposure. Flow rate was varied to ensure full developed flow before analysis. Profiles extracted from concentration arrays were analysed to understand particle segregation in horizontal pipe arrangements and attenuation. Remote in-line characterisation allows safe analysis, whilst in-line characterisation saves time and money. Moving forward, complex non-defined nuclear simulants will be analysed to investigate the UVP's limitations.

Exhibit

An instrument and equipment exhibition will be located in the Ballroom near the registration area and meeting rooms.

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, 29 November, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary beverage.

Tuesday, 30 November, 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: 4 December, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. break serving coffee.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:45 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 are as follows:

Engineering Acoustics (4:45 p.m.)	402
Acoustical Oceanography	Quinault
Animal Bioacoustics	302/303
Architectural Acoustics	Elwha A
Physical Acoustics	402/403
Psychological and Physiological Acoustics	Elwha B
Signal Processing in Acoustics	502
Structural Acoustics and Vibration	301/304

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Elwha A

Committees meeting on Thursday are as follows:

Computational Acoustics	306/307
Musical Acoustics	302
Noise	Elwha A
Speech Communication	Quinault
Underwater Acoustics	Elwha B

Session 3aAA

Architectural Acoustics and ASA Committee on Standards: Restaurant Acoustics

Wade R. Bray, Cochair
HEAD Acoustics, Inc., 6964 Kensington Road, Brighton, MI 48116

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30a, Herzogenrath 52134, Germany

Chair's Introduction—9:00

Invited Papers

9:05

3aAA1. The soundscape of restaurants part 4 analysis and practical design directions. Keely M. Siebein (Siebein Acoust., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com), Gary W. Siebein, Hyun Paek, Marylin Roa, Jennifer Miller, and Matthew Vetterick (Siebein Assoc., Inc., Gainesville, FL)

This paper addresses important sonic issues in restaurants and how they can be addressed from an architectural acoustic perspective. Restaurants have complex sound fields composed of many sound sources, listeners and paths. Some of these sound paths are useful, and others are not. Sound paths from one person speaking to another across a table are typically useful (Near), as diners want to hear each other and converse with those at their table and the waiter. Direct and reflected sound paths that travel from a person speaking to non-intended listeners across the room (Far) are not useful. Speech Transmission Indices were measured for "near" and "far" listeners in restaurants. This idea is illustrated with recent case studies. They are also examined with the C50 metric. It is desirable to have higher near STI and lower far STI, when the background and residual noise levels in the space are considered. This study presents the results along with practical design recommendations on how to improve the STI of near sound paths and reduce the STI of far sound paths, such as spatial separation, part height dividers, strategically placed absorbent material, groupings of tables, subdivision of room, and background music among other strategies.

9:30

3aAA2. Insight on the acoustics in restaurants from combined analyses of logged sound levels, occupancy, and customer responses. Blake R. Krapfl (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, University of Nebraska - Lincoln, Omaha, NE 68182-0816, bkrapfl2@huskers.unl.edu), Joshua Palakapilly (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), Gregory Scott (SoundPrint, New York, NY), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

This presentation reports on analyses of acoustical measurements gathered in restaurants to study relationships between sound levels, occupancy, architectural design, and associated customer responses. Sound level meters and dosimeters logged the sound levels in occupied restaurants, and the collected octave band data have been examined. The results of using machine learning algorithms to detect occupancy from thermal camera images deployed in the same time periods are presented. Finally, restaurant patrons were encouraged to provide feedback using the SoundPrint app on the same evenings as level and occupancy data were collected, and connections between that subjective data and the objective metrics are discussed.

9:55–10:10 Break

10:10

3aAA3. The successful application of acoustic lighting in restaurants. Zackery Belanger (Arcgeometer LC, 1111 Bellevue St., Ste. 360, Detroit, MI 48207, zb@arcgeometer.com)

Contemporary restaurants suffer from acoustic deficiencies for myriad reasons, including minimal visual design, a prevalence of cleanable and durable surfaces, low room volume, high patron densities, the perceived cost of acoustic absorption, the late realization that a problem exists (after opening), lack of confidence in the effectiveness of proposed solutions, and the potential negative impact on visual aesthetics. Acoustic lighting is a new sector of the acoustic product market that shows potential in alleviating some of these problems. In this study, the basics of acoustic lighting history, design, and implementation are introduced, and the benefits and drawbacks of acoustic lighting are covered. Two case studies will be presented with measured data—a restaurant and a tap room—which are both located in Seattle. Finally, the potential future of acoustic lighting will be anticipated as this new approach solidifies in acoustic architecture.

10:35

3aAA4. Acoustics in the culture of context. Ronald A. Altoon (Altoon Strategic, LLC, 5805 White Oak Ave., PO Box 16249, Encino, CA 91316, altoon@altoon.com)

Not being an acoustician, I have limited knowledge of such expertise. I trust advice of consultants in your science to inform consideration of acoustical issues. Having designed projects in 46 countries I understand context is often represented incompletely in serving community needs. I will present more thorough representations of contextual forces informing my design, infusing regional voices in built work. Cultural influencers optimize design of sedate individual Korean restaurant rooms in contrast to strategies deployed in American restaurants where noise is utilized to maximize sales. I would intend to provoke this discussion. Context is generally defined as the physical setting affecting of buildings. Historically, identifiable nationalities or tribes have evolved organically over time, producing significantly unique DNA of each. These differences are most apparent in crafting buildings and conceiving art, music, and cuisine where acoustic social protocols are voiced. NATURAL FORCES affecting human comfort GEOGRAPHIC FORCES influencing integration with nature HISTORIC FORCES defining legacy of place URBAN FORCES defining constraints and opportunities PHYSICAL FORCES influencing disposition of a program on a specific site HUMAN FORCES defining community MARKET FORCES creating possibility DIGITAL FORCES redefining engagement ENGINEERING FORCES affecting psychological balance

11:00

3aAA5. A general overview: The appearance and evolution of spatial electroacoustic systems for ambient and other applications in the restaurant environment. Wade R. Bray (HEAD Acoust., Inc., 6964 Kensington Rd., Brighton, MI 48116, wbray@headacoustics.com)

An important new factor has recently entered the restaurant acoustics environment: spatial electroacoustic systems, deployed in coordination with physical acoustic materials and procedures and architectural/interior design options. The purposes are on the one hand to eliminate noise and Lombard-speech-runaway problems while giving diners the impression of a completely natural human-driven restaurant ambience, and on the other hand to introduce additional dimensions of acoustic and visual architectural creativity while maintaining individual-table speech privacy and communication comfort. One developer has entered this field, Meyer Sound of Berkeley, California. The Constellation restaurant electroacoustic concept is discussed here in very general, orientational terms to set the stage for future detailed presentations by the practitioners. Current implementations have evolved from both the performing-arts world (Jaffe Acoustics / Jaffe Holden Acoustics ERES, Lexicon LARES and the more recent Meyer Sound Constellation performing arts electroacoustic systems) and the business world (concealed full-duplex teleconferencing systems and subtle yet effective “speech lift” and open-plan masking systems). The new restaurant systems can be used not only for controlling and optimizing ambient backgrounds, but also for differentiating the apparent acoustics of different areas and for permitting, even in a zoned manner, new opportunities in music performance.

11:25–11:55

Session 3aAB

Animal Bioacoustics: Animal Bioacoustics Poster Session (Poster Session)

Benjamin N. Taft, Chair
Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405

All posters will be on display from 8:20 a.m. to 10:20 a.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:20 a.m. to 9:20 a.m. and authors of even-numbered papers will be at their posters from 9:20 a.m. to 10:20 a.m.

Contributed Papers

3aAB1. The whale song translation project—An experiment to assess humpback whale response to voice-selected visual feedback cues. Howard S. Pines (Retired, Cisco Systems, Wireless Network Business Unit, Working From Home, El Cerrito, CA 94530, howardpines@ieee.org)

To better understand the behavioral and communication capabilities of Megaptera Novaeangliae, the findings of a recent whale song study suggest an intriguing experiment to assess humpback whale response to acoustically selected visual-feedback cues. The analysis of high-complexity, frequency-modulated song units indicates a Shannon-Hartley-compliant sub-unit architecture similar to human vowel generation. Like constant-pitch English language vowels, which are differentiated by their two most energetic peak resonance frequencies, humpbacks also exhibit precision vocal control of the production of a variety of sub-units of distinct and differentiable harmonic frequency combinations. Humans navigate mobile phone and tablet-PC informational, gaming, and adaptive learning apps using visual feedback from tactile-selected touchscreen icons and hyperlinks. In lieu of tactile manipulation, an alternative approach to touchscreen control is vocal selection of icons and links corresponding to the generation of specific vowel resonance frequencies and sub-unit harmonic frequencies. A software prototype of a voice-controlled "touchscreen" gaming experiment demonstrates how humans and humpback whales could conceivably interact or how humpbacks could engage in informational transactions. The prototype also incorporates video "training" examples designed to guide subjects in the voiced selection of visual symbols assigned to the sub-regions of a big-screen display.

3aAB2. Frequency changes in humpback whale (*Megaptera novaeangliae*) song units in response to small boat noise Astarte Brown Advised by Kerri D. Seger. Astarte Brown (Ecology & Evolutionary Biology, Univ. of California, Santa Cruz, 1156 High St., Santa Cruz, CA 95064, astarteb@ucsc.edu), Kerri D. Seger (Appl. Ocean Sci., Fairfax Station, VA), Maria Paula Rey-Baquero, and L. V. Huertas-Amaya (Pontificia Universidad Javeriana, Bogotá, Colombia)

Anthropogenic noise compromises the effectiveness of communication in marine animals. This can interfere with natural auditory signal processing (for example, "masking"). Some animals have been shown to change the frequencies of their vocalizations to avoid masking. Humpback whales produce songs with low fundamental frequencies which often overlap with the low frequency content of vessel noise. In the Gulf of Tribugá, a part of the breeding grounds for humpback whale Stock G, a marine port construction project has been proposed. If constructed, underwater noise would significantly increase and potentially cause acoustic behavioral changes, negatively impacting communication efforts among the whales. This study was supported by SURIEA, and tested whether humpback whales in the Gulf of Tribugá changed the frequency content of their songs during and after boat noise as compared to before it was present. Out of 38 comparisons, song frequencies significantly changed 9 times. Additionally, out of 20 comparisons, bandwidth narrowed 11 times, broadened 9 times, shifted higher 6 times, and shifted lower 2 times. These results give greater insight into how humpback whales are able to adjust their song structure in response to small boat noise and indicate that adaptations to some anthropogenic noise sources are possible.

3aAB3. The social signaling behavior of humpback whales on the Hawaiian breeding grounds investigated using acoustic tags. Jessica Carvalho (Univ. of Algarve, 12 Charlotte's Way, Danbury, CT 06811, jcarvalho.dhs@gmail.com), Marc Lammers (Hawaiian Islands Humpback Whale National Marine Sanctuary, Kihei, HI), Rita Castilho (Ctr. of Marine Sci., Univ. of Algarve, Faro, Portugal), and Adam Pack (Departments of Psych. and Biology, Univ. of Hawai'i at Hilo, Hilo, HI)

Humpback whales (*Megaptera novaeangliae*) are one of the most social of all baleen whale species. Despite extensive research into humpback whale song, gaps remain in the understanding of humpback whale communication. These gaps are particularly evident with respect to humpback whale non-song social vocalizations. This study expands upon the current understanding of non-song social call use and function by comparing call prevalence, features, and dynamics across humpback whale groups of three different compositions: dyads, escorted mother-calf pairs, and competition groups. Recordings were collected from 12 deployments of Acousonde^{TM} acoustic and data logging tags on whales off Maui, Hawaii during the winter breeding seasons of 2019-2021. Individual social calls were selected based on visual and aural inspection of spectrograms using Raven Pro 1.6 software, with a total of 518 calls chosen throughout the 60.4 h of acoustic recordings. Of these calls, 49.2% occurred in competition groups, 33.6% in escorted mother-calf pairs, and 17.2% in dyads. Furthermore, we conducted statistical analyses to characterize and classify the social calls recorded and assess the variability of the type, quantity, and frequency of social calls used in groups of differing compositions. This study provides new insights into humpback whale vocal communication behavior in the Hawaiian Islands breeding grounds, particularly with respect to three principal social groups, whose non-song vocal communications have been understudied.

3aAB4. Evaluating ultrasonic vocalizations as predictors of social behavior during group interaction. Christyana Kawar (Univ. of Delaware, 105 The Green Rm. 108, Newark, DE 19716, ckawar@udel.edu), Rachel S. Clein, Megan R. Warren, and Joshua P. Neunuebel (Univ. of Delaware, Newark, DE)

During social interaction, animals integrate sensory information—including auditory cues—to influence future behavioral decisions. For example, animals respond to distinct vocalizations with unique actions (Seyfarth *et al.*, 1980). Mice produce ultrasonic vocalizations (USVs) associated with discrete behaviors (Sangiamo *et al.*, 2020); however, it is unclear if USVs inform subsequent actions. To investigate the predictive power of USVs on behavior, while mice freely interacted ($n = 11$ groups, 2 males and 2 females per group), we continuously recorded video and audio data using

a sound source localization system with an 8-channel microphone array to accurately assign vocalizations to individuals. Using machine learning-based approaches to categorize vocalizations based on acoustic properties and extract behaviors in which two mice play unique roles (actor or recipient), we found a significant interaction between particular social behaviors and preceding vocalizations. A permutation test indicated that, depending on the behavioral role of the vocalizer, specific vocalizations preceded dominant (courtship or aggressive) or submissive (avoidant) behaviors at above-chance levels. Additionally, though classifiers trained to distinguish behavioral role based on preceding vocal emission failed for some behaviors, they were successful for particular dominant behaviors. These results suggest a potentially predictive relationship between acoustic communication and subsequent behaviors.

3aAB5. Finding each other: *Achroia grisella*'s oddly simple directional ear. Lara Díaz-García (Ctr. for Ultrasonic Eng., Univ. of Strathclyde, 99 George St., Glasgow G1 1RD, United Kingdom, lara.diaz-garcia@strath.ac.uk), Andrew Reid, Joseph Jackson, and James Windmill (Ctr. for Ultrasonic Eng., Univ. of Strathclyde, Glasgow, United Kingdom)

Achroia grisella is a pyraloid moth capable of directional hearing. It is generally accepted that hearing arose in moths to avoid bats, their main predators, and, in this particular case, was later repurposed into a rudimentary mating tool. The male emits ultrasonic signals of a wavelength several times larger than the distance between their tympana during its mating process. Notwithstanding, the female moths are capable of somewhat efficiently reaching the males after some zigzagging. Another remarkable feature is the simple structure of the ear, consisting of a cluster of just four receptor cells attached directly to the tympanic membrane. Moths with only one functioning ear have been seen to reach the males still, which implies the tympanum structure itself must confer the moth with monoaural directionality. A model is developed in COMSOL to explain the behavior of the moth ear. The complexity of the model is increased by improving the resemblance to the natural one, starting from a circular plate and progressing to, eventually, a damped elliptical plate with two different thicknesses and an attached point mass.

3aAB6. Physics of flatulence. David S. Ancalle (Mech. Eng., Georgia Insi. of Technol., 801 Ferst Dr., Atlanta, GA 30332, ancalle@gatech.edu), Charles L. Finn, Nelson Jiang, Seth A. Stewart, and David L. Hu (Mech. Eng., Georgia Insitute of Technol., Atlanta, GA)

Flatulence consists of intestinal gas expelled through the anus. While flatulence is a common part of daily life, sudden changes can be important disease indicators for physicians. In this combined experimental and theoretical study, we rationalize the duration and acoustic frequency of flatulence among mammals, from rats to elephants. We perform high-speed videography and acoustic recordings of guinea pigs, humans, and whoopee cushions. The acoustic frequency is generated by the vibration of the skin of the anus, similar to the buzzing of lips in a trumpet's mouthpiece. Thus, larger animals generate lower acoustic frequencies due to the lower resonant frequencies of the thicker and larger anal skin.

3aAB7. Novel software for deep-learning based acoustic data analysis. Fabio Frazao (Comput. Sci., Dalhousie Univ., 6050 University Ave., Halifax, NS B3H 1W5, Canada, fsfrazao@dal.ca), Steven Bergner (Computing Sci., Simon Fraser Univ., Burnaby, BC, Canada), Mike Dowd (Dalhousie Univ., Halifax, NS, Canada), Ruth Joy (School of Environ. Sci., Simon Fraser Univ., Burnaby, BC, Canada), Oliver S. Kirsebom (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Paul Nguyen Hong Duc (School of Mathematics and Statistics, Carleton Univ., Ottawa, ON, Canada), Bruno Padovese (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Marine Randon (School of Environ. Sci., Simon Fraser Univ., Burnaby, BC, Canada), Amalis Riera Vuibert (Comput. Sci., Dalhousie Univ., Victoria, BC, Canada), Sadman Sakib (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Scott Veirs, Val Veirs (Beam Reach SPC, Seattle, WA), Jennifer Wladichuk (Univ. of Victoria/JASCO Appl. Sci., Victoria, BC, Canada), and Harald Yurk (Ecosystem Sci. Div., Fisheries and Oceans Canada, Vancouver, BC, Canada)

In recent years, deep neural networks have been successfully applied to solve a range of detection and classification tasks in underwater acoustics, outperforming existing methods. However, deep learning models are "data hungry" requiring large amounts of accurately labelled acoustic samples to train. Moreover, a certain amount of "fine tuning" is often required to achieve satisfactory performance in a new acoustic environment. Thus, the development of deep learning models depends on the input of expert human analysts, both for building the initial training set and for adjusting the model's performance. However, open-source software to facilitate this collaboration between machine learning developers and acousticians is currently lacking. To address this need, our team is building a web-based application for collaboratively annotating sound samples and validating model predictions. The user interface is designed to be familiar to acousticians, while machine learning developers have access to a dashboard allowing them to efficiently leverage the acousticians' expert knowledge. In this contribution, an overview of the application will be given and its functionalities will be demonstrated through its application to the HALLO (Humans and ALgorithms Listening for Orcas) project. Future developments will also be described, highlighting complementary applications under development such as a model adaptation tool.

3aAB8. Ketos—A deep learning package for creating acoustic detectors and classifiers. Oliver S. Kirsebom (Comput. Sci., Dalhousie Univ., 6050 University Ave., Halifax, NS B3H 4R2, Canada, oliver.kirsebom@dal.ca), Fabio Frazao, Bruno Padovese, Sadman Sakib, and Stan Matwin (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada)

Passive acoustic monitoring is a useful technique for studying aquatic animals, but sustained observing systems require automated algorithms for detecting and classifying sounds of interest. In the last decade, deep neural networks have proven highly successful at solving a wide range of pattern recognition tasks, and recently, we have seen the first promising applications of deep neural networks to detection and classification tasks in marine bioacoustics. Deep neural networks exhibit a high degree of versatility and adaptability: the same network architecture can be trained to accomplish a multitude of tasks by feeding appropriate training data to the network without the need to modify the underlying algorithm. Thus, neural networks have the potential to transform our approach to developing acoustic detection and classification programs, enabling researchers in the field to develop or re-purpose their own programs. MERIDIAN is contributing towards this goal through the development of the open-source Python package Ketos, which provides a high-level programming interface for building training datasets and developing neural network based detectors and classifiers for analyzing underwater acoustics data. In this contribution, an overview of the software package will be given and its functionalities will be demonstrated through case studies.

3aAB9. Classification of simulated two-highlight echoes based on time separation in the bottlenose dolphin (*Tursiops truncatus*). Alyssa W. Accomando (Biologic and Bioacoustic Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, alyssa.accomando@nmmpfoundation.org), Jason Mulsow, Dorian Houser (Biologic and Bioacoustic Res., National Marine Mammal Foundation, San Diego, CA), and James J. Finneran (NIWC Pacific Code 56710, U.S. Navy Marine Mammal Program, San Diego, CA)

Previous studies have suggested that dolphins perceive echo spectral features on both large (macrospectrum) and small (microspectrum) scales. The current study was based on a finding that these percepts are—to some degree—dependent on the dolphin's $\sim$250-μs "critical interval" of echolocation. Two dolphins were trained to provide a behavioral response upon detecting a passively presented, simulated two-highlight echo. This "target" had consistent spectral features according to a 120-μs inter-highlight interval (IHI). The target was presented among distractor echoes with various macrospectra and IHIs from 50 to 500 μs (i.e., various microspectra). Following acquisition of this discrimination task, probe stimuli with the macrostructure of the target but IHIs from 50 to 500 μs were presented. Both dolphins responded more frequently to probes with IHIs of 80 to 200 μs during initial presentations. Response strategies diverged, however, with increasing probe presentations: one dolphin progressively responded to a narrower range of probe IHIs, while the second increased response rates for probes with IHIs $>$250 μs. These results support previous conclusions that perception of macrostructure for complex echoes is non-constant as echo IHI decreases below approximately 80 μs, but results near 250 μs (i.e., the critical interval upper limit) were more ambiguous. [Work funded by ONR.]

3aAB10. Just around the corner: Acoustic mirroring enables echolocating bats to detect occluded objects. Kathryne Allen (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N Charles St., Ames Hall, Ste. 135, Baltimore, MD 21218, kallen59@jhu.edu), Miao Fu, Christopher Farid, and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

The ability to detect salient targets and segregate them from background clutter is a key function of sensory systems. The echolocating bat *Eptesicus fuscus* is an ideal model for investigating the influence of clutter on target detection, as they must navigate dense foliage at high speed, while tracking moving prey. This difficult task requires bats to reliably detect and track objects in the environment despite changes in distance, angle of view, and interference from background echoes, which obscure targets from sonar "view" and creates barriers bats must navigate around. Counterintuitively, recent studies reveal that environmental clutter echoes may improve performance under some conditions. Neurophysiological recordings in the bat inferior colliculus indicate that acoustic scattering caused by foliage can improve target discrimination when the separation between target and clutter is $\sim$20 cm. These findings suggest that acoustic reflections from clutter may provide additional "glimpses" of targets and improve target discrimination. Here, we test the ability of echolocating bats to make use of acoustic mirrors from clutter objects for target detection using a yes/no behavioral assay inspired by Non-Line of Sight Imaging experiments. Much like technology employed by machine imaging, robotics, and self-driving automobiles, we test the potential for bats to "see around corners" using only echoes reflected from objects in the environment to locate a target hidden behind an occluder.

WEDNESDAY MORNING, 1 DECEMBER 2021 401 (L)/404 (O), 9:00 A.M. TO 11:00 A.M.

Session 3aBAa

Biomedical Acoustics and Physical Acoustics: Biomedical Acoustics Modeling Workshop

Vera A. Khokhlova, Cochair
Physics Faculty, University of Washington/Moscow State University, Moscow 119991, Russian Federation

Petr V. Yuldashev, Cochair
Lomonosov Moscow State University, Moscow, Leninskie Gory, Moscow 119991, Russian Federation

Invited Paper

9:00

3aBAa1. Capabilities and features of the "HIFU beam" nonlinear modeling tool for axially symmetric acoustic fields generated by focused therapeutic transducers in layered media. Petr V. Yuldashev, Maria M. Karzova (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation), Wayne Kreider (CIMU / APL, Univ. of Washington, Seattle, WA), Pavel Rosnitskiy, Oleg A. Sapozhnikov (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russia), and Vera A. Khokhlova (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russia, va.khokhlova@gmail.com)

The goal of this introductory talk is to present an overview of the software termed "HIFU beam" that has been designed for simulating HIFU fields generated by axially symmetric single-element transducers and annular arrays in flat-layered media that mimic biological tissues. Numerical models include the Khokhlov–Zabolotskaya–Kuznetsov (KZK) and one-way Westervelt equations with thermoviscous absorption as well as power-law absorption to capture frequency-dependent effects. The numerical algorithms include combinations of time- and frequency-domain finite-difference schemes to simulate formation of high-amplitude shocks at the focus. The software comprises a MATLAB toolbox combined with a user-friendly graphical interface and binary executable compiled from FORTRAN source code; it is available for Windows operating systems only. Simulation examples representing typical cases of ultrasound transducers, focusing conditions, and propagation in layered media are presented to illustrate software features for High Intensity Focused Ultrasound (HIFU) applications. The software tool "HIFU beam" is freely available and can be downloaded, along with a detailed help file, from the website http://limu.msu.ru. The software can help to predict nonlinear HIFU pressures generated by existing transducers and help in the design of transducers for achieving specified acoustic outputs. [Work supported by RSF 20-12-00145, NIH R01EB7643, and R01EB25187.]

Session 3aBAb

Biomedical Acoustics: Biomedical Acoustics in Ophthalmology I

Jonathan Mamou, Cochair
Riverside Research, 156 William St., 9th Floor, New York, NY 10038

Xiaoming Zhang, Cochair
Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Invited Papers

8:40

3aBAb1. Acoustic micro-tapping optical coherence elastography (AuT-OCE) to quantify biomechanical changes following corneal collagen crosslinking: *Ex vivo* study. Mitchell Kirby, Ivan Pelivanov, John Pitre (Bioengineering, Univ. of Washington, Seattle, WA), Ryan Wallace (Ophthalmology and Bioengineering, Univ. of Washington, Seattle, WA), Matthew O'Donnell (Bioengineering, Univ. of Washington, Seattle, WA), Ruikang K. Wang (Ophthalmology and Bioengineering, Univ. of Washington, Seattle, WA), and Tueng T. Shen (Ophthalmology and Bioengineering, Univ. of Washington, 908 Jefferson St., Seattle, WA 98104, ttshen@uw.edu)

In degenerative corneal diseases, such as keratoconus, local deterioration in mechanical properties results in corneal deformation and vision loss. Collagen cross-linking (CXL) has been used to delay progression of keratoconus with some success in slowing the disease, both long-term and immediate outcomes remain unpredictable for each patient. There is a clear unmet clinical need to develop a personalized biomechanical model based on quantitative maps of corneal mechanical moduli to quantify both biomechanical properties and predict final corneal shape. In this study, we use a novel non-contact, non-invasive AuT-OCE method to quantify corneal elasticity. Because of the anisotropic corneal property due to its collagen structure, a nearly incompressible transversely isotropic (NITI) model was developed to characterize its elasticity. It has been shown that in-plane tensile and out-of-plane shear properties are defined by different moduli, E and G, respectively. Both E and G experienced significantly different stiffening responses to CXL. The increase in G relative to E suggests that while CXL strengthened the tensile strength of the tissue, the ability of the cornea to resist shearing forces was more dramatically altered. AmT-OCE appears capable of changing diagnostic criteria of ectatic corneal diseases, leading to early diagnosis and more precise surgical planning.

9:00

3aBAb2. Ocular biomechanics and glaucoma. Arthur J. Sit (Ophthalmology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, sit.arthur@mayo.edu), Arash Kazemi (Ophthalmology, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Mayo Clinic, Rochester, MN)

Glaucoma is the leading cause of irreversible blindness worldwide. Ocular biomechanical properties have been implicated in glaucoma pathogenesis, but existing commercial devices cannot determine changes in specific tissues. We recently performed a study to compare corneal wave speed (a measure of corneal elasticity) and ocular rigidity (a whole eye measurement of ocular biomechanics) between glaucomatous and normal eyes. Twenty glaucomatous eyes from 10 patients and 20 normal eyes from 13 controls, matched for age, intraocular pressure (IOP), and axial length were included. Ocular rigidity was calculated based on the difference in supine IOP by pneumatonometry with and without a 10-g weight. Corneal wave speed was determined by ultrasound surface wave elastography in which a small, 0.1 s harmonic vibration at 100 Hz was generated through the closed eyelids. Wave propagation was captured by an ultrasound transducer, and wave speed was determined from the phase change with distance. No significant differences in corneal wave speed between glaucomatous and normal eyes were detected ($P = 0.17$). However, ocular rigidity was significantly lower in glaucomatous eyes (0.0218 ± 0.0033 vs 0.0252 ± 0.0050 μl^{-1}, $P = 0.01$). While lower ocular rigidity is consistent with altered biomechanics in glaucoma, the lack of a difference in corneal wave speed suggests that corneal tissue may not be significantly affected, and scleral changes likely play a more important role.

9:20

3aBAb3. An ultrasound vibro-elastography technique for assessing papilledema. John J. Chen (Ophthalmology and Neurology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, chen.john@mayo.edu), Boran Zhou (Emory Univ., Atlanta, GA), Arash Kazemi, Arthur J. Sit (Ophthalmology, Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Mayo Clinic, Rochester, MN)

Papilledema is optic nerve swelling caused by increased intracranial hypertension, which has the potential to cause significant vision loss. Papilledema from idiopathic intracranial hypertension (IIH) is typically bilateral and symmetric, but can sometimes be asymmetric and even unilateral. The cause for this asymmetry is unknown. The purpose of this study was to develop ultrasound vibro-elastography (UVE) for assessing patients with papilledema to evaluate for biomechanical differences in these eyes. Nine patients with papilledema from IIH and 9 age-matched healthy control subjects were enrolled. A local harmonic vibration was used to generate wave propagation

through the eyelid. Three excitation frequencies of 100, 150, and 200 Hz were used. A 6.4 MHz ultrasound probe was used to measure wave propagation in the eye structures. Wave speeds were measured in the posterior sclera of the maculae. The magnitudes of the wave speed of the IIH patients' posterior sclera were significantly higher than those of healthy subjects. In patients with unilateral papilledema, the wave speed of the posterior sclera were higher in the eyes with papilledema than in the contralateral eyes without papilledema. UVE provides a noninvasive technique to measure the wave speed of posterior sclera, which is stiffer in eyes with papilledema.

9:40

3aBAb4. Treatment of neovascularization in the eye by photo-mediated ultrasound therapy. Xinmai Yang (Dept. of Mech. Eng., Univ. of Kansas, 1530 W 15th St., 3138 Learned Hall, Lawrence, KS 66045, xmyang@ku.edu), Yannis Paulus, and Xueding Wang (Univ. of Michigan, Ann Arbor, MI)

Neovascularization in the eye is a common disorder. Abnormal microvessels can develop in the cornea, retina, or choroid of the eye, leading to severe vision loss if left untreated. Current treatment methods carry disadvantages or side effects, such as drug-resistance or damage to the surrounding healthy tissues. By applying synchronized laser pulses and ultrasound bursts, we developed a technique, termed photo-mediated ultrasound therapy (PUT), as a localized antivascular method based on the enhanced cavitation mechanism. PUT takes advantage of the high native optical contrast between blood and surrounding tissues, and has the unique capability to self-target microvessels without damaging surrounding tissues. The technique utilizes nanosecond laser pulses, which is spatially and temporally synchronized with low duty-cycle ultrasound bursts of sub-MHz frequencies at the targeted microvessels. We have tested the feasibility of PUT on the removal of neovascularization in the cornea, retina and choroid in a rabbit eye model. We found that neovascularization in the eye could be greatly reduced after PUT, and the occlusion of microvessels could last up to 4 weeks. In conclusion, PUT holds significant promises as a non-invasive method to precisely remove microvessels in neurovascular eye diseases by more selectively treating vasculature with minimized side-effects.

10:00–10:20 Break

10:20

3aBAb5. Non-contact acoustic micro-tapping optical coherence elastography (AμT-OCE) in the cornea: Mechanical model, *ex vivo* and *in vivo* results, and direct comparison with mechanical tests. Ivan Pelivanov, Mitchell Kirby (Bioengineering, Univ. of Washington, Seattle, WA), Maju Kuriakose (Bioengineering, Univ. of Washington, 616 NE Northlake Pl, Benjamin Hall Blvd, Rm. 363, Seattle, WA 98105, majumk@uw.edu), Hong-Cin Liou, Matthew O'Donnell (Bioengineering, Univ. of Washington, Seattle, WA), Ruikang K. Wang, and Tueng T. Shen (Ophthalmology, Univ. of Washington, Seattle, WA)

The cornea is one of the primary determinants of visual performance. Its highly pressurized anisotropic structure of collagen fibrils embedded in a hydrated proteoglycan matrix forms the clear refractive primary lens of the eye. Despite recent success in corneal topography, there are no clinical tools to predict shape changes from vision-correction therapies such as LASIK and CXL due to the absence of non-contact methods that can map corneal elasticity *in vivo*. Here we review our newest results in quantification of corneal elasticity. Recently, we produced a technology combining non-contact excitation of sub-mm wavelength mechanical waves in the cornea using air-coupled acoustic micro-tapping (AμT), with 1.6 MHz scan-rate phase-sensitive OCT to track their propagation three-dimensionally in real time. However, corneal moduli reconstruction from measured wavefields requires an anisotropic elastic model and cornea's bounded, non-flat geometry must be taken into account. Thus, a nearly incompressible transversally isotropic (NITI) model of corneal elasticity was developed. We have performed multiple *ex vivo* and *in vivo* studies in rabbit, porcine and human corneas and compared AμT-OCE results directly with conventional mechanical tests (tensile extensometry and shear rheometry). These studies demonstrate that in-plane Young's and out-of-plane shear moduli can be accurately mapped with non-contact, non-invasive AμT-OCE.

10:40

3aBAb6. Acoustic radiation force optical coherence elastography for ocular biomechanics. Kirill Larin (Univ. Of Houston, 4800 Calhoun Rd., 3605 Cullen Blvd, Rm. 2028, Houston, TX 77204, klarin@uh.edu)

The mechanical properties of the cornea, limbus, and sclera provide important information regarding eye health. Optical coherence elastography (OCE) is highly suitable for assessing ocular tissues' mechanical properties due to its high resolution and superior displacement sensitivity. Typically, elastography involves an external mechanical excitation of the tissue. We recently adopted a method of Air-Coupled Acoustic Radiation Force for the generation of elastic waves in ocular tissues. This method shows several clear advantages other than the traditionally used air-puff excitation. However, the safety of this method needs to be studied before its clinical applications. Also, there is increased interest in passive elastography, where the mechanical response to natural physiological forces like heartbeat and respiration is measured. I will also demonstrate examples of using heartbeat for quantification of tissues' mechanical properties *in vivo*.

11:00

3aBAb7. Novel modeling of ultrasonic wave propagation in ophthalmic tissue. Sleiman R. Ghorayeb (Radiology and Molecular Medicine, Ultrasound Res. Lab, Hofstra Northwell School of Medicine, 133 Hofstra University, 207 Weed Hall, Hempstead, NY 11549, Sleiman.R.Ghorayeb@hofstra.edu)

Ultrasound has been widely used to diagnose a variety of anomalies present in ophthalmic tissue (detached retinas, tumors, cataracts, etc.), and to obtain a number of measurements in the eye like intraocular length, lens thickness, and so on. However, there has not been a clear understanding of how ultrasonic waves propagate and interact with the layers of the eye, and whether safety concerns are an issue. This presentation discusses two novel modeling techniques to complement these already existing tools to enhance ultrasonic evaluations of the eye by providing additional insight and a theoretical frame of reference. Axisymmetric finite element and PSpice transmission line models simulating a single-element transducer based on Redwood's version of Mason's equivalent circuit, a focusing lens, and a multi-layered medium that mimics propagation of ultrasound in the different layers of the eye are presented. Results are obtained for a

normal eye and for one in the case of a retinal detachment. A-scans are also obtained *in vitro* using an extracted normal sheep's eye for comparison. Both sets of results clearly show the transmission and reflection of the ultrasonic waves as they travel through the layers within the eye structure and confirm the validity of these modeling techniques.

11:20

3aBAb8. Measurement of the arterial wall elasticity using the velocity of flexural waves in retinal blood vessels. Gabrielle Laloy-Borgna (INSERM U1032, 151 Cours Albert Thomas, 69428 Lyon, France, gabrielle.laloy-borgna@inserm.fr), Léo Puyo (Univ. of LÜbeck, Paris, France), Michael Atlan (Inst. Langevin, Paris, France), and Stefan Catheline (INSERM U1032, Lyon, France)

Arterial stiffness (typically a Young modulus of 100 kPa) is known to be an indicator of the cardiovascular health of a patient. It is commonly used in clinics to prevent and anticipate cardiovascular diseases. The existing methods allowing to measure the arterial wall elasticity measuring the velocity of the natural pulse wave lack of accuracy. We propose here a method consisting in using laser Doppler holography experiments to estimate the arterial wall elasticity. Laser Doppler holography experiments are performed on patients and noise correlation algorithms are used on these data to measure the wave velocity. We measured separately the wavelength and the frequency of elastic waves, allowing to calculate their velocity. Due to the temporal resolution of the raw data and to the way the data are processed, the waves we detected are not the classical axisymmetric pulse wave. The physics of guided waves indicates that both antisymmetric (called A_0) and symmetric (called S_0) waves can propagate in tubes, and in this case only antisymmetric flexural waves are detected. They are both sensitive to the arterial wall elasticity, and at low frequencies, flexural (or antisymmetric) waves are much slower than the axisymmetric pulse wave, about 3 mm/s, allowing to increase greatly the measurement accuracy. When knowing the diameter of the blood vessel, for the first time it is possible to retrieve the arterial wall elasticity thanks to the antisymmetric wave velocity measurement.

WEDNESDAY MORNING, 1 DECEMBER 2021 — 502 (L)/503 (O), 9:00 A.M. TO 11:30 A.M.

Session 3aCA

Computational Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, Signal Processing in Acoustics, and Acoustical Oceanography: Showcases of High Performance Computing in Acoustics I

Kuangcheng Wu, Cochair
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Ralph T. Muehleisen, Cochair
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Shung H. Sung, Cochair
Troy, MI

Chair's Introduction—9:00

Invited Papers

9:05

3aCA1. Manufacturing-aware design of multi-material asymmetric acoustic absorbers. Tyler J. Wiest, Carolyn C. Seepersad (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

Advances in additive manufacturing technology are promising for the creation of materials to control acoustic waves. This work presents the design of an acoustic metasurface (AMS) created from a periodic two-dimensional array of multi-material scatterers embedded in an elastomeric matrix. Finite element analysis (FEA) of the AMS shows that the absorption of acoustic energy is dependent on

direction of incident wave propagation if the material properties are asymmetric within the scatterers while the acoustic transmission remains symmetric. Asymmetry of acoustic absorption is highly sensitive to small changes in scatterer geometry and property distribution. The AMS design must therefore consider inherent variability in manufacturing processes and the resultant stochastic asymmetric absorption performance of the as-built AMS, which is computationally expensive to determine using FEA. A machine learning classifier is trained to replace FEA and enable efficient robust design of the AMS via Monte Carlo simulations. The classifier is trained to be valid for a variety of characteristic manufacturing variations and the trivial expense to call it leads to significantly faster design with awareness of manufacturing variability. This work demonstrates how robust design approaches can be used to select fabrication methods suitable for acoustic materials. [Work supported by NSF and ONR.]

9:25

3aCA2. High performance computing and inverse problems in acoustics and structural acoustics. Gregory Bunting, Scott Miller, Clark Dohrmann (Computational Solid Mech. and Structural Dynam., Sandia National Labs., Albuquerque, NM), and Timothy Walsh (Simulation Modeling Sci., Sandia National Labs., Sandia National Labs., MS 0897, Albuquerque, NM 47906, tfwalsh@sandia.gov)

Engineering applications in acoustics and structural acoustics commonly require finite element models with large numbers of degrees of freedom. Examples include applications with infinite and semi-infinite domains, as well as problems at high frequency (e.g., ultrasound) where the number of wavelengths in the domain becomes large. The finite element solution of such problems, and their inclusion inside an optimization loop for inverse problems, inevitably involves large numbers of degrees of freedom and systems of equations. Modern high-performance computing is enabling numerical solutions of acoustics applications that were previously intractable. In this talk, we will present an overview of massively parallel acoustic and structural acoustic solution capabilities in Sandia's Sierra-SD (Structural-Dynamics) software. Sierra-SD is a massively parallel finite element application for structural dynamics and acoustics, having recently demonstrated the implicit solution of models with over 2×10^9 degrees of freedom and running on up to one hundred thousand distributed memory cores. Domain decomposition and MPI parallelism are used to split a large domain into many small domains that can be solved in parallel. Large-scale applications in ship shock loading and vibration of reentry bodies will be shown as demonstrations. [SNL is managed and operated by NTESS under DOE NNSA Contract DE-NA0003525.]

9:45

3aCA3. Multi-level parallelism for structural acoustic uncertainty quantification. Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., P.O. Box 30, M.S. 3220B, State College, PA 16801, axw274@psu.edu), Kyle Myers, and Micah Shepherd (Appl. Res. Lab., Penn State Univ., State College, PA)

Uncertainty quantification (UQ) of structural acoustic systems is currently of great interest and is expected to only become more prevalent in the future. Techniques such as generalized polynomial chaos, Gaussian metamodeling, and others allow a variety of UQ analyses to be performed by forming a surrogate model of the system of interest using only blackbox evaluations of the system and do so with a much reduced number of evaluations compared to the Monte Carlo method. However, to properly resolve the complicated behavior of acoustic or vibratory systems, and in particular how the uncertainty affects that behavior, this can still take a large number of system evaluations even when using these advanced UQ techniques. This work focuses on the tradeoffs between parallelizing the UQ method compared to the structural-acoustic blackbox and how a variety of implementation choices affect the run-time on high performance computing resources. Example problems with differing system sizes are used to demonstrate interesting features.

10:05

3aCA4. Introduction to the Department of Energy Leadership Computing Facilities. Ralph T. Muehleisen (Energy Systems, 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 look at your computation and see that cloud computing is not a viable solution? Do you think your problem could benefit from *really* high performance computing but don't know where to start in trying to access and use it? Acoustics researchers typically fail to think big with respect to their simulations—finding ways to reduce and limit their problems to fit a single workstation, a small cluster, a small HPC, or the cloud. The Department of Energy (DOE) Leadership Computing Facilities (LCF) are a set of the highest performance publicly accessible computing facilities in the U.S. Summit at Oak Ridge National Laboratory and Perlmutter at Lawrence Berkeley National Laboratory are #2 and #5 of the Top500 HPC list. Two Exascale computers: Frontier at Oak Ridge and Aurora at Argonne National Laboratory are set to arrive in the next couple of years. This presentation will introduce the DOE LCF and several of the allocation programs by which researchers and industry can gain access and training to use these incredible facilities.

10:25–10:40 Break

10:40

3aCA5. High-performance computing for long-range underwater acoustics. Noriyuki Kushida (Comprehensive Nuclear-Test-Ban Treaty Organization, Wagramastresse 5, Wien 1220, Austria, noriyuki.kushida@ctbto.org), Tiago Oliveira (Univ. of Aveiro, Aveiro, Portugal), and Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, RI)

The Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO) has been recording underwater signals using seven hydrophone stations which cover the entire oceans and provides the data publicly available. One of the lessons learned during the data analysis, such as in the ARA San Juan submarine tragedy, is that underwater sound propagation paths become complex in a long-range propagation, and as such, computer modelling, particularly three-dimensional (3D), can play an important role in understanding sound source location. The biggest stumbling block toward long-range 3D modeling is obviously its huge computational resource demand. In this presentation, we will discuss the GPU implementation of the 3D Split Step Fourier Parabolic Equation (SSFPE) solver, and the domain decomposition parallelization of our newly developed 2D + 1D Finite-Difference Time-Domain (FDTD) solver. The obtained computational performances are: (1) 3D SSFPE on GPU performs approximately 20 times faster than the original Matlab code and (2)

the parallel speedup of the FDTD solver with domain decomposition up to 4 nodes is 98%. Although the performances on a larger system are still of interest, those results already enable us to tackle realistic problems. Results highlight the importance of using high performance computing for realistic global scale underwater acoustic simulations.

Contributed Papers

11:00

3aCA6. Pros and cons of accelerated computing on T-phase propagation simulation with a 3D parabolic equation model. Tiago Oliveira (Univ. of Aveiro, Campus Universitário Santiago, Aveiro 3810-193, Portugal, toliveira@ua.pt), Noriyuki Kushida (Comprehensive Nuclear-Test-Ban Treaty Organization, Wien, Austria), and Ying-Tsong Lin (AOPE, Woods Hole Oceanographic Inst., Woods Hole, RI)

Submarine seismic activities can generate low frequency (<40 Hz) acoustic waves, known as T-phases, that propagate in the ocean with low attenuation for thousands of km. In this regard, high-performance computing is essential to simulate the propagation from an earthquake source to a receiver. This work discusses the pros and cons of accelerated computing on the simulation of T-phase propagation using a 3D geodesic Cartesian parabolic equation (PE) model. The approximately 15 000 km path of the T-phases generated in the Kermadec Trench (Pacific Ocean) to the Ascension Island (Atlantic Ocean) is simulated. This simulation requires an approximate $11 \times 2000 \times 15000$ km^3 (depth $\times$ cross-range $\times$ range) simulation domain to evaluate reflections and diffractions correctly. In general, this type of 3DPE long-range marching solution simulation in GPU is advantageous and up to 20 times faster than in sequential computation, while eight times faster than in multi-core computation using 28 cores. However, when high-resolution cross-range discretization (<50 m) is used, the GPU memory limits the computation. In these cases, the multi-core computation can become a better option to perform simulations. Numerical results revealed the importance of accelerated computing to understand 3D effects on T-phases passing the Drake Passage.

11:15

3aCA7. Geoacoustic inversion for a 14-km autonomous underwater vehicle survey on the Malta Plateau. Tim Sonnemann (Dept. of Geoscience, Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, tim.sonnemann@ucalgary.ca), Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, Calgary, AB, Canada), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

We consider signal processing and inversion of 1487 source instances recorded on a towed array along a 14-km seabed survey on the Malta Plateau. The data were acquired by autonomous underwater vehicle (AUV) and processed as reflection coefficients versus grazing angle and frequency. Data acquisition caused several artifacts that are studied by various data representations. These representations expose periodic fluctuations of reflection coefficients with angle and frequency that do not depend on seabed location. Averaging over source instances, frequencies, and angles reduces the artifacts. The Bayesian inversion assumes a one-dimensional seabed structure for each data set and is parametrized by an unknown number of homogeneous layers, sound velocities, densities, and attenuations. Results from individual inversions and sequential Monte Carlo sampling are compared. We demonstrate that removing data artifacts reduces over-fitting problems from previous considerations of the same data. Comparisons to piston and gravity core estimates, and separate wide-angle data show good agreement with the AUV results for two locations along the track. However, results at greater depths exhibit high uncertainty and strong influence of chosen prior boundaries. When considering results for all 1487 data sets, dipping and terminating layers are found along the track with high resolution ($\sim$10 cm).

Session 3aEA

Engineering Acoustics and Signal Processing in Acoustics: Acoustics and Human-Machine Interface

Gary W. Elko, Chair
mh Acoustics, 25A Summit Ave., Summit, NJ 07901

Chair's Introduction—8:45

Invited Papers

8:50

3aEA1. Robust speech processing for voice interfaces and video content understanding at Facebook. Michael Seltzer (Facebook AI, 1101 Dexter Ave. N, Seattle, WA 98109, mikeseltzer@fb.com)

As voice interfaces to devices and digital assistants have increased in popularity, so too, have the challenging environments in which they are expected to perform. In this talk, we'll present an overview of the signal processing and speech recognition AI modeling techniques that we have developed at Facebook to enable robust voice interaction on Portal video calling devices and Oculus VR headsets. We will also describe progress in captioning and understanding the wide variety of video content shared on Facebook apps, where the acoustic conditions are diverse and challenging and the audio is typically captured on commodity mobile phones. While such systems have been historically developed to run on powerful servers in the cloud, there is increasing interest in speech models that can run locally on the client device. We will describe the challenges of on-device processing and our recent progress in creating efficient, low-footprint speech models. Finally, we will present the challenges and future directions we are exploring to enable rich voice interactions on the next generation of computing devices, including augmented reality glasses.

9:15

3aEA2. Sound capture and speech enhancement for speech-enabled devices. Ivan J. Tashev (Microsoft Res., One Microsoft Way, Redmond, WA 98054, ivantash@microsoft.com) and Sebastain Braun (Microsoft Res., Redmond, WA)

In this talk we will make an overview of the acoustical design of the sound capture systems and discuss the general architecture of speech enhancement pipelines for the needs of distant speech recognition. The talk will discuss both classical algorithms using statistical signal processing and deep learning using neural networks. It will be illustrated with real-life examples from the acoustical design and speech enhancement audio pipelines in Kinect, HoloLens, and Microsoft Teams.

9:40

3aEA3. Reference-free AEC for Alexa Barge-In. Yiteng Huang (Alexa ML, Amazon, 7 West 34th St., 6th Fl., New York, NY 10001, yithuang@amazon.com) and Tao Zhang (Alexa ML, Amazon, Cambridge, MA)

When an Alexa built-in device is in playback mode, users need to make extra effort to cut through acoustic echo from the audio playback (music or synthesized speech signals) to activate the wakeword detector and start talking to Alexa, i.e., to barge into the conversation. These are high-friction events where Alexa customers typically experience a higher false reject rate (FRR). The acoustic echo canceller (AEC) and beamformer in the multichannel Audio Front End (AFE) are essential to mitigate the effect. But when playback volume is high, AFE outputs still contain strong residual echoes which downstream wakeword detection models could have difficulty to handle. In this talk, we will present an innovative multi-microphones yet array-agnostic, reference-free AEC tailored specifically for wakeword detection. As opposed to the traditional AEC, the new approach has five practically attractive benefits: (1) reference-free and array agnostic (a blind method suitable for in-model implementations); (2) inherently immune to loudspeaker nonlinearities; (3) free of synchronization/alignment hassle; (4) more microphones, more gains; (5) kills two birds with one stone (cancels both echo and noise). A proof-of-concept experiment will be discussed to validate the effectiveness of the proposed novel algorithm.

10:05

3aEA4. One engineer's approach to audio system design. Olen Rasp (Devices and Services, Google, 1600 Amphitheatre Parkway, Mountain View, CA 94043, olen@google.com)

Designing and implementing a complete audio system presents a unique set of challenges and relies upon a broad range of disciplines: covering acoustical, mechanical and electrical systems. This session will cover one approach to designing small, mass-produced, self-contained audio systems by describing key steps and explaining why they're important. Topics to be covered include: simulation, measurement and tuning.

10:30–10:45 Break

Contributed Papers

10:45

3aEA5. Circular microphone array beamforming to improve speech recognition for virtual agents. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com), Jens Meyer (mh Acoust., Fairfax, VT), Heinz Teutsch, and Tomas Gaensler (mh Acoust., Summit, NJ)

Under funding from an NIH NIDCD SBIR phase 1 and phase 2 grant, mh acoustics built a 16-element concentric circular microphone array using both analog and digital MEMS microphones. The circular microphone array design was trademarked as the EigenEar® microphone array and the unique beamforming processing is covered by U.S. patent 8903106. The presentation aims at showing that a circular third-order differential beamformer is capable of improving the recognition rate of popular voice recognition services used by a number of commercially available virtual assistants. These services offer an API to a client allowing speech data to be uploaded, analyzed, and returned in text form to the client. Experiments were done to compare the recognition accuracy from two separate audio streams. The EigenEar microphone array has two output channels and one channel contained the output of one omnidirectional microphone in the array and the other channel contained the output of the beamformer. The two audio streams are time synchronous so both streams presented to the virtual assistant are time aligned. The talk will describe the EigenEar circular microphone array and how the experiment was setup and the ASR results obtained from the virtual agents.

11:00

3aEA6. Vibro-tactile display for the visually impaired using miniature electrostatic transducers. Lillian Fullford (Mech. Eng., Tufts Univ., Medford, MA), Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu), and Jonathan Bernstein (Draper, Cambridge, MA)

A vibro-tactile electronic display is being developed to aid the visually impaired community by increasing access to graphical and textual information. A high-density vibro-tactile arrayed display would improve the user experience while moving away from external displays on the market. We propose to use millimeter scale electrostatic transducers operating at frequencies well below resonance as candidate array pixel elements. Motion detectable by a fingertip, mechanical impedance of a fingertip, and frequency at which fingertips can detect contact were considered. Motion of 5 μm with a fingertip touching the device and 10 μm without a finger touching the device is desired with the device at a mechanical impedance of 480 N/m, which is twice the mechanical impedance of the fingertip (240 N/m measured by Diller and Kyung). Transducer membrane thickness, force on the fingertip, and gap width are being optimized through prototyping and finite element analysis. A prototyped row of pixels was manufactured using 0.051 mm thick metalized Mylar film with an air gap of 0.02 mm and a diameter of 1 to 2 mm per pixel, driven by AC electrostatics at 400 V peak to peak. Motion analysis is being conducted using a laser vibrometer. Preliminary results demonstrate motion of 10 μm peak to peak at low frequencies, with further analysis and optimization ongoing.

11:15

3aEA7. Multi-user interactive systems for immersive virtual environments. Samuel Chabot (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), Jonathan Mathews (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, mathej4@rpi.edu), and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Human-scale immersive spaces are increasingly prevalent in real-world contexts as viable applications for large-scale displays, speaker arrays, and reactive platforms continue to be discovered. One of the primary challenges hindering adoption of these modalities is the integration of intuitive interaction frameworks, allowing individuals in the space to engage naturally with the environment. In the Collaborative Research-Augmented Immersive Virtual Environment, or CRAIVE-Lab, at Rensselaer Polytechnic Institute, we have integrated a variety of technologies to foster natural engagement with the space. We developed a dynamic display environment to enable user-specific contribution to and control of the virtual space. In addition, we installed a sensor network consisting of both video and audio sensor arrays to provide fused location data and voice activity detection for each user in the space without body-worn devices. Integration of these technologies into the environment facilitates an immersive experience by fostering natural interaction between an individual and the lab system.

11:30–12:00
Panel Discussion

Session 3aMUa

Musical Acoustics: Making Music During a Pandemic

Thomas R. Moore, Chair
Department of Physics, Rollins College, Box 2743, Rollins College, Winter Park, FL 32789

Chair's Introduction—9:00

Invited Paper

9:05

3aMUa1. Measurements and simulations of aerosol released while singing and playing wind instruments. Tehya Stockman (Civil, Environ., and Architectural Eng., Univ. of Colorado Boulder, 609 UCB, Boulder, CO 80309, tehya.stockman@colorado.edu), Shengwei Zhu (Mech. Eng., Univ. of Maryland, College Park, MD), Abhishek Kumar (Mech. Eng., Univ. of Colorado Boulder, Boulder, CO), Lingzhe Wang (Mech. Eng., Univ. of Maryland, College Park, MD), Sameer Patel (Dept. of Civil Eng., Indian Inst. of Technol., Gandhinagar, Gujrat, India), James Weaver (National Federation of State High School Associations, Indianapolis, IN), Mark Spede (Dept. of Performing Arts, Clemson Univ., Clemson, SC), Donald Milton (Maryland Inst. for Appl. Environ. Health, Univ. of Maryland, College Park, MD), Darin Toohey (Atmospheric and Oceanic Sci., Univ. of Colorado Boulder, Boulder, CO), Marina E. Vance, Jean Hertzberg (Mech. Eng., Univ. of Colorado Boulder, Boulder, CO), Jelena Srebric (Mech. Eng., Univ. of Maryland, College Park, MD), and Shelly Miller (Mech. Eng., Univ. of Colorado Boulder, Boulder, CO)

Outbreaks from choirs showed that singing brings potential risk of COVID-19 infection. There is less known about the risks of airborne infection from other musical performance, such as playing wind instruments or performing theatre. In addition, it is important to understand methods to reduce infection risk. In this study, we used a variety of methods, including flow visualization, aerosol and CO_2 measurements, and computational fluid dynamics (CFD) modeling to understand the different components that can lead to transmission risk from musical performance and risk mitigation. This study was possible because of a partnership across academic departments and institutions and collaboration with the National Federation of State High School Associations and the College Band Directors National Association. Funding for this project was provided by an international coalition of music organizations. We found that plumes from musical performance were highly directional, unsteady, and vary considerably in time and space. Aerosol number concentration measured at the bell of the clarinet were comparable to singing. Face and bell masks attenuated plume velocities and lengths and decreased aerosol concentrations measured in front of the masks. CFD modeling showed differences between indoor and outdoor environments and that lowest risk of airborne COVID-19 infection occurred at less than 30 min of exposure indoors and less than 60 minutes outdoors.

Contributed Papers

9:25

3aMUa2. Exploring the effects of masks and instrument covers on air flow. Sydney Rollins (Phys., Whitman College, 345 Boyer Ave., Whitman College, Walla Walla, WA 99362, rollinsr@whitman.edu) and Kurt R. Hoffman (Phys., Whitman College, Walla Walla, WA)

In this desperate time of COVID, can we still do the things we love safely? Efforts to keep schools safe for music rehearsals during the past academic year included singing with masks and covering various parts of musical instruments with cloth barriers to limit the flow of exhaled droplets into the air. We assembled a Schlieren system to test the effects of these different adaptations to modify the flow of air for different instruments, including the voice. This paper presents the results of our measurements and an assessment of their effectiveness at reducing or redirecting the flow of air. With the arrival of the Delta variant, new questions are being raised about safety and exposure due to the possibility of breakthrough infections. Therefore, we are revisiting these remediation methods to evaluate their potential effectiveness in light of changing safety risks.

9:40

3aMUa3. Aerosol propagation and acoustic effects while singing with a face mask. Thomas R. Moore (Dept. of Phys., Rollins College, Box 2743, Rollins College, Winter Park, FL 32789, tmoore@rollins.edu)

The flow of aspirated breath while singing was studied using transmission electronic speckle pattern interferometry. The resulting images show that when a vocalist wears a facemask the aspirated aerosol generally does not propagate through the mask, but instead exits through the sides. The aerosol is then entrained in the upward flow created by the heat from the body of the vocalist. Studies of a professional soprano singing the same phrase while wearing several different masks revealed that all of the masks that were tested resulted in no detectable transmission of the aspirated aerosol through the mask. The acoustic spectra indicate that all of the masks reduce the relative power in frequencies above 1 kHz, but a singer's mask has a higher transmission in this frequency range than the more commonly available masks.

Session 3aMUb

Musical Acoustics: General Topics in Musical Acoustics I

Christopher Elmer, Chair
American Mathematical Society, 17231 Lands End, Chelsea, MI 48118

Contributed Papers

10:10

3aMUb1. Efficient digital waveguide synthesis of a pipe organ. Champ C. Darabundit (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct, Stanford, CA 94305, champ@ccrma.stanford.edu) and Julius O. Smith (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., Stanford, CA)

The modern pipe organ can have as many as 10 000 pipes, each of which is a simple sound generator. In order to digitally synthesize a pipe organ, each pipe requires its own digital waveguide model due to its individual geometry and state. To complicate the matter, pipe geometry can vary greatly across different pipe organ ranks, pipes can be reed-driven or air-jet-driven, and it is possible for a single key to sound many ranks at once. As a result, developing an acoustic model of even a small pipe organ is intensive both in the modeling and in the computational demands for real-time synthesis. For modeling the pipes, we propose a scalable lossy digital waveguide framework whose parameters can be changed based on a pipe's geometric model. We utilize the Faust functional programming language to produce high-performance digital signal processing code.

10:25

3aMUb2. Machine learning recognition of vibrational modes in ESPI data. Maya Greene (Phys., Whitman College, 345 Boyer Ave., Whitman College, Walla Walla, WA 99362, greenemw@whitman.edu) and Kurt R. Hoffman (Phys., Whitman College, Walla Walla, WA)

We present a neural network (NN), constructed in Mathematica, that classifies images generated by electronic speckle pattern interferometry (ESPI). The goal of this project was to use the program to identify features in the images representing resonance behavior of the object. Much of our data was collected from unfinished wood surfaces, resulting in only the appearance of nodal lines in our ESPI data rather than the concentric rings that are hallmarks of data from finished instruments. The NN described here was trained to give four different output classifications: no resonant features, concentric ring patterns, clear nodal line patterns, and weak resonance features that correspond to ambiguous features observed at the high and low edges of the resonance band. The NN was trained using a collection of images from scans by an interferometer of multiple wood plates, an aluminium plate, and guitar top plates. In order to improve the accuracy of the classifier, we also explored image filtering techniques to increase the contrast of the images. This NN will eventually automate the process of finding resonant frequencies in real time when scanning new surfaces.

10:40

3aMUb3. Measuring guitar top plate resonances at stages of fabrication. Max Horne (Phys., Whitman College, 345 Boyer Ave., Whitman College, Walla Walla, WA 99362, hornemc@whitman.edu) and Kurt R. Hoffman (Phys., Whitman College, Walla Walla, WA)

This project investigates how the shape, quality, and boundary conditions of an object affect its normal modes of vibration. Here we report the results of electronic speckle pattern interferometry (ESPI) measurements on several potential guitar top plates in moving from a solid rectangle shape to the final top plate shape with a sound hole and bracing. The measured nodal patterns were similar for two distinct top plates as they were cut and sanded into final form. The measured frequencies of each normal mode were similar with typical variations of less than 10%. The experimental data was compared to theoretical data modeled in COMSOL to test the accuracy of the input parameters characterizing the wood. We found that the model results agree with the experimental data well at low frequencies, but the accuracy of the model decreases when higher frequency modes are considered.

10:55

3aMUb4. Transient oscillations in steelpan drums tracked via machine learning. Scott H. Hawley (Dept. of Chemistry and Phys., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu) and Andrew C. Morrison (Natural Sci., Joliet Junior College, Joliet, IL)

Applying a machine learning based object detector to high-speed videos of Caribbean steelpan drums illuminated by laser electronic speckle pattern interferometry (ESPI), we can track the development of sympathetic vibrations in response to drum strikes. This object detector was trained on a dataset of crowdsourced human-annotated images obtained in the Steelpan Vibrations Project on the Zooniverse platform. When the model is used to supply annotations to thousands of unlabeled video frames, we measure visual oscillations at frequencies consistent with audio recordings of these drum strikes. In this talk we share early physics results such as the unanticipated phenomenon of sympathetic oscillations of higher-octave notes that significantly precede the rise in sound intensity of the corresponding second harmonic tones; the mechanism responsible for this remains unidentified.

11:10

3aMUb5. Using convolutional neural networks to estimate pitch directly from steelpan audio signals. Colin Malloy (School of Music, Univ. of Victoria, MacLaurin Bldg., B102, Victoria, BC, Canada, malloyc@uvic.ca) and Jason Ye (none, Victoria, BC, Canada)

Estimating the pitch, or fundamental frequency, of a monophonic audio signal is a fundamental task in computational audio analysis with many downstream applications such as automatic transcription. The current general state of the art method for pitch detection is CRéPE: a Convolutional Representation for Pitch Estimation. CRéPE is a deep convolutional neural network designed to estimate the fundamental frequency of an audio signal directly from the waveform. However, CRéPE, and other general pitch detection methods, do not perform well on steelpan audio. This is likely due to the steelpan's complex spectral characteristics that differentiate it timbrally from other sound sources. We combine a deep convolutional neural network architecture based on CRéPE with a training dataset of steelpan audio from several distinct sounding tenor steelpans to achieve improved tenor steelpan pitch detection directly from the audio signal. We assess our model's ability to generalize by evaluating it with a test dataset that includes audio samples from steelpans that have no samples as part of the training set and compare these results to CRéPE's performance on the same test dataset.

11:25

3aMUb6. Steelpan fundamental frequency estimation through audio feature extraction and deep neural networks. Colin Malloy (School of Music, Univ. of Victoria, MacLaurin Bldg., B102, Victoria, BC, Canada, malloyc@uvic.ca)

The estimation of fundamental frequency, or pitch, is a fundamental task in computational audio analysis with a variety of applications. Steelpan audio has proven difficult for general pitch detection methods CRéPE and pYin. CRéPE is a method that uses a deep convolutional neural network to perform pitch estimation directly from the audio signal while pYIN is a digital signal processing-based approach. Audio feature extraction is the process of using digital signal processing techniques to extract low level audio information from signals. Combining audio feature extraction with logistic regression is currently the best performing steelpan pitch estimation method, but the efficacy of using deep neural networks in lieu of traditional machine learning algorithms has yet to be determined. This paper compares the performance and computational requirements of a deep neural network-based architecture against logistic regression as well as the established pYIN and CRéPE pitch detection methods to determine which method is the most accurate and efficient. All of these methods are evaluated on a test dataset containing one-hit audio samples from several distinct sounding steelpans. Generalization to other steelpans is assessed by including samples in the test dataset from steelpans for which no samples are in the training dataset.

WEDNESDAY MORNING, 1 DECEMBER 2021 — 301 (L)/304 (O), 8:40 A.M. TO 12:00 NOON

Session 3aNS

Noise, Physical Acoustics, ASA Committee on Standards, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Jet and Launch Vehicle Noise

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

Caroline P. Lubert, Cochair
Mathematics and Statistics, James Madison University, 301 Dixie Avenue, Harrisonburg, VA 22801

Chair's Introduction—8:40

Invited Papers

8:45

3aNS1. Description of jet noise sources in a generalized aeroacoustic framework. Chitrarth Prasad (Dept of Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E549, Columbus, OH 43210, chitrarth2009@gmail.com) and Datta Gaitonde (Dept of Mech. and Aerosp. Eng., The Ohio State Univ., Columbus, OH)

The present investigation examines the use of a Generalized Aeroacoustic Framework (GAF) to provide insights into the mechanisms which convert turbulence energy into sound. A key step is the selection of Fluctuating Stagnation Enthalpy (FSE), from Momentum Potential Theory, as a generalized acoustic variable. The Navier–Stokes equations are rewritten as an exact, inhomogeneous, convected wave equation, which governs the generation and propagation of FSE from the jet shear layer to the far-field. Using benchmark LES databases as truth models, it is shown that the FSE, scaled by the ambient density, is a surrogate for pressure fluctuations away from the jet shear layer. Thus, sources terms in the FSE wave equation provide an improved understanding of the local mechanisms responsible for noise radiation from the jet shear layer. A systematic inspection of the instantaneous distributions of the FSE sources identifies the most acoustically relevant mechanisms in the jet shear layer. The linear component of the mechanism is responsible for the downstream noise signature due to the initial Kelvin–Helmholtz instability whereas the non-linear component is dominant further downstream and can be related to intermittent acoustic bursts near the core-collapse.

9:05

3aNS2. Examination of Mach wave coalescence in a Mach 3 jet flow. William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, The University of Texas at Austin, Austin, TX 78713-8029, william.willis@utexas.edu), John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Dept. of Medicine, Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA), Charles E. Tinney, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Prior acoustic measurements of the sound field produced by a laboratory-scale, Mach 3 jet flow [Baars *et al.*, *AIAA* (2013); Fiévet *et al.*, *AIAA* (2016)] showed a discrepancy between the theoretical prediction that shocks should not form, based on effective Gol'dberg numbers, and the apparent observation of steepened waves. This suggests an additional mechanism relating to wave coalescence is responsible for increased wave steepening. To better understand this new mechanism, high framerate schlieren images of sound waves propagating from the post potential core of the same Mach 3 jet are studied. A shock detection algorithm is developed to isolate shock-like features in the images and along the propagation path, which are then used to determine occurrences of coalescence-induced steepening in the flow. Numerical models of coalescing waves using the Khokhlov–Zabolotskaya–Kuznetzov (KZK) equation are then leveraged to determine identifiable characteristics of coalescence events to improve detection. POD-based reduced-order representations of the isolated events are then studied in a Lagrangian frame. Ensemble averages of tracked events are used to identify common patterns for coalescence, which is believed to contribute to "crackle" noise in full-scale jet flows. [WAW is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

9:25

3aNS3. On the possibility of nonlinear reflections in shock-containing noise near a high-performance military aircraft. Aaron B. Vaughn (Dept. of Phys. and Astronomy, Brigham Young Univ., Eyring Sci. Center N201, Provo, UT 84602, aaron.b.vaughn@nasa.gov), Kevin M. Leete, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Bradley R. Adams (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), and J. M. Downing (Blue Ridge Res. and Consulting, LCC, Asheville, NC)

The skewness of the pressure time derivative, or derivative skewness, is a useful indicator of crackle perception in jet noise. Near a military aircraft, derivative skewness values are greater at ground-based measurements relative to nearby off-ground locations. As proposed in Vaughn *et al.* [*J. Acoust. Soc. Am.* **149**(4), 2403–2414 (2021)], nonlinear reflections are a possible explanation for the increased derivative skewness values at the ground. For an ensemble of windowed shock events, the propagation angle, apparent source location, and angle of incidence relative to the ground are estimated using a two-point cross correlation. The shocks are then classified into shock reflection regimes based on their combination of angle of incidence and measured shock strength. A significant number of shocks measured along the ground array fall in the nonlinear reflection regime. A nonlinear shock reflection results in a pressure increase at the shock greater than twice the free-field pressure, which causes an increase in the derivative skewness value. Therefore, measurements at the ground may overpredict the crackle perception of a standing observer.

9:45

3aNS4. Application of spectral proper orthogonal decomposition and resolvent analysis to periodically forced supersonic jets. Liam Heidt (Mech. and Civil Eng., California Inst. of Technol., 1200 E California Blvd, MC 105-50, Pasadena, CA 91125, lheidt@caltech.edu), Tim Colonius (Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Akhil Nekkanti, Oliver T. Schmidt (Jacobs School of Eng., Univ. of California San Diego, La Jolla, CA), Igor A. Maia, and Peter Jordan (Département of Fluides Thermique et Combustion, Inst. Pprime-CNRS-Université de Poitiers-ENSMA, Poitiers, France)

The mechanisms by which near-nozzle forcing alters the turbulence structure and far-field sound of turbulent jets are not well understood. We perform large-eddy simulations of subsonic and supersonic axisymmetric isothermal turbulent jets subjected to an axisymmetric periodic forcing. The triple decomposition framework and spectral proper orthogonal decomposition are used to study the effect of the forcing on the underlying turbulence spectrum and determine how the periodic forcing alters coherent non-phase-locked structures. For subsonic jets, high-amplitude (1% of the jet velocity) low-frequency, $St_f = 0.3$, forcing is required to achieve a small change to the underlying turbulence spectrum and most energetic modes of the subsonic jet, despite producing a very energetic, phase-locked (tonal) response. The changes in the spectrum and the phase-locked structures are predicted well via resolvent analysis and linearized Navier-Stokes computations performed on the new turbulent mean, respectively. This suggests that there is little nonlinear interaction between the phase-locked structures and natural turbulence. High-frequency forcing, $St_f = 1.5$, shows similarly linear behavior except for around $St \approx 0.75$, which appears to be associated with vortex pairings triggered by the natural turbulence. Results of a similar analysis performed on a Mach 1.5 ideally expanded isothermal jet will also be presented.

10:05–10:25 Break

10:25

3aNS5. Insights into heated, supersonic jet noise gained from writing a review article on launch vehicle noise. Caroline P. Lubert (Mathematics and Statistics, James Madison Univ., Harrisonburg, VA), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu), and Seiji Tsutsumi (Japan Aerosp. Exploration Agency, Sagamihara, Kanagawa, Japan)

At the time of writing this abstract, a review article is being finalized for publication in JASA entitled a "Supersonic jet noise from launch vehicles: 50 years since NASA SP-8072," it describes the current state-of-the-art in rocket and launch vehicle noise research, including how highly heated, supersonic rocket plumes differ from other jets, the origin and nature of the radiated sound from both free and impinging rocket plumes, and techniques for pad noise mitigation during rocket launches. Existing and candidate methods (both empirical and numerical) for modeling rocket launch noise are reviewed, including the ubiquitous NASA SP-8072 methodology, which has formed the cornerstone of many rocket launch noise prediction models over the past half century. This talk will briefly discuss some of the insights gained as a result of writing this review and show how they led to the inevitable conclusion that an entirely new approach to rocket plume noise modeling—one that incorporates the underlying physics of the noise generation mechanisms—must be pursued.

10:45

3aNS6. Acoustical measurements of a Delta IV Heavy launch (NROL-82). Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT, grant_hart@byu.edu), Logan T. Mathews, Mark C. Anderson, J. T. Durrant, Michael S. Bassett, Samuel A. Olausson, Griffin Houston, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

On 26 April 2021, the BYU Acoustics Research Group made noise measurements during launch of a Delta IV Heavy rocket (NROL-82) from Vandenberg Space Force Base in California. Measurement locations ranged in distance from 330 m to 19 km from the launch pad. Some sensors were arranged in a rough arc of 1i–4 km radius and others were placed along both an east-west line and a north-south line. Because of nearby mountains, some measurement stations were able to be located above the vehicle at the pad. We were therefore able to measure the noise in the different directions and at different distances, spanning a large angular range. This presentation will discuss an overview of noise level, frequency, and temporal characteristics. Additionally, two stations consisted of 5-meter radius, multimicrophone vector probes. Using these probes we were able to track the trajectory of the rocket ascent; our acoustically traced trajectory agrees well with the actual trajectory data. The results of these analyses will be presented.

11:00

3aNS7. Overall sound pressure levels and peak directivity of the NROL-82 Delta IV Heavy Launch. Griffin Houston (Dept. of Phys. and Astronomy, Brigham Young Univ., 256 N 100 E, Provo, UT 84606, griffinahouston@gmail.com), Samuel A. Olausson, Logan T. Mathews, Mark C. Anderson, J. T. Durrant, Michael S. Bassett, Grant W. Hart, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Rocket launch noise has historically been analyzed for directivity and sound power. To compare against previous findings, this study uses data acquired from the NROL-82 Delta IV Heavy launch from Vandenberg Space Force Base, CA, USA. Rocket trajectory data are used to calculate the 3D angle between the plume and several measurement stations in the near and far fields. Overall sound pressure level, directivity, and sound power level are calculated, and the results compared across stations.

11:15

3aNS8. Predicting supersonic jet mode-switching using non-negative matrix factorization. Aprameya Satish (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, aprameya.satish@gtri.gatech.edu), Aharon Karon, and Alessio Medda (Georgia Tech Res. Inst., Smyrna, GA)

Imperfectly expanded supersonic jets contain a shock-cell structure in the jet plume which plays a role in generating screech tones along with other jet noise sources. Depending on the Mach number of the jet, two types of jet motion or “jet modes”—a toroidal mode and flapping mode may occur, resulting in the generation of different screech tones that are not harmonics of each other. At a given jet condition, these jet modes (and the corresponding screech tones) can switch between each other based on environmental conditions in a seemingly non-deterministic manner. Understanding when mode-switching occurs at a fine time resolution can provide better understanding of the sensitivity of the jet structure dynamics. This work presents the preliminary results of a novel method that predicts when a particular screech tone is active using non-negative matrix factorization. Using this technique, spectrograms of jet noise are decomposed into a set of basis vectors corresponding to the screech tones present in the jet, and a set of activation functions that indicate when each screech basis is dominant. An activation function comparison technique is then used to identify activations corresponding to mode-switching.

11:30

3aNS9. On the stability of parallel shear flows with embedded swirl. Lu Zhao (Aerosp. Eng., Univ. of Kansas, 2600 W 6th St., Apt. C6, Lawrence, KS 66049, luzhao@ku.edu), Saeed Farokhi, and Ray Taghavi (Aerosp. Eng., Univ. of Kansas, Lawrence, KS)

The temporal stability of inviscid parallel shear flows with embedded swirl is investigated. The base flow is divided into three regions: an inner irrotational jet flow, a thin swirl layer on the outer rim of the nozzle, and an external potential flow outside the nozzle. The swirl distribution in the thin shear layer is of solid-body rotation and of Rankine type outside the nozzle. The axial velocity profile in the jet base flow is assumed to be top-hat. The linear dispersion relation was derived and solved numerically. A parametric study on the effect of swirl parameter, S, and the swirl layer thickness fraction, d, on the amplification rates of the unstable modes is performed. A helical instability wave with azimuthal wave number, $m = -3$, is the most unstable mode with $S = 0.8$ and $= 0.08$. Our results indicate that the embedded thin layer swirl is capable of triggering the centrifugal instability waves to promote jet mixing. This powerful technique may be employed to mitigate jet noise without incurring significant thrust loss.

11:45

3aNS10. Physics-based filtering of irrotational wavepackets from high-speed Schlieren. Chitrarth Prasad (Dept of Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E549, Columbus, OH 43210, chitrarth2009@gmail.com) and Datta Gaitonde (Dept of Mech. and Aerosp. Eng., The Ohio State Univ., Columbus, OH)

The eduction of acoustically efficient wavepacket structures from experimental measurements or high-fidelity numerical data aids in understanding different noise generation mechanisms in jets. This is often achieved by employing either signal processing techniques or modal decompositions. The former often require some user-defined thresholds that must be adjusted for each operating condition whereas the latter are more robust and can be applied to any jet flow-field irrespective of the operating conditions. The application of physics-based decompositions requires time-accurate spatio-temporally resolved data of the type obtained from well-validated numerical databases, which require substantial computational resources. There is an inherent advantage in extending decomposition-based approaches to experimental measurements, which can provide a substantially longer time series of data compared to numerical simulations. This investigation explores the application of Doak’s Momentum Potential Theory to high-speed schlieren imaging, which is a widely used flow visualization tool in jet noise research. The procedure is applied to different test-cases, selected based on the availability of high-quality experimental measurements and numerical data. The coherent wavepackets obtained exhibit superior spatio-temporal coherence and radiative efficiency that capture the noise radiation mechanisms in the flow.

Session 3aPA

Physical Acoustics, Signal Processing in Acoustics, ASA Committee on Standards, Engineering Acoustics, and Computational Acoustics: Infrasound I

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Invited Papers

8:00

3aPA1. The International Data Centre infrasound processing system, a 25 years travel. Pierrick Mialle (CTBTO Preparatory Commission, Vienna Int. Ctr., P.O. Box 1200, Vienna 1400, Austria, pierrick.mialle@ctbto.org), Paulina Bittner, David Applbaum, and Ronan Le Bras (CTBTO Preparatory Commission, Vienna, Austria)

In 2001, when the first data from an International Monitoring System infrasound station started to arrive in near real-time at the International Data Centre (IDC), its infrasound processing system was in a premature state. The IDC embarked for a multi-year design and development of its dedicated processing system, which led to operational IDC automatic processing and interactive analysis systems in 2010. In the next ten years the IDC produced over 40,000 infrasound events reviewed by expert analysts. In an effort to continue advancing its methods, improving its automatic system and providing software packages to Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO) users, the IDC focused on several projects. First, the automatic system for the identification of valid signals was redesigned with the development of DTK-(G)PMCC (Progressive Multi-Channel Correlation), which is made available to CTBTO users within NDC-in-a-Box. And second, an infrasound model was developed for automatic waveform network processing software NET-VISA with an emphasis on the optimization of the network detection threshold by identifying ways to refine signal characterization methodology and association criteria. Ongoing and future improvements of the IDC processing system are planned to further reduce analyst workload that includes atmospheric propagation modeling and enhancements of the automatic pipeline components.

8:20

3aPA2. Initial performance analysis of an array of low-cost infrasound sensors with 43 elements. Garth Frazier (NCPA, Univ. of MS, NCPA, University of MS, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu), Carrick Talmadge (NCPA, Univ. of MS, Oxford, MS), Roger M. Waxler (NCPA, Univ. of MS, University, MS), Hank Buchanan (NCPA, Univ. of MS, Oxford, MS), William (Chip) Audette, and Chris Brooks (Creare, Hanover, NH)

Many infrasound sensor arrays are limited in the number of elements because of the cost of individual sensors. While some of these sensors are of exceptional quality and have calibrated (amplitude and phase) response to frequencies below 0.01 Hz, not all infrasound sensing applications require this degree of accuracy and precision. Moreover, to achieve significant reduction of the influence of wind noise at infrasound frequencies below 1 Hz requires large and/or complex mechanical wind screen apparatus. There are applications for which the benefits of having a so-called "large-N" array of lower cost, lower performance sensors outweigh the benefits of very high performance sensors. These benefits include wind noise reduction without large windscreens and the ability to resolve several acoustic sources simultaneously. This presentation will present preliminary array processing performance results from the deployment of an array of 43 recently developed, experimental low-cost infrasound sensors.

8:40

3aPA3. Evidence for short temporal atmospheric (tropospheric) variations observed by infrasonic signals. Gil Averbuch (Earth Sci., Southern Methodist Univ., 515 W 10th St., Apt. 116, Dallas, TX 75208, gaverbuch@smu.edu), Miro R. Giannone, Stephen Arrowsmith (Earth Sci., Southern Methodist Univ., Dallas, TX), and Jacob Anderson (Dept. of Geoscience, Boise State Univ., Boise, ID)

Infrasound monitoring is used in the forensic analysis of events, to study the physical processes of sources of interest and to probe the atmosphere. The dynamical nature of the atmosphere and the use of infrasound as a forensic tool lead to the following questions: (1) what is the time-scale of atmospheric variability that affects infrasonic signals? (2) can we link variations of infrasound signals to specific atmospheric phenomena? This study addresses these questions by monitoring a repetitive infrasound source and its corresponding tropospheric returns 54 km away. Source-receiver empirical Green's functions are obtained every 20 s and used to demonstrate the effect of atmospheric temporal variability on infrasound propagation. In addition, observations are compared to predicted simulated signals based on realistic atmospheric conditions. It is shown that infrasound properties change within tens of seconds. Particularly, phases can appear and disappear, the propagation time decreases, and the signals' energy fluctuates. Furthermore, the similarity between the

predicted and observed signals varies significantly within a few minutes. The observed changes are related to variations in temperature and wind, which are coupled to dynamic processes such as radiation, gravity waves, and turbulence. Therefore, this study highlights the potential of high temporal infrasound-based atmospheric sounding.

9:00

3aPA4. Source estimation through sensitivity-based mode decomposition of sequential data. Christophe Millet (CEA, Bruyères-le-Châtel, France, christophe.millet@cea.fr)

Characterizing explosions from infrasound is usually performed using posterior probability density functions (PDFs), which involves a large number of propagation model runs. These propagation models require atmospheric specifications and a model for representing small-scale atmospheric fluctuations, such as gravity waves. While defining a proper probabilistic model for the input parameters of these models may become a burden, in real-world cases, only a limited number of input parameters happens to influence the acoustic waveforms significantly. These parameters can be selected using quantitative importance measures or sensitivity indices that allow one to rank the sources of uncertainty. In this work, we introduce a method which is able to extract information from these sensitivity indices in time-frequency space, using waveforms generated by a full-wave propagation model. The extracted modes can be used to determine the relevant patterns that contribute most to the sensitivity captured in the data sequence or to project the perturbed model onto a system of fewer degrees of freedom. The restriction of the likelihood to subdomains where the sensitivity to gravity waves is weak allows a better estimate of the posterior PDF for source parameters. Demonstrations of the method are presented using waveforms recorded at thousands of kilometers of explosions.

9:20

3aPA5. An external calibrator system for the hyperion sensors. Carrick Talmadge (NCPA, Univ. of MS, Oxford, MS, clt@olemiss.edu)

The NCPA has developed an external calibrator system which is nearing maturity. This calibrator is attached as a replacement sensor lid for the Hyperion sensor, and injects the signal into the back volume of the Hyperion sensor. When the external calibrator is mated to the sensor, the integrated sensor-external calibrator package behaves as a self-calibrating sensor. The external calibrator allows the Hyperion to operate nominally and without significant change in response with the external calibrator installed. The calibrator is driven by an external signal generated by the digitizer (e.g., the CAL signal on a GEOTECH). This system is capable of producing signals with amplitudes greater than 20-Pa in the operational environment between 0.01 and 10 Hz. We report here on the performance metrics (frequency flatness, level linearity, etc.) as well as summarize the theory of operation of the device.

9:40–9:55 Break

Contributed Papers

9:55

3aPA6. Infrasound phase arrival uncertainty and seismoacoustic event location comparison. Clinton Koch (Sandia National Labs, 1515 Eubank Blvd SE, Albuquerque, NM 87123, Clikoch@sandia.gov), Fransiska Dugick, Elizabeth Berg, Sarah Albert, Matthew Peterson, and Raquel Guzman (Sandia National Labs, Albuquerque, NM)

Using a catalogue of over 60 seismoacoustic events compiled for a diverse variety of source-types we estimate probabilistic locations using seismic and acoustic phenomena, both individually, and combined. Previous studies have shown that combining seismic and acoustic observations can improve location estimates under specific circumstances. By examining a wide variety of seismoacoustic arrivals we further explore the capabilities and limitations of seismoacoustic event locations. The size and shape of the confidence region associated with a given event location is dependent on pick and model uncertainty, thus, accurate estimates of these parameters are crucial. Errors in arrival time picking depend on the signal-to-noise ratio at the station and the arrival properties. Here, we employ an amplitude based bootstrapping approach to approximate a data driven pick uncertainty for infrasound arrivals. To account for simplistic atmospheric models, estimates of travel time variations using celerity predictions based on ground truth locations are used to obtain phase dependent model uncertainties as a function of distance. The resulting event locations are then statistically compared by source, arrival, and instrument type to demonstrate the applicability of seismoacoustic event location analysis under different circumstances. [SNL is managed and operated by NTESS under DOE NNSA Contract DE-NA0003525.]

10:10

3aPA7. Progress toward an acoustic point to point ray tracing method in a stratified atmosphere with wind. Stephanie L. Heath (Aeroacoustics Branch, NASA, MS 463, 2 N. Dryden St., Hampton, VA 23681, stephanie.l.heath@nasa.gov) and Ethan Cobb (Aeroacoustics Branch Internship, Universities Space Res. Associate, NASA, New York, NY)

NASA is investigating acoustic propagation capabilities to accommodate a point to point propagation prediction scheme defining a ray between a specific source–observer pair in an inhomogeneous stratified atmosphere with wind. A 4D space-time ray method based on the null geodesic equations yields two main benefits over classical 3D Euclidian reduced ray equations: (1) it poses a valid two point boundary value problem for the rays connecting two locations; and (2) the solution method is valid throughout the whole domain including at ray turning points without the need for piecewise definitions. The resulting two point boundary value problem and the constant horizontal slowness vectors, which are a result of the stratified medium assumption, allow the desired (t, x, y, z) ray trace. The presentation highlights the equations and inputs required to solve the 4D ray trace. The solution to this two point boundary value problem will create the groundwork for efficient future NASA propagation codes.

10:25

3aPA8. A Midsummer Flights' Dream: Balloon-borne infrasound-based aerial seismology. Léo Martire (NASA Jet Propulsion Lab., California Inst. of Technol., 306 N Sierra Bonita Ave., Pasadena, CA 91106, leo.martire@outlook.com), Siddharth Krishnamoorthy, Attila Komjathy (NASA Jet Propulsion Lab., California Inst. of Technol., Pasadena, CA), Daniel Bowman (Sandia National Labs., Albuquerque, NM), Michael Pauken (NASA Jet Propulsion Lab., California Inst. of Technol., Pasadena, CA), Jamey Jacob, Brian R. Elbing, Emalee Hough, Zach Yap, Molly Lammes, Hannah Linzy, Zachary Morrison, Taylor Swaim, Alexis Vance, Payton Simmons (Oklahoma State Univ., Stillwater, OK), and James Cutts (NASA Jet Propulsion Lab., California Inst. of Technol., Pasadena, CA)

Aerial seismology (AS) consists in detecting and characterizing atmospheric infrasound induced by ground displacements. Several studies have demonstrated that seismo-acoustic infrasound is highly similar to the seismic ground motion, making it a viable alternative for seismology on planetary bodies where deploying seismometers is technologically challenging. Until recently, such experiments were restricted to artificially-generated seismic events. However, on July 22, 2019, Brissaud *et al*. detected for the first time the infrasonic signature of a magnitude 4.2 earthquake from a high-altitude solar-heated balloon. In order to further demonstrate the feasibility of AS, additional detections are needed. Using similar balloons and payloads, stratospheric flights were launched several times per week during the summer of 2021 over Oklahoma's seismogenic zone. In this work, we present this campaign and its initial results. We describe the flight system, instrumentation, and details of the campaign design. We focus on the possible detections of earthquakes during the campaign using numerical simulations. For selected events, we present the balloon pressure data and extract potential arrivals. We perform full-waveform simulations using SPECFEM2D-DG to confirm those arrivals. We conclude with statistical arguments for earthquake detections and recommendations for future flights. Earthquakes are numerous in Oklahoma, but are of low and decreasing magnitudes. Nonetheless, this campaign's dataset is unique for the development of AS.

10:40

3aPA9. Seismoacoustic modeling of a below-ground explosion using finite element analysis. Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., PO Box 1663, M/S F665, Los Alamos, NM 87545, pblom@lanl.gov)

The generation of seismic and acoustic energy by below-ground explosions is a complicated, inelastic process in which damage and failure dynamics near the source and at the rock-air interface have a significant impact on the resulting seismoacoustic wavefield. Recent investigations by Blom *et al*. (2020) have demonstrated that combining the Rayleigh integral with a parametric spall model describing the ground motion characteristics for a given emplacement (i.e., depth, yield, and geology characteristics) results in accurate predictions for acoustic signals produced by below-ground explosions. In order to supplement this parametric, physics-based model, numerical simulations have been used to investigate the seismic and acoustic signals produced by a below-ground explosion using finite element methods including material crush at the source and tensile failure at the surface. The aim of these simulations is to enhance our understanding of the seismoacoustic signal generation by this complicated spatially and temporally extended source and investigate how acoustic and seismic observations can be leveraged to characterize below-ground explosions. An overview of the finite element simulations and the various challenges involved in their execution will be presented along with preliminary results.

10:55

3aPA10. PLetma—Parallel software package for intensive infrasound simulation. Codor Khodr (Laboratoire de Mécanique des Fluides et d'Acoustique, UMR 5509, École Centrale Lyon, 36, Ave., Guy de Collongue, Écully 69130, France, codor.khodr@ec-lyon.fr), Didier Dragna, Philippe Blanc-Benon (Laboratoire de Mécanique des Fluides et d'Acoustique, UMR 5509, École Centrale Lyon, Ecully, France), Régis Marchiano (Inst. Jean le Rond d'Alembert, UMR 7190, Sorbonne Univ., Paris, France), Ludovic Aubry, Olaf Gainville, and Christophe Millet (CEA, DAM, Bruyères-le-Châtel, France)

The Plateform of the Laboratory of Studies and Modeling in Acoustics (PLetma) is a suite of scientific softwares developed by the partner institutions of LETMA since 2015. Currently under development, it contains four different models, namely 3D ray-tracing, 3D non-linear one-way equation, 3D normal modes and 2D direct Navier–Stokes, as well as user interfaces for model execution and data visualization. The aim of this package is to provide a set of complementary tools for the modeling of broadband infrasound signals in a moving inhomogeneous atmosphere. It allows to cover a large panel of low-frequency propagation problems arising in geophysics or engineering. Implemented with MPI and OpenMP, the package relies on parallelism to improve computational efficiency of simulations. PLetma provides the infrastructure for incorporation of existing propagation and variability modeling capabilities. Comparisons between the four methods have been successfully undertaken on a test case [Sabatini *et al*., *JASA* (2016)]. Sensitivity analyses have been conducted to understand the effect of environmental parameters on modeling of long-range infrasound propagation. [This work was conducted within the framework of LETMA (Laboratoire ETudes et Modélisation Acoustique), a Contractual Research Laboratory shared between CEA, CNRS, Ecole Centrale de Lyon, C-Innov, and Sorbonne Université.]

11:10

3aPA11. Three-dimensional parabolic equation for infrasound propagation above topography. Codor Khodr (LMFA, École Centrale Lyon, 36, Ave. Guy de Collongue, Écully 69130, France, codor.khodr@ec-lyon.fr), Mahdi Azarpeyvand (Aerosp. Eng., Univ. of Bristol, Bristol, United Kingdom), and David Green (AWE Blacknest, Brimpton, United Kingdom)

The parabolic equation (PE) is one of the most popular methods for the modeling of low-frequency sound in the atmosphere. Over the past decades, considerable efforts have been dedicated to the inclusion of medium inhomogeneities, such as wind and turbulence, extending the PE to more realistic atmospheric conditions. On the other hand, few models take topography into account, while its effects on infrasound propagation are important in some cases. A new development has enabled the inclusion of irregular terrain in the narrow-angle 3DPE for a moving inhomogeneous medium [Khodr *et al*., *JASA* (2020)]. A coordinate transform similar to the Beilis-Tappert mapping used in 2D [Parakkal *et al*., *JASA* (2012)] is introduced to account for the variation in topography. The numerical solution relies on a Crank–Nicolson implicit scheme along the propagating direction. The pressure field at every step is computed with an iterative fixed-point algorithm which consists of a succession of tridiagonal systems that can be efficiently solved. A validation is performed against 3D Boundary Element simulations for propagation above a Gaussian hill in a homogeneous atmosphere. The existence of transversal scattering in the shadow zone, not represented by 2D models, is highlighted. Further developments include the extension of the method to a split-step wide-angle formulation. [This work was conducted at the University of Bristol in partnership with AWE Blacknest.]

11:25

3aPA12. Atmospheric parabolic equation with Galerkin discretization and boundary fitted grid for modeling infrasound propagation over topography. Gordon M. Ochi (Construction Eng. Res. Lab., U.S. Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil) and Michelle E. Swearingen (Construction Eng. Res. Lab., U.S. Army ERDC, Champaign, IL)

Studies utilizing the parabolic equation (PE) to model infrasound propagation over topography have traditionally implemented them as 2D or Nx2D PE models. These methods, to one extent or another, trade-off accuracy for enhanced computational speed. Three-dimensional parabolic equation (3DPE) models are seldom seen in atmospheric acoustic propagation studies compared to their 2D and Nx2D counterparts, despite the fact that theta coupling and proper modeling of horizontal refraction and diffraction in a 3DPE model should enable greater accuracy. In this work a 3DPE in cylindrical coordinates is developed under the alternating direction implicit scheme, with a Galerkin discretization and boundary fitted grid, such that the PE is capable of handling arbitrary and irregular topography features. Finally, 2D, Nx2D, and 3D implementations of the aforementioned PE model are examined for cases of propagation over a simple hill in homogeneous, downward refracting, and upward refracting atmospheres. Insights are drawn with regards to the accuracy at a variety of positions and ranges, and the computational speeds of the various methods.

11:40

3aPA13. Simulations of local and regional scale infrasound propagation in the Arctic. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Michael J. Shaw, Vladimir Ostashev, Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Michelle E. Swearingen (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Sarah McComas (Geotechnical and Structures Lab., U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

High-resolution, three-dimensional reconstructions of the Arctic atmosphere (the Arctic System Reanalysis version 2) are employed as input to wide-angle parabolic equation calculations to simulate infrasound propagation through Arctic weather conditions. The calculations involve horizontal distances out to 200 km (i.e., local and short regional scales), for which interactions with the troposphere and lower stratosphere dominate. The atmospheric phenomena examined include sudden stratospheric warmings (SSWs), strong polar lows, and boundary-layer phenomena such as low-level jets (LLJs), katabatic winds, and surface-based temperature inversions. The SSWs and polar lows are found to be opposites from an acoustic refraction standpoint: the former weaken upward refraction by temperature gradients in the troposphere, thus creating strongly asymmetric wind-dominated refraction, whereas the latter enhance negative temperature gradients in the troposphere, thus leading to prevailing upward refraction conditions. The LLJs and surface-based inversions are found to create particularly strong surface ducting conditions. Horizontal variations in the atmospheric profiles, in response to changes in topography and surface property transitions such as ice boundaries, significantly impact the propagation.

Session 3aPP

Psychological and Physiological Acoustics and Speech Communication: Acoustic Outreach to Early Career Scientists in Clinical and Physiological Research: Top-Down Influences on Auditory Processing

Bonnie K. Lau, Cochair
University of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195

Andrew R. Dykstra, Cochair
University of Miami, 1251 Memorial Dr., Coral Gables, FL 33146

Chair's Introduction—8:25

Invited Papers

8:30

3aPP1. Top-down and bottom-up expectations in the prefrontal and auditory cortices. Lalitta Suriya-Arunroj (National Primate Res. Ctr. of Thailand, National Primate Res. Ctr. of Thailand, Chulalongkorn Univ., Huai Haeng, Kaeng Khoi District, Saraburi 18110, Thailand, lalitta.suriya.arunroj@gmail.com), Joshua I. Gold (Dept. of Neurosci., Univ. of Pennsylvania, Philadelphia, PA), and Yale E. Cohen (Dept. of Otorhinolaryngology-Head and Neck Surgery, Univ. of Pennsylvania School of Medicine, Philadelphia, PA)

Auditory perceptual decision-making is modulated by interactions between bottom-up (sensory-driven) and top-down (expectation-driven) processes. Despite the importance of these interactions, little is known about their underlying neural mechanisms. We investigated these mechanisms by recording neural activity in the auditory and prefrontal cortices of rhesus monkeys while they performed a challenging auditory decision-making task. The monkeys decided whether the last ("test tone") in a sequence of tone bursts, embedded in broadband noise, was low or high frequency. Task difficulty was titrated by varying the sound level of the test tone relative to the noisy background. Bottom-up expectations were manipulated by presenting three identical low- or high-frequency tone bursts ("pre-tones"), establishing sequence regularity. Top-down processing was manipulated by presenting a visual cue that indicated the prior probability that the subsequent test-tone would be high or low frequency ("pre-cue"). The monkeys' behavioral choices and response times were biased by both the pre-tones and the pre-cues, with stronger and more consistent effects by the pre-cues. Neural activity was modulated preferentially by the pre-tones in auditory cortex and by the pre-cues in prefrontal cortex. These findings imply functional segregation between bottom-up and top-down processing in the primate brain during auditory perceptual decision-making.

8:50

3aPP2. The effect of mild dementia on speech perception in quiet and noise. Kate McClannahan (Otolaryngol., Washington Univ. School of Medicine, 1 Brookings Dr., Campus Box 1125, St. Louis, MO 63130, k.mcclannahan@wustl.edu), Amelia Mainardi, Austin Luor (Otolaryngol., Washington Univ. School of Medicine, St. Louis, MO), Yi-Fang Chiu (Speech, Lang., and Hearing Sci., St. Louis Univ., St. Louis, MO), Mitchell Sommers (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO), and Jonathan Peelle (Otolaryngol., Washington Univ. School of Medicine, St. Louis, MO)

Difficulty understanding speech, particularly in background noise, is a common complaint of older adults. In quiet, speech perception is often assumed to be relatively automatic. In background noise, however, additional cognitive abilities likely play a more substantial role in successful communication (Koeritzer *et al*., 2018; Rönnberg *et al*., 2013). We used a word identification task in quiet and two levels of speech-shaped noise (+6 and +3 SNR), to examine the impact of mild dementia on speech perception. In addition to noise, the number of similar sounding words (i.e., number of neighbors) also influences spoken word recognition — words with high neighborhood density (many competitors) are more difficult to perceive compared to low-density words (Luce and Pisoni, 1998). We predicted that individuals with dementia would perform significantly worse in conditions with the highest cognitive demands (high-density words, least favorable SNR). Preliminary results revealed that identification scores were significantly lower for conditions with more background noise and that, even after controlling for hearing loss, individuals with dementia had significantly reduced performance in all conditions. These results suggest that after accounting for differences in hearing, individuals with mild dementia still struggle with speech understanding, even in the absence of background noise.

9:10

3aPP3. The neural processing of sound in infants. Bonnie K. Lau (Otolaryngology-HNS, Univ. of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195, blau@uw.edu), Samu Taulu (Phys., Univ. of Washington, Seattle, WA), Patricia K. Kuhl, and Adrian KC Lee (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Early auditory abilities play an important role in infants' ability to acquire language, appreciate music, and navigate the complex acoustic environments around them. However, the neural mechanisms that support infant sound processing are not well understood. This study used magnetoencephalography (MEG) with recent advancements in movement compensation to obtain functional measures of auditory processing in awake infants. MEG responses were recorded longitudinally at 3, 6, and 11 months of age to an English bi-syllabic word and an amplitude-modulated complex tone as a non-speech but spectro-temporally complex control stimulus. MEG recordings were made with an Elekta Neuromag® whole-head 306-channel MEG system. The neural generators of the MEG signals were determined using an equivalent current dipole (ECD) model. High quality MEG data with good signal-to-noise ratios were recorded in infants as young as 3 months of age in both conditions. ECD field maps showed focal sources in the brain confirming the ECD model is a robust source localization approach that can fit auditory neural sources in infants. Our results show that dipole modeling of MEG signals in combination with advanced movement compensation offers a temporally precise method of investigating how the cortical processing of sound changes across the first year of life.

9:30

3aPP4. Transformation of acoustic information to sensory decisions in parietal cortex. Justin Yao (Ctr. for Neural Sci., New York Univ., 4 Washington Pl., Fl. 6 Rm. 621, New York, NY 10003, jdyao@nyu.edu) and Dan Sanes (Ctr. for Neural Sci., New York Univ., New York, NY)

Integrating sensory information over time is fundamental for guiding perceptual decisions. This process is facilitated by the transformation of stimulus representations downstream of sensory cortex. Here, we ask how the neural representations of auditory information are transformed in the auditory-recipient parietal cortex, a region that is causally associated with sound-driven perceptual decisions. Wireless recordings of neural activity were conducted in parietal cortex while gerbils performed an alternative forced-choice (AFC) auditory temporal integration task. Gerbils were required to discriminate amplitude modulated (AM) noise at 4 vs 10 Hz across various signal durations (100–2000 ms). Task performance was poor at short AM durations (100–300 ms), improved with longer durations, and reached an optimum at $\geq$600 ms. During task performance, individual neurons displayed a rise in firing rate that peaked prior to the animals' decision. Population activity from simultaneously recorded parietal cortex neurons was strongly correlated with behavior outcomes, leaving open whether parietal representation was purely motor-related or included a sensory component. Thus, we evaluated population activity during disengaged sessions, and found a representation of the acoustic stimulus. Our findings suggest that parietal cortex neurons represent both the acoustic stimulus and sound-driven decision, demonstrating the progression of auditory processing downstream of auditory cortex.

9:50–10:05 Break

10:05

3aPP5. Auditory figure-ground segregation is impaired by aging and age-related hearing loss. Adam Boncz (Inst. of Cognit. Neurosci. and Psych., RCNS Budapest, 2 Magyar Tudosok Korutja, Budapest 1117, Hungary, boncz.adam@ttk.hu), Orsolya Szalárdy (Inst. of Behavioural Sci., Semmelweis Univ., Budapest, Hungary), Péter Velősy (Dept. of Cognit. Sci., Budapest Univ. of Technol. and Economics, Budapest, Hungary), Istvan Winkler, and Brigitta Tóth (Inst. of Cognit. Neurosci. and Psych., RCNS Budapest, Budapest, Hungary)

Listening in a noisy environment relies on our ability to extract structure from noisy sensory input while segregating it from the rest of the scene (Figure-Ground Segregation—FGS). To assess whether FGS is impaired by aging, source localized brain activity was analyzed from young, older adults, and hearing-impaired (HI) elderly during object detection in noise. Listeners' task was to report whether they detected a figure (rising sound stream) within either high or low noise. Individualized figure and noise levels allowed us to compare groups across the same performance levels (65% vs 85% accuracy). HI group was impaired in integrating sounds of the figures. Figures elicited an early sensory processing-related (object-related negativity- ORN) and a later perceptual decision-related (P600) brain response. In the elderly, compared to the young the ORN was delayed in general and smaller in the inferior frontal gyrus (IFG). P600 activity was generated within a less distributed network in both the elderly and HI relative to the young. In the HI, compared to the elderly the ORN was smaller in the auditory cortex and IFG. Our results suggest that early FGS processes are disturbed only neurally in the elderly and overtly in HI.

10:25

3aPP6. How, when, and where predictions combine with speech in auditory cortexd. Ediz Sohoglu (School of Psych., Univ. of Sussex, Pevensey Bldg., Brighton, East Sussex BN1 9QH, United Kingdom, E.Sohoglu@sussex.ac.uk)

Human speech perception is rapid, accurate and robust to noise. These characteristics are enabled by top-down predictive computations that combine bottom-up sensory input with prior expectations. Here, I distinguish between two accounts of how predictions combine with speech signals in auditory cortex. Is expected speech content enhanced to leave a "sharpened" version of the sensory input? Or is expected content subtracted away to leave only those parts that are unexpected, i.e., "prediction error"? Based on fMRI and MEG data, I show that neural representations of spoken words are suppressed when listeners have strong predictions based on long-term lexical knowledge. This neural signature is uniquely consistent with the prediction error account, occurs in superior temporal cortex within 400 ms of speech input and in the delta/theta frequency band. These results provide a comprehensive window into predictive processing of speech, showing how, when and where predictions combine with sensory signals.

10:45

3aPP7. Laminar specificity of the auditory awareness negativity under multitone masking: A biophysical modeling study. Carolina Fernandez Pujol, Elizabeth Blundon (Biomedical Eng., Univ. of Miami, Coral Gables, FL), and Andrew R. Dykstra (Biomedical Eng., Univ. of Miami, 1151 Richmond St., Western Interdisciplinary Res. Bldg., London, ON N6A 3K7, Canada, andrew.r.dykstra@gmail.com)

Electroencephalography (EEG) and magnetoencephalography (MEG) are excellent mediums for capturing human neural activity on a millisecond time scale, yet little is known about their underlying laminar and biophysical basis. Here, we used a reduced but realistic cortical circuit model—Human Neocortical Neurosolver (HNN)—to shed light on the laminar specificity of brain responses associated with auditory conscious perception under multitone masking. HNN provides a canonical model of a neocortical column circuit, including both excitatory pyramidal and inhibitory basket neurons in layers II/III and layer V. We found that the difference in event-related responses between perceived and unperceived target tones could be accounted for by additional input to supragranular layers arriving from either the non-lemniscal thalamus or cortico-cortical feedback connections. Layer-specific spiking activity of the circuit revealed that the additional negative-going peak that was present for detected but not undetected target tones was accompanied by increased firing of layer-V pyramidal neurons. These results are consistent with current cellular models of conscious processing and help bridge the gap between the macro and micro levels of analysis of perception-related brain activity.

11:05–11:30
Panel Discussion

WEDNESDAY MORNING, 1 DECEMBER 2021   501 (L)/504 (O), 8:20 A.M. TO 11:50 A.M.

Session 3aSA

Structural Acoustics and Vibration, Engineering Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Experimental Methods for Material Characterization in Structural Acoustics and Vibration

Stephanie G. Konarski, Cochair
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Benjamin S. Beck, Cochair
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Colby W. Cushing, Cochair
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Kyle S. Spratt, Cochair
Applied Research Laboratories, The University of Texas at Austin, Austin, TX 78713

Chair's Introduction—8:20

Invited Papers

8:25

3aSA1. Experimental characterizations of band gaps in 3D printed phononic materials. Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801, kmatlack@illinois.edu)

Phononic materials contain spatial periodicity of material properties or geometry, which cause band gaps and other unnatural wave phenomena to form. While these behaviors have been well studied theoretically and experimentally in representative systems, the growth of 3D printing caused a surge of phononic materials to be physically realized and studied in the last several years. Since 3D printing enables almost any geometry to be fabricated in a variety of materials, it opens the possibility for band gaps to be used for frequency-targeted vibration mitigation designed directly into structural components. This talk presents our recent work on 3D printed phononic materials with tailored vibration mitigation, focusing on experimental characterizations of band gaps. Two phononic materials will be discussed: (1) lattice-resonator metastructures, that combine a 3D printed lattice with embedded masses to form band gaps that

depend on lattice geometry; and (2) radial phononic materials, that combine radially-dependent properties with periodicity to form band gaps for radially-propagating torsional waves. Experimental intricacies will be highlighted, including material damping, fluid–structure interactions within the band gap, and isolating modes and wave polarizations. Such thorough experimental characterizations of band gaps in phononic materials enable the development of robust and predictive modeling and design capabilities.

8:45

3aSA2. Ultrasonic characterization of cold sintered ZnO components with varying relative densities. Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., 212 Earth-Engr Sci. Bldg, University Park, PA 16802, aza821@psu.edu), Susan Trolier-McKinstry, and Elizabeth Trautman (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

The cold sintering process is a low temperature processing technique used to synthesize metals, ceramics, and composites. Some cold sintered parts do not reach the mechanical properties of conventionally sintered parts even at high relative densities. These diminished mechanical properties are presumed to be in part a result of either in inhomogeneous removal of transient liquid phases or inability to reach the proper cold sintering temperature/pressure everywhere in the part. In this presentation, we evaluate the spatial variability in the properties of cold sintered ZnO parts with decreasing relative densities through ultrasonic testing. The presence of homogeneously distributed porosity is assessed by ultrasonic wave speed, while macro-flaws larger than the average pore size are assessed through ultrasonic attenuation. The results are compared to the response of conventionally sintered ZnO samples with densities ranging from 86% to 89%. Conventionally sintered and cold sintered parts exhibited different slopes in the decrease of wave speed with density. This difference suggests microstructural discrepancies beyond porosity in cold versus conventionally sintered ZnO components. Attenuation results suggested the presence of macro-flaws even in the samples with highest relative density. These results underscore the importance of non-destructive characterization of the cold sintered components.

9:05

3aSA3. Resonant ultrasound spectroscopy for anisotropic materials with misaligned geometric and material axes. Luke Beardslee (Los Alamos National Lab, P.O. Box 1663, Los Alamos, NM 87545, lbeardslee@lanl.gov), Marcel Remillieux (Geophys., Los Alamos National Lab., Los Alamos, NM), and Timothy J. Ulrich (Los Alamos National Lab, Los Alamos, NM)

Resonant ultrasound spectroscopy (RUS) is a powerful non-destructive technique used to measure the full elastic tensor of a material. The original formulation of this technique has restrained its application to relatively simple situations in which the sample has a trivial geometry (e.g., cube, cylinder, sphere), is made of one material only, with perfect alignment of material and geometric axes. In many practical applications, these ideal conditions cannot be met. Recently, RUS has been enhanced by the authors to handle these more complex systems by combining the finite-element method with a genetic algorithm. The capabilities of this new RUS approach are demonstrated experimentally using both a non-uniform hemisphere and a cylindrical sample made of compressed particles. The particular manufacturing process of the cylindrical part results in a sample with anisotropy and a misalignment of the geometric and material axes. Although the sample has a simple geometry it consist of a complex material. Confidence in the RUS inversion process is assured by a careful comparison of the mode shapes measured experimentally and simulated with the proper material orientation and elastic tensor.

Contributed Papers

9:25

3aSA4. Abstract withdrawn.

9:40

3aSA5. Audio-frequency characterization of cranial sutures via non-contact vibration experiments. Eetu Kohtanen (G.W. Woodruff School of Mech. Eng., Georgia Tech, Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332, ekohtanen3@gatech.edu), Matteo Mazzotti, Massimo Ruzzene (Paul M. Rady Dept. of Mech. Eng., CU Boulder, Boulder, CO), and Alper Erturk (G.W. Woodruff School of Mech. Eng., Georgia Tech, Atlanta, GA)

There are many factors that affect the structural dynamics of the human skull, including but not limited to skull-specific geometry, the composite nature of cranial bone including the cortical tables and the diploë, and the presence of sutures, with the latter being the focus of this study. Sutures are joints between the cranial bones that tend to decrease the overall bone strength in three-point flexural tests, but their effects on the dynamic properties of the bone including mode shapes and natural frequencies remain largely unexplored. Knowledge of the suture dynamics in human skulls is relevant to various research domains spanning from hearing by bone conduction to cranial wave propagation, and an accurate representation of the cranial sutures can be important to establish high-fidelity skull models to use in such applications. To this end, non-contact vibration experiments are carried out to extract the mode shapes, natural frequencies, and damping ratios of several skull segments containing sutures (specifically the coronal, sagittal, lambdoid, and lateral sutures). Furthermore, high-fidelity finite element models of the bone segments are constructed from fine resolution (50 μm) computed tomographic scans. These numerical models are employed to identify the appropriate suture elastic properties that most closely replicate the experimental results.

9:55–10:10 Break

Invited Papers

10:10

3aSA6. Flexural wave parameter measurements of viscoelastic panels using a laser doppler correlation vibrometry. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu) and Max Miller (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Characterizing broadband dynamic properties of viscoelastic panel materials represents a challenging task, particularly when the frequency band of interest needs to be extended toward low frequency range. This work applies a laser Doppler vibrometer in measuring flexural wave properties of the viscoelastic panel materials. The measurement relies on a time-domain broadband correlation technique which experimentally measures impulse responses at two points radially away from a flexural wave exciter. Using reasonable sized panel samples, boundary conditions of the measurements present insignificant obstacles towards high frequency range, since a time-domain windowing facilitates extraction of direct wave components. While boundary reflections will still obscure measurement results towards low frequencies. This work examines a possible mitigation approach by rotating the sample panel, the coherence of direct wave pulses measured equidistant from the exciter and the lack thereof disturbing boundary reflections enables largely removing the undesirable reflections. In this way angular averaging significantly reduces boundary reflections. This paper discusses experimental investigations on the characterization method of flexural wave parameters including wave speed, flexural modules and loss factor.

10:30

3aSA7. Measurement of frequency-dependent shear properties of highly compliant materials using Kibble's method. Nicholas Vlajic (Penn State Appl. Res. Lab., P.O. Box 30, State College, PA 16803, nav5000@psu.edu), Benjamin Beck, and Christopher Coughlin (Penn State Appl. Res. Lab., State College, PA)

We present a method to determine the frequency-dependent properties of highly-compliant materials by measuring the response of rod-like specimens subjected to an applied torque. Kibble's method, which utilizes a velocity measurement and two electrical measurements, is used to calculate the applied torque on the specimen as a function of frequency. Shear modulus and loss factor in shear are determined by fitting mechanical models to transfer functions between the applied torque and torsional response of the sample. We demonstrate this technique on a prototype apparatus using different polymer materials and specimen geometry. Estimates of the shear modulus and loss factor in shear are determined over a range of frequencies that include the first six torsional resonant modes. These material properties are compared with data obtained using a commercial Dynamic Material Analyzer wherein the samples are excited in a bending mode.

Contributed Papers

10:50

3aSA8. Development of a water-filled four-sensor impedance tube. Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX, mcneese@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman, and Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Four-sensor impedance tubes are readily utilized for characterization of the acoustic properties of materials in a gaseous host medium, as described in the standard ASTM E2611-19. Similar acoustic characterization is desired for materials submersed in water. Previous studies have documented the challenges of performing measurements in water-filled impedance tubes due to interaction with the elastic tube wall and difficulty in obtaining a suitable end-termination condition. Building upon previous demonstration of a two-sensor water-filled impedance tube [*J. Acoust. Soc. Am.* **113**, 3245 (2003)], this paper discusses the development and demonstration of a four-sensor water-filled impedance tube with an anechoic termination. The technique utilizes two sensors on each side of the test section to simultaneously characterize both the incident and transmitted standing wave field and extract the complex impedance and wavenumber from the test sample. Discussion will focus on the challenges of performing such measurements in a water-filled tube along with methods for overcoming these challenges. Validation measurements of homogeneous materials in the 1–10 kHz band will be presented to demonstrate that both the complex impedance and wavenumber of a material submersed in water can be extracted using this approach. [Work supported by ONR.]

11:05

3aSA9. Resonator measurements and modeling of the low-frequency sound speed of seagrass leaves. Nicholas A. Torres (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 301 E Dean Keeton St., Austin, TX 78712, natorres@utexas.edu), Nathan G. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Megan S. Ballard, Kevin M. Lee (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), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Previous investigations of seagrass have revealed acoustic phenomena that depend on the material properties of the plant tissue and gas contained within the lacuna of the leaves. However, the application of predictive models to describe these observations has been limited to effective medium models that neglect both the effects of elastic properties of seagrass tissue as well as its structure. To address this, we developed a physiological model of a seagrass leaf that includes the epidermis, a continuous layer on the surface of the leaf that provides rigidity, and the aerenchyma, a soft plant tissue containing the lacuna. To validate the seagrass leaf model, acoustic resonator experiments were performed to measure the low-frequency sound speed of seagrass leaves, and the physical dimensions of the epidermis and aerenchyma were assessed through the examination of microscopic images of leaf cross sections. Using this imagery, a finite element model of the seagrass leaves was created, and COMSOL was used to simulate the resonator experiment. The comparison of the measured and modeled acoustic data provides insights into the parameters of the seagrass tissue that affect the

acoustic response, specifically the shear moduli of the epidermis and aerenchyma. [Work supported by ARL IR&D and ONR.]

11:20

3aSA10. Attenuation of external vibrations transmitted into platforms by granular media for special applications. Hasson M. Tavossi (Dept. of Eng. Technol., College of Sci. & Technol., Savannah State Univ., 219 College St., m3219 College St., Savannah, GA 314o4, tavossih@savannahstate.edu)

Mitigation of mechanical vibrations from external sources into structures by granular media is investigated to prevent transmission into the structure by surrounding granular media layers. Mechanical properties of granular media can be selected for absorption of specific frequency band of external vibrations. The absorption frequency band of the granular media can be move by design to required frequency band where scattering and dissipation of vibrational energy is maximized. Samples of granular media layers are subjected to external vibrations of different intensity and frequency. Experimental values are presented for attenuation versus frequency for different layers of selected grain sizes, at audible frequencies. The goal is to determine media properties that maximize vibration attenuation for special applications of vibration free platform for such specialized application as supports for atomic force microscopy supports and more recent gravitational-wave detectors.

11:35

3aSA11. Visco-elastic material characterization by means of full field Lamb waves. Adil H. Orta (Wave Propagation and Signal Processing (WPSP), Dept. of Phys., KU Leuven, Etienne Sabbelaan 53, Kulak Kortrijk Campus, Kortrijk 8500, Belgium, adilhan.orta@kuleuven.be), Joost Segers (Mech. of Mater. and Structures (MMS), Dept. of Mater., Textiles and Chemical Eng., Ghent Univ., Gent, Belgium), Nicolaas B. Roozen (Lab. Acoust., Div. of Soft Matter and Biophys., Dept. of Phys. and Astronomy, KU Leuven, Leuven, Belgium), Wim Van Paepegem, Mathias Kersemans (Mech. of Mater. and Structures (MMS), Dept. of Mater., Textiles and Chemical Eng., Ghent Univ., Gent, Belgium), and Koen Van Den Abeele (Wave Propagation and Signal Processing (WPSP), Dept. of Phys., KU Leuven, Kortrijk, Belgium)

In structural design and structural health monitoring (SHM), the identification of the visco-elastic stiffness tensor of composite materials is considered fundamental, however, the reconstruction of the stiffness coefficients is not straightforward due to the inherent presence of anisotropy and heterogeneity. In the current study, an inverse procedure is proposed based on full field wave dispersion curves to characterize the (18) visco-elastic stiffness parameters of an orthotropic composite plate. The procedure is illustrated by means of finite element method simulations conducted with broadband signals. First, the FEM simulation data, consisting of the in-plane and out-of-plane velocities on the 2D top surface of a composite plate, were converted into the frequency-complex wavenumber domain, including information on elasticity and viscosity, by using the matrix pencil method. The converted data set was then considered as input to a surrogate optimization inversion algorithm, using the semi analytical finite element method (SAFE) as the forward model owing to its accuracy, efficiency and robustness. The full tensorial visco-elastic stiffness parameters were calculated and updated by minimizing the error between the simulated and the calculated complex wavenumbers. The final reconstructed stiffness properties in case of an aluminum plate and of a unidirectional composite laminate show good agreement with the exact input stiffness values, with average errors below 5% for all 18 visco-elastic stiffness parameters.

Session 3aSC

Speech Communication: Articulation and Voice (Poster Session)

Sarah Harper, Chair
Linguistics, University of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

3aSC1. An ultrasound study of rest position and pre-acoustic articulation in adults and children. Pertti Palo (Speech & 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)

The position of the tongue during rest and its relationship to subsequent pre-acoustic speech movements have shed light on speech motor planning. They may find applications in technologies such as silent speech interfaces. Palo ["Measuring pre-speech articulation," Ph.D. thesis (Queen Margaret University, Edinburgh, 2019)] found that the duration of pre-acoustic speech movements are strongly correlated with acoustic utterance duration, but it remains an open question whether this correlation is modulated by different rest positions. This study investigates variability in rest position among children and adults, and examines whether rest position affects pre-acoustic speech movements and their correlation with acoustic utterance durations. [This work has been supported by a personal grant from The Emil Aaltonen Foundation (Palo) and by NSF Grant No. 1551131 (Lulich).]

3aSC2. Articulatory and acoustic analysis of voice acting. Colette Feehan (Linguist, Indiana Univ., Bloomington, 107 s Indiana Ave., Bloomington, IN 47404, cmfeehan@indiana.edu)

Voice actors are an under-utilized population for linguistic study. Looking at voice actors' novel productions of typical speech sounds can tell us more about over which muscles we have volitional control. This can help test models of speech production, inform speech therapy, and create better pedagogy for teaching vocal performance. While there have been several studies analyzing the acoustics produced by voice actors (Teshigawara, 2001; 2003; 2004; 2009; 2011; Teshigawara *et al.*, 2007; Uchida, 2007) there are fewer studies that have investigated the articulation of voice acting (Teshigawara & Murano, 2004). This study used acoustic analysis, 3D ultrasound, and electroglottography to investigate the speech produced by six professional and amateur voice actors (2 female, 4 male). Measures of average fundamental frequency (F0) and F0 range were collected from each actor speaking in their regular adult voice and a simulated child voice. 3D ultrasound was used to look at tongue placement and horizontal kymograms of the ultrasound images were also used to infer hyoid bone movement. Additionally, acoustic estimates of vocal tract length were compared to electroglottography measurements of laryngeal raising for two of the subjects. This presentation will describe the within-subject variation across each actor's adult and simulated child voices.

3aSC3. Effect of vocal tract morphology on tongue shaping for American English /ɹ/. Yijing Lu (Linguist., Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, yijinglu@usc.edu), Haley Hsu, Louis Goldstein (Linguist., Univ. of Southern California, Los Angeles, CA), and Asterios Toutios (Elec. and Comput. Eng., Univ. of Southern California, Los Angeles, CA)

There is a lack of general agreement among previous studies (e.g., Bakst, 2016; Dediu & Moisik, 2019; Westbury *et al.*, 1998) on whether measurements of vocal tract morphology are robust predictors of inter-speaker variation in tongue shaping for American English /ɹ/. One possible reason is the different quantifications of /ɹ/ tongue shapes that were employed. The current study compares the relationships between a single set of anatomical measurements and three different measures of lingual articulation for /ɹ/ in /ɑɹɑ/ in midsagittal real-time MRI data. A novel method was developed to quantify the palatal constriction location and length, which served as the first two measures of tongue shape. A linear Support Vector Machine divided the constriction location and length measures into regions that approximate the visually identified categories of "retroflex" and "bunched." The third shape measurement is the signed distance of each token of /ɹ/ to the division boundary, representing the degree of "retroflexion" or "bunchedness" based on palatal constriction properties. These three measures showed marginally to moderately significant linear relationships with two specific measures of individual speakers' vocal tract anatomy: the degree of mandibular inclination and the length of the oral cavity roof. Overall, the effect of anatomy on the lingual articulation of /ɹ/ is not strong. [Work supported by NSF, Grant 1908865.]

3aSC4. Modeling speaker-specific vocal tract kinematics from gestural scores. Yijing Lu (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, yijinglu@usc.edu), Justin Ly, Shrikanth Narayanan (Elec. and Comput. Eng., Univ. of Southern California, Los Angeles, CA), Louis Goldstein (Linguist, Univ. of Southern California, Los Angeles, CA), and Asterios Toutios (Elec. and Comput. Eng., Univ. of Southern California, Los Angeles, CA)

The theory of Task Dynamics provides a method of predicting articulatory kinematics from a discrete phonologically-relevant representation ("gestural score"). However, because the implementations of that model (e.g., Nam *et al.*, 2004) have generally used a simplified articulatory geometry (Mermelstein *et al.*, 1981) whose forward model (from articulator to constriction coordinates) can be analytically derived, quantitative predictions of the model for individual human vocal tracts have not been possible.

Recently, methods of deriving individual speaker forward models from real-time MRI data have been developed (Sorensen *et al*., 2019). This has further allowed development of task dynamic models for individual speakers, which make quantitative predictions. Thus far, however, these models (Alexander *et al*., 2019) could only synthesize limited types of utterances due to their inability to model temporally overlapping gestures. An updated implementation is presented, which can accommodate overlapping gestures and incorporates an optimization loop to improve the fit of modeled articulatory trajectories to the observed ones. Using an analysis-by-synthesis approach, the updated implementation can be utilized: (1) to refine the hypothesized speaker-general gestural parameters (target, stiffness) for individual speakers; (2) to test different degrees of temporal overlapping among multiple gestures such as a CCVC syllable. [Work supported by NSF, Grant 1908865.]

3aSC5. Effects of phonotactic legality on gestural coordination in consonant clusters: An electromagnetic articulography study. Hung-Shao Cheng (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, hscheng@nyu.edu), Matthew Masapollo (Univ. of Florida, Gainesville, FL), Christina Hagedorn (English, College of Staten Island (CSI) CUNY, Staten Island, NY), and Adam Buchwald (Communicative Sci. and Disord., New York Univ., New York, NY)

Cross-language studies of speech production have shown that English speakers can produce phonotactically illegal onset fricative-nasal clusters (e.g., /fn/) with high accuracy based on acoustic analyses. However, it remains unclear whether the articulatory gestures affiliated with the fricative-nasal segments are produced with comparable gestural timing to native onset clusters (e.g., /sm/, /fl/). We here used electromagnetic articulography (EMA) to investigate whether the production of non-native /fn/ onset clusters exhibits a comparable amount of consonant-vowel gestural overlap to native onset clusters (i.e., /sm/ and /fl/) in nonwords. Previous articulatory investigations have demonstrated that native English onset clusters exhibit an increase in gestural overlap between the vowel-adjacent consonant and the vowel (e.g., SMAGDEEP) when compared to the corresponding singleton (MAGDEEP). We controlled for the vocal tract configuration (e.g., jaw position) by comparing each onset cluster to heterosyllabic sequences consisting of the same consonant sequences (e.g., /sm/ vs /s#m/). While the data collection is still ongoing, the preliminary results ($n = 3$) suggest that when the complexity of the onset increased, vowel-adjacent consonantal gestures showed greater temporal overlap with vocalic gestures for phonotactically legal sequences (SMAGDEEP versus MAGDEEP) but not illegal sequences (FNAGDEEP versus NAGDEEP).

3aSC6. Gestural timing and stability in hausa implosives. Miran Oh (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu), Louis Goldstein, and Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA)

Articulatory dynamics of glottalic consonants—ejectives and implosives—show strictly different intergestural timing patterns that those of pulmonic consonants. For example, oral constriction and vertical larynx gestures in Hausa have been shown to be sequentially produced for glottalic consonants and simultaneously produced for their pulmonic counterparts (Oh *et al*., ASA 2018, PaPE 2019). In the present work, we predict that glottalic consonants will exhibit tighter, more stable intergestural coordination than pulmonic consonants, due to the superordinate aerodynamic goal in glottalics—namely, that to generate a rapid oral air pressure change the vertical larynx action must begin during the oral closure interval. The current study examines data from three Hausa speakers producing voiced implosives and voiced plosives, using real-time MRI to obtain kinematic information on the oral constriction and the vertical larynx action. A region-of-interest analysis quantifies oral constrictions and a centroid analysis tracks vertical larynx actions. Findings confirm that intergestural timing patterns are contrastive between implosives and voiced stops. Moreover, results on relative timing stability, indexed by token-to-token variability and covariance relations between timing lags and gestural durations, indicate that the intergestural timing for implosives is more stable than for voiced stops. [Work supported by NIH and NSF.]

3aSC7. Human beatboxing: Physiological aspects of drum imitation. Alexis Dehais-Underdown (Laboratoire de Phonétique et Phonologie (UMR 7018 - CNRS/Sorbonne-Nouvelle), 19, rue des Bernardins, Paris 75005, France, alexis.dehais-underdown@sorbonne-nouvelle.fr), Paul Vignes (Laboratoire de Phonétique et Phonologie (UMR 7018 - CNRS/Sorbonne-Nouvelle), Paris, France), Lise Crevier-Buchman (Laboratoire de Phonétique et Phonologie (UMR 7018 - CNRS/Sorbonne-Nouvelle) & Service d'ORL et de Chirurgie Cervico-Faciale - Hôpital Foch, Paris, France), and Didier Demolin (Laboratoire de Phonétique et Phonologie (UMR 7018 - CNRS/Sorbonne-Nouvelle), Paris, France)

Human Beatboxing (HBB) is a musical technique produced with vocal tract movements. In this study we investigate HBB production mechanisms of 3 drums imitations, that is the labial kick drum, the lingual closed hi-hat and the velar ingressive snare drum, produced in isolation by 5 beatboxers. Based on aerodynamic (intraoral pressure and oral airflow), acoustic and laryngoscopic data, we were able to identify gestures. The kick was produced as a labial stop and the hi-hat as an coronal affricate; both were initiated by a glottalic egressive airstream. Differences in intraoral pressure, oral airflow and laryngeal configuration were observed across beatboxers. The snare was produced as a velar pulmonic ingressive affricate [↓k͡L̥]; however 1 beatboxer produced a velar glottalic ingressive stop + a velar pulmonic ingressive fricative [k↓L̥]. The combination of glottalic and pulmonic airstreams raises questions concerning the coordination of initiatory mechanisms and its planification. HBB differs from speech production because it is a musical system and thus do not rely on linguistic constraints such as phonotactic; it implies that articulatory patterns are not as restricted as in speech. Thus, it is a rich paradigm to investigate vocal tract physiology and mental representations of non-linguistic sounds.

3aSC8. Labiodentalization of bilabial stops in spontaneous english smiled speech. Linda X. Wu (Linguist, Univ. of British Columbia, 209-5760 Hampton Pl., Vancouver, BC V6T 2G1, Canada, lindaw0207@gmail.com), Yadong Liu, Charissa Purnomo (Linguist, Univ. of British Columbia, Vancouver, BC, Canada), Gracellia Purnomo (School of Audiol. and Speech Sci., Univ. of British Columbia, Vancouver, BC, Canada), Melissa Wang, and Bryan Gick (Linguist, Univ. of British Columbia, Vancouver, BC, Canada)

The co-occurrence of facial expressions and speech production presents a conflict of opposing muscular activation. Existing English research indicates a pattern of bilabial stops being resolved as labiodentals during smiled speech. Electromyographic data suggests that suppression is imposed on one muscle force to resolve the smiling-lip closure conflict [Liu *et al*., in ISSP Proceeding (2020), pp. 130–133]. These results have limited generalizability as data was collected from one speaker performing reading tasks under laboratory context. The present study continues the investigation of bilabial productions by investigating bilabial phonemes in spontaneous smiled speech. Natural speech samples from 16 native English speakers were extracted from YouTube interviews and vlogs. Bilabial phonemes in neutral and smiling conditions were analyzed for labiodentalization using facial imaging software. The intensity of Facial Action Units (FAU) "lip corner puller" and "lip tightener" during labial token productions were examined. Results show that labiodentalized variants of bilabial tokens also occurred during natural smiled speech. FAU intensity measurements indicate lower "lip corner puller" intensity during bilabial closures than labiodental ones when smiling. This suggests that smile suppression was observed when lip closure was prioritized, validating earlier laboratory conclusions that selective muscular suppression is utilized in the resolution of conflict between articulators.

3aSC9. Muscle strain analysis from diffusion tractography and dynamic magnetic resonance imaging of the moving tongue. Fangxu Xing (Radiology, Harvard Med. School, 55 Fruit St., Thiers Hall, Rm. 304A, Boston, MA 02114, fxing1@mgh.harvard.edu), Xiaofeng Liu, Tim Reese (Radiology, Harvard Med. School, Boston, MA), Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD), Van Wedeen (Radiology, Harvard Med. School, Boston, MA), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), Georges El Fakhri, and Jonghye Woo (Radiology, Harvard Med. School, Boston, MA)

Strain estimation of a deforming human tongue is an important task in motor control studies. Various magnetic resonance imaging (MRI) modalities provided versatile acquisition tools, facilitating accurate strain computation and quantitative analysis. We report a method that combines data from diffusion and dynamic MRI to yield strain values along internal tongue muscle fiber directions. Dynamic MRI provides spatiotemporal muscle motion. Diffusion tractography provides muscle fiber geometry. Since the two datasets exist in different physical spaces and tractography is susceptible to numerical errors if spatially transformed, we transformed only muscle motion fields from the dynamic MRI space to the diffusion MRI space using diffeomorphic registration. Strain tensors were then computed and projected onto the fiber directions. Masks were used to extract strain of individual muscles. Three tongue motions while pronouncing two utterances and making a forward protrusion were processed. Resulting strain patterns on seven internal muscles showed that some muscles stayed stretched or compressed over all motion time while other muscles switched between stretches and compressions depending on specific tongue shapes. The proposed method enables accurate assessment of variations in muscle strains, which potentially reveals muscle activation patterns and provides insights into the tongue's detailed mechanics in speech motor control.

3aSC10. Deep learning-based age estimation using B-mode ultrasound tongue imaging. Kele Xu (National Key Lab. of Parallel and Distributed Processing (PDL), 107, Yanwachi, Changsha 410073, China, kelele.xu@Gmail.com), Tamás G. Csapó (Budapest Univ. of Technol. and Economics, Budapest, Hungary), and Ming Feng (Tongji Univ., Shanghai, China)

The feasibility of age estimation is explored using the ultrasound tongue image of the speakers. Motivated by the success of deep learning, a deep convolutional neural network model is trained on the UltraSuite dataset. The deep model achieves mean absolute error (MAE) of 2.03 years for the data from typically developing children, while MAE is 4.87 for the data from the children with speech sound disorders, which suggest that age estimation using ultrasound is more challenging for the children with speech sound disorder. Also, we explore to visualize what does the deep model learn for the age estimation task. We firstly visualize the convolutional layers in the learned convolutional neural networks. We observe that the deep model not only focuses on the contour in the ultrasound tongue image, but also pays more attention to the regions corresponding to the tendon and tongue root regions, which may provide guidance for future ultrasound tongue imaging interpretation tasks. The developed method can be used a tool to evaluate the performance of speech therapy sessions.

3aSC11. Velopharyngeal opening in French: Revisiting velum configuration in nasal, oral, and rest intervals. Md Jahurul Islam (Linguist, The Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jahurul.islam741@gmail.com), Gillian de Boer, and Bryan Gick (Linguist, The Univ. of British Columbia, Vancouver, BC, Canada)

Velopharyngeal Opening (VPO) is associated with nasality in speech. Earlier studies on French suggested that there is a continuum of VPO: nasal vowels, nasal consonants, contextually nasalized vowels and, finally, oral segments. However, most of these studies used isolated words/syllables, very small numbers of participants, and/or indirect measures of the VPO. In this study, we tested whether the predictions from the previous studies extend to sentence-level speech with a substantially larger speech sample drawn from more participants using a more direct measurement of the VPO. Using the Université Laval X-ray videofluorography database containing speech from nine Quebecois French speakers, the VPO was measured for speech segments and rest positions. Results indicate that VPO was greatest during between-utterance intervals, followed by (in descending order) phonemically nasal segments, contextually nasalized segments, and oral segments. Contrary to previous suggestions of differences of VPO in nasal segments, no difference was found between nasal vowels and nasal consonants suggesting that such classification is not necessary (and the traditional binary [±nasal] feature still holds). Furthermore, in contrast to previous reports, anticipatory nasalization was found to have greater VPO than carry-over nasalization.

3aSC12. The Non-Native Articulatory corpus—French: An ultrasound corpus of French as an additional language. Vincent Chanethom (Modern and Classical Lang., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, vchaneth@gmu.edu), Giulia Masella Soldati, Haley A. Todd, Hannah M. Brennan (English, George Mason Univ., Fairfax, VA), Jaxon V. Myers, Dominique A. Hannon (Modern and Classical Lang., George Mason Univ., Fairfax, VA), Brenna J. Yaghmour, and Harim Kwon (English, George Mason Univ., Fairfax, VA)

We present a new corpus of spoken French, the Non-Native Articulatory corpus (NNA)—French, which is being made available to the research community. The NNA—French is a collection of simultaneous ultrasound and audio recordings produced by French speakers from various linguistic backgrounds, including both native and nonnative speakers of the language. Each speaker read a list of 60 short sentences in French, and the acoustic signals and the mid-sagittal ultrasound images of the tongue movement (~80 fps) were simultaneously recorded. Speakers also completed a comprehensive language background questionnaire. Each speech sample is presented with the speaker's demographic information (gender, age, birth place), language background (native language, any additional languages spoken), and French experience and proficiency. The articulatory information from the ultrasound images allows various types of phonetic research, such as comparing French speakers from diverse background, investigating articulatory strategies used by learners of French, and examining phonological development of French learners at different proficiency levels. The NNA—French continues to grow and we expect it to be a valuable resource for researchers in the field of phonetics and second language learning, as well as for French language instructors.

3aSC13. Segmental effects on the transmission of variability between articulation and acoustics. Sarah Harper (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, skharper@usc.edu)

Previous research on articulatory-acoustic relations has generated conflicting results regarding the extent to which variability in articulation can be recovered from the acoustic signal (e.g., Guenther *et al.*, 1999; Whalen *et al.*, 2018). This study extends previous research probing the relationship between articulatory and acoustic variability by examining the extent to which individual differences in articulatory variability are recoverable from acoustics, and vice versa, for a set of American English consonants. A series of linear mixed effects regression models were fit to articulatory and acoustic data from tokens of /s/, /ʃ/, /l/, and /ɹ/ produced by 40 speakers in the Wisconsin X-Ray Microbeam Corpus (Westbury, 1994) to evaluate the extent to which the variability exhibited in one of these physical dimensions was recoverable from the signal in the other domain. The output of these statistical models was then compared to measurements of individual speakers' variability along specific articulatory and acoustic dimensions to evaluate whether individual differences in variability were conveyed between articulation and acoustics. The results of the analyses suggest that the variability exhibited in one physical domain is to some extent recoverable from the other, although this capacity varies across segments and articulatory/acoustic dimensions. [Work supported by NIH.]

3aSC14. Contribution of vocal fold thickness to gender perception. Zhaoyan Zhang (Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu), Jody Kreiman, and Jason Zhang (Head and Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA)

Male vocal folds are often longer and thicker than female vocal folds. The length difference is generally considered responsible for the notable

difference in fundamental frequency between men and women, which plays an important role in gender perception. The role of the thickness difference in gender perception is less clear, although our recent studies demonstrated an important role of thickness in the control of voice quality. The goal of this study is to investigate the contribution of changes in vocal fold thickness and stiffness to gender perception. Synthetic voices were generated using a three-dimensional vocal fold model with parametric variations in vocal fold geometry, stiffness, adduction, and subglottal pressure. The vocal tract was kept constant in order to focus on the contribution of the voice source. Listening subjects were asked to judge speaker gender in a multiple choice task. Preliminary results showed a significant effect of vocal fold thickness and transverse stiffness on gender perception, with voices produced by thicker vocal folds and lower transverse stiffnesses more likely to be judged as male. Acoustically, gender perception scores were correlated with the fundamental frequency and spectral shape. [Work supported by NIH.]

3aSC15. Acoustic spaces for normal and pathological voices. Yoonjeong Lee (Departments of Linguist and Head & Neck Surgery, Univ. of California, Los Angeles, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeonglee@ucla.edu) and Jody Kreiman (Departments of Linguist and Head & Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA)

What does it mean for a voice to sound normal? Our recent work comparing listener judgements of talkers with and without a diagnosed voice disorder showed that judgments of "normal" versus "not normal" and strategies for estimating how much a given voice deviates from normal depended on the listener, the context, the purpose of the judgement, and other factors as well as on the voice. F0, F1 and F2, and F0 variability were significant predictors of both categorical judgments and of scalar normalness ratings, but between-listener inconsistency in the perceptual models was observed. The present study examined acoustic spaces for the same normal and pathological voices by performing principal component analysis on scaled values of F0, formant frequencies, spectral noise, source spectral shape, and their variability. Acoustic spaces for normal and disordered talkers had similar structures, and included measures related to source spectral shape, F1, and higher formant frequencies. Overall, variability in noise was only important for pathological voices. The ways in which acoustic voice spaces are related to listener perceptions of a voice are discussed. [Work supported by NIH/NSF.]

3aSC16. A cross-linguistic investigation of acoustic voice spaces. Yoonjeong Lee (Departments of Linguist and Head & Neck Surgery, Univ. of California, Los Angeles, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeonglee@ucla.edu), Marc Garellek (Dept. of Linguist, Univ. of California, San Diego, La Jolla, CA), Christina Esposito (Dept. of Linguist, Macalester College, Saint Paul, MN), and Jody Kreiman (Departments of Linguist and Head & Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA)

This study examines the extent to which the phonological structure of a language impacts acoustic variation in voice spaces for individuals and populations of speakers. Our recent work on two typologically different languages, American English and Seoul Korean, showed striking similarities in the acoustic spaces derived from Korean and English voices, but also notable differences in the importance of F0 and variability in vowel quality that were related to phrasal intonation patterns present in the Seoul dialect, but not in English. The present study examined eight speakers (5F) of Hmong, a language with tone and phonation contrasts. We performed principal component analysis on scaled values of F0, formants, spectral noise, source spectral shape, and their variability, measured from vowels, and compared results to Korean and English (which lack tone and phonation contrasts). Results revealed both substantial similarities and differences across languages. F0 commonly emerged only for Hmong and Korean voices, and H1-H2 (correlated with phonation) accounted for substantial variation only in Hmong voices. Our findings suggest that both biologically and phonologically relevant factors shape acoustic voice spaces, and point to a potential mechanism for the "own language" advantage in speaker perception. [Work supported by NIH/NSF.]

3aSC17. Prediction of vocal roughness using measures of temporal envelope fluctuation obtained from an auditory model. Yeonggwang Park, Erol J. Ozmeral, Supraja Anand (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), Rahul Shrivastav (Office of the Vice President for Instruction, Univ. of Georgia, Athens, GA), and David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu)

Vocal roughness often is present in many voice disorders, but the assessment of roughness mainly depends on the subjective auditory-perceptual evaluation and lacks unique acoustic correlates. This study aimed to apply the concept of roughness in general sound quality perception to vocal roughness assessment and to characterize the relationship between vocal roughness and temporal envelope fluctuation measures obtained from a biologically inspired auditory processing model. Ten /ɑ/ recordings with a wide range of roughness were selected from an existing database. Ten listeners evaluated perceived roughness using a single-variable matching task. Temporal envelope fluctuations of the recordings were analyzed with an auditory processing model of amplitude modulation based on a modulation filterbank tuned to different modulation frequencies. Pitch strength and the smoothed cepstral peak prominence (CPPS) also were computed for comparison. Individual simple regression models yielded envelope standard deviation from a modulation filter with a low center frequency (64.3 Hz) as a statistically significant predictor of vocal roughness with a strong coefficient of determination ($R^2 = 0.81$). Pitch strength and CPPS were not significant predictors of roughness. These results support the utility of envelope fluctuation measures from an auditory model as objective correlates of vocal roughness. [Work supported by NIH R01DC009029.]

3aSC18. On the significance of three-dimensional imaging for interpreting vocal fold dynamics. David Berry (Head and Neck Surgery, Univ. of California, Los Angeles, 31-24 Rehab, Los Angeles, CA 90095-1794, daberry@ucla.edu)

Physical mechanisms of regular and irregular vocal fold vibration were first studied in a finite element model of vocal fold vibration using the method of empirical eigenfunctions. Later the same method was used to successfully study physical mechanisms of vocal fold vibration in laboratory hemilarynx experiments (in which the medial surface of vocal fold was imaged). In subsequent clinical studies, however, the method exhibited significantly less interpretive power, presumably because of the limitations imposed by imaging from a 2D superior view. Our hypothesis is that the interpretive power of the method of empirical eigenfunctions will be significantly enhanced in clinical settings if 3D imaging is used instead of 2D imaging. To test this hypothesis, the method of empirical eigenfunctions was employed on the same finite element model implemented previously, as dynamical information was systematically removed from the analysis (interior tissue dynamics, medial surface dynamics, vertical dynamics of the superior surface, etc.). The natural modes of vibration were re-computed at each stage, and the deformations from the original natural modes were observed. The results suggest that in clinical settings, the method of empirical eigenfunctions retains significant interpretive power regarding physical mechanisms of irregular vibration when 3D imaging is used.

3aSC19. Seeking the source of vocal fold lesions towards measuring contact pressures during phonation with a pressure-sensing-matrix at hemi-larynx experiments. Florian Scheible (Inst. of Measurement and Sensor Technol., UMIT TIROL - The Tyrolean Private Univ., Eduard-Wallnöfer-Zentrum 1, Hal in Tirol, Tirol 6060, Austria, florian.scheible@umit.at), Reinhard Veltrup, Casey Schaan (Div. of Phoniatrics and Pediatric Audiol., Dept. of Otorhinolaryngology-Head and Neck Surgery, University Hospital Erlangen Med. School, Friedrich-Alexander-University Erlangen-Nürnberg, Erlangen, Germany), Raphael Lamprecht (Inst. of Measurement and Sensor Technol., UMIT TIROL - The Tyrolean Private Univ., Hall in Tirol, Austria), Marion Semmler (Div. of Phoniatrics and Pediatric Audiol.,Dept. of Otorhinolaryngology-Head and Neck Surgery, University Hospital Erlangen Med. School, Friedrich-Alexander-University Erlangen-Nürnberg, Erlangen, Germany), Manuel Staggl, and Alexander Sutor (Inst. of Measurement and Sensor Technol., UMIT TIROL - The Tyrolean Private Univ., Hal in Tirol, Tirol, Austria)

During phonation, the vocal folds execute a complex movement. Related to the subglottal pressure and tension of the vocal folds themselves, this movement may differ, furthermore depending on the kind of phonation a collision between both vocal folds may occur. If the vocal folds are overstrained, this can cause lesions. To gain knowledge about the vocal folds movements and the way of their collisions, a high-speed evaluation unit for a pressure-mapping sensor was developed. The sensor brings along a local resolution of 27.6 measure points per square centimeter and with the developed unit an adjustable framerate up to 1 kHz can be realized. Due to a sensor thickness of about 0.2 mm, an integration into hemi-larynx flow experiments is quite simple without vigorous changes on the setup. Because of an unrestricted field of view it is possible to record the movements with a high-speed camera. These recordings will be correlated with contact pressures in the post processing. During these experiments, different flowrates of the excitation air and various abduction and elongation steps on the hemi-larynx are obtained to study the effect of those manipulations.

3aSC20. Elastography of a gelatin M5 model mimicking the true vocal folds. Raphael Lamprecht (Inst. of Measurement and Sensor Technol., UMIT TIROL - The Tyrolean Private Univ., Eduard-Wallnöfer-Zentrum 1, Hall in Tirol 6060, Austria, raphael.lamprecht@umit.at), Florian Scheible (Inst. of Measurement and Sensor Technol., UMIT TIROL - The Tyrolean Private Univ., Hal in Tirol, Tirol, Austria), Marion Semmler (Dept. of Otorhinolaryngology-Head and Neck Surgery, Div. of Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen Med. School, Friedrich-Alexander-Univ. Erlangen-Nürnberg, Erlangen, Germany), and Alexander Sutor (Inst. of Measurement and Sensor Technol., UMIT TIROL - The Tyrolean Private Univ., Hal in Tirol, Tirol, Austria)

The material properties of the vocal folds play an important role in phonation, but are also of interest for basic research. In the diagnosis of e.g. vocal fold palsy ultrasound is still an emerging technology. Ultrasound elastography, a technique measuring the tissue stiffness, is widely employed in the diagnosis of e.g. liver fibrosis and thyroid nodules. Using a quasi-static approach, this technique can also be applied to the vocal folds, improving the understanding of their biomechanical properties and, in the future, the diagnosis of pathological changes in the vocal folds. We use a gelatin phantom of the vocal folds, modeled after the well-known M5 model of the true vocal folds and augmented with additional material that mimics the surrounding tissue. The phantom is compressed with an ultrasound transducer (central frequency of 15 MHz) in the inferior direction, with the image plane in the frontal plane. Ultrasound images of the compression process and the acting force are recorded. Based on this information, a developed elastography algorithm evaluates the data. The results of the experimental setup are compared with simulation results generated using FEBio, a software specialized in tissue simulation.

3aSC21. Acoustic and aerodynamic aspects of nasality in Guarani. Rita Demasi (Lab. of Phonet. and Phonology, CNRS-UMR 7018, Sorbonne Nouvelle, Paris, France) and Didier Demolin (Lab. of Phonet. and Phonology, CNRS-UMR 7018, Sorbonne nouvelle, Paris, France, didier.demolin@sorbonne-nouvelle.fr)

Guarani presents some interesting characteristics for nasality: nasalization of voiceless and voiced fricatives and long nasal spans in which two types of nasals (strong and weak) are observed. This study is based on recordings made with 4 native Guarani speakers from Paraguay (1 woman and 3 men) using synchronized acoustic and aerodynamic (oral and nasal airflow) recordings. Data show that voiced fricatives lose their frication when nasalized ṽ > ʋ̃[1]. For voiceless fricatives some nasal airflow can be observed, but no acoustic effects are observed below a certain threshold. Above this threshold, the acoustic spectrum is damped and the frication noise intensity is reduced, creating acoustically predictable phonetic variants s̃ > θ; x̃ > h. Thus, perceptually oral segments can be produced with a velopharyngeal port opening during voiceless fricatives, letting weak nasal airflow passing through (0.1–0.2 dm^3/s). Two types of nasals are observed in nasal spans: strong and weak [2]. Strong nasals occurring after voiceless obstruents have a sharp nasal airflow peak, up to 2 dm^3/s, after the oral constriction. This shows that the velum closure is achieved after the oral closure. Weak nasals occur in the environment of sonorants and vowels with weaker nasal airflow values. [1] J. J. Ohala and M. Ohala (1993). [2] E. Gregores and J. A. Suarez (1967).

Session 3aSP

Signal Processing in Acoustics, Physical Acoustics, Underwater Acoustics, Biomedical Acoustics, and Structural Acoustics and Vibration: Time Reversal Acoustics

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Chair's Introduction—8:20

Invited Papers

8:25

3aSP1. Sponge-layer damping techniques for enhancing the resolution and localization accuracy of acoustic time-reversal simulation. Akhilesh Mimani (Dept. of Mech. Eng., Indian Inst. of Technol. KANPUR (IIT-K), Kanpur, Uttar Pradesh 208 016, India, amimani@iitk.ac.in)

This work reviews the development of a set of sponge-layer damping techniques aimed at enhancing the focal-resolution of acoustic source(s) using Time-Reversal (TR) simulations. The method, termed as the Point-Time-Reversal-Sponge-layer (PTRSL) is based on damping the radial components of fluxes propagating towards and away from the assumed location resulting in local focusing of acoustic power. First, the test-case of an idealized monopole source was considered whereby it is demonstrated that a repeated application of PTRSL produced a point-like resolution while using a sensitivity analysis, it can resolve two or more sources in-proximity. Next, the method was demonstrated for canonical experimental aeroacoustic problems, namely, the rod-airfoil interaction and airfoil trailing-edge self-noise. When self-scattering due to airfoil was modeled, focal spots resembling a cardioid were obtained, however, large side-lobes posed difficulty in readily identifying the true source location. To overcome the problem, a (Line) LTRSL technique was developed which assumes that dipole sources of varying strength are distributed along the chord, and computes the root-mean-square (RMS) average of the PTRSL maps for each location. An iterative implementation of LTRSL was shown to suppress the side-lobes, and deliver the true scattered location of dipole source(s) along the chord whilst simultaneously producing a highly enhanced resolution.

8:45

3aSP2. Incomplete acoustic time reversal, frequency shifting, and matched field processing. David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan,1231 Beal Ave., Ann Arbor, MI 48109, drd@umich.edu)

In a reciprocal time-invariant medium, sound can travel in both directions along acoustic paths. This bi-directionality enables acoustic retrofocusing via time reversal (TR) and acoustic source localization via matched field processing (MFP). Given this common basis, the simplest frequency-domain formulations of acoustic TR and MFP are essentially identical; both involve a quadratic product of forward- and backward-propagating complex fields. For this presentation, all scenarios begin with the acoustic field from a remote source that propagates forward through a potentially-complicated acoustic environment to a transducer array where it is recorded. The difference between TR and MFP then arises because the second backward-propagating (time-reversed) acoustic field may be (i) a genuine field in the actual environment as in acoustic TR, or (ii) a predicted field developed from a model of the environment as in MFP. This presentation describes a third possibility, frequency-difference MFP or incomplete acoustic TR, that combines (i), (ii), and a multiplicative property of harmonic plane waves to produce frequency-downshifting. The resulting unconventional formulation involves a cubic product of complex fields and allows robust reduced-resolution source localization with sparse arrays in complicated uncertain environments. Example results from simulated, laboratory, and ocean propagation data are shown. [Sponsored by ONR, NAVSEA, NSF].

9:05

3aSP3. Passive time-reversal mirror for acoustic characterization of the seabed. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu), Dexter Tan (Phys. Dept., Naval Postgrad. School, Singapore, Singapore), and Tsu Wei Tan (Dept. Marine Sci., ROC Naval Acad., Kaohsiung, Taiwan)

This paper investigates application of a signal processing technique, which is motivated by time reversal, to geoacoustic inversions in a shallow-water waveguide. The signal processing mimics operation of a physical time-reversal mirror. The input data for the simulated time-reversal mirror can be obtained using either a compact, broadband sound source or cross-correlation of the diffuse noise recorded by spatially separated receivers. Low-frequency sound propagation is considered over ranges that are large compared to the water depth. The approach exploits the notion that the "best" focusing of the backpropagated time-reversed signal occurs at the "right

point," when the backpropagation takes place in the same propagation medium as the one, where the data have been acquired. Various metrics of the focusing quality are considered. A combination of spatial and temporal characteristics of the focus is proposed that leads to a unique solution of the geoacoustic inverse problem for a single-element passive time-reversal mirror. [Work supported by NSF.]

9:25

3aSP4. Subsurface target illumination in elastic hosts: a comparison of time-reversal and inverse-source approaches. Heedong Goh (Civil, Architectural and Environ. Eng., The Univ. of Texas at Austin, 301 East Dean Keeton St. Stop C1748, Austin, TX 78712, heedong.goh@utexas.edu), Seungbum Koo (Civil, Architectural and Environ. Eng., The Univ. of Texas at Austin, Austin, TX), Pranav Karve (Civil and Environ. Eng., Vanderbilt Univ., Nashville, TN), and Loukas Kallivokas (Civil, Architectural and Environ. Eng., The Univ. of Texas at Austin, Austin, TX)

The need for focusing wave energy to selected targets embedded within a host is commonly shared among various engineering fields, whether for stimulation and imaging in exploration geophysics, for contaminant removal in geo-environmental engineering, or for diagnostic/therapeutic purposes in medicine. In this presentation, we discuss the problem of focusing wave energy to multiple subsurface targets embedded within a semi-infinite heterogeneous elastic host. We review the advantages and disadvantages of two distinct approaches: first, using the apparatus of PDE-constrained optimization, we describe an inverse source (IS) approach, where we seek to design optimal excitations (both in space and time) in order to maximize target illumination. The second approach is rooted on the time-reversibility (TR) of the lossless wave equation; however the unboundness of the host, and equipment limitations that necessitate that recorded Dirichlet data be time reversed as Neumann data, create particularly adverse conditions for time-reversal. To overcome, we discuss a switching time-reversing mirror, and report on numerical experiments that show illumination resolution improvement. We define target illumination metrics to assess the relative performance of the IS and TR methods, and, using numerical experiments, we show that both approaches have similar efficiency.

9:45

3aSP5. Statistical classification of scatterer ultrasonic signature in bio-mechanical media and spatial localization of digital data using nonlinear time reversal. Zuzana Dvorakova (Inserm U1253 iBrain, INSA Ctr. Val de Loire, Blois, France), Vaclav Kus (Dept. of Mathematics, Czech Tech. Univ., Prague, Czechia), Zdenek Prevorovsky (Inst. of Thermomechanics ASCR, Prague, Czechia), and Serge Dos Santos (Inserm U1253 iBrain, INSA Ctr. Val de Loire, 3 Rue de la Chocolaterie, Blois 41034, France, serge.dossantos@insa-cvl.fr)

Various nonlinear methods have been investigated in recent years offering improvements of over conventional B-mode in medical imaging and non-destructive testing (NDT) [1], including Nonlinear Elastic Wave Spectroscopy (NEWS) methods. Furthermore, new optimized excitations such as Nonlinear Time Reversal (TR) based methods are classified within TR-NEWS methods [2]. In this context, one of the strategic plan of the international NDT community is to define standards for developing advanced and digitalized automated NDT set-up in modern mass production, defining the concept of NDT 4.0 or NDE 4.0 [3]. In the typical NDE4.0 TR-NEWS equipment, an optimized chirp-coded signal in the range of 0.6–3 MHz is used as a compression coding. The signal classification is performed using the Fuzzy Classification (FC) method and the Divergence Decision Tree (DDT) algorithm using specific ϕ-divergence spectral measures extracted from the received ultrasonic response containing acoustic nonlinearities. Finally, experimental setup and scatterer classification procedures have been further validated in ondotology with tooth echodentographic experiment where the ultrasonic responses to chirp-coded signals were monitored with a standard Polytec vibrometer, showing the efficiency of TR-NEWS methods for understanding of ultrasonic wave propagation in complex media.

10:05–10:20 Break

Contributed Papers

10:20

3aSP6. Quantifying nonlinear characteristics in time reversal focusing of airborne sound. Brian D. Patchett (Phys. & Astronomy, Brigham Young Univ., 800 W University Pkwy, MS-179, Orem, UT 84058, brian.d.patchett@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal can be used to focus sound to a chosen location within a reverberant environment. Virtual sources in the reverberant environment increase the amplitude of the focusing. Recent investigations into the focusing of high-amplitude sound waves, up to peak levels of 200 dB, has led to the observation of nonlinear characteristics in the focus event. This study investigates these nonlinear characteristics, including distortion of the focus wave, shift in harmonic content, and nonlinear increase in the peak pressure compressions. This study also attempts to quantify a threshold where the focus events begin to transition from a linear to the nonlinear regime. Additionally, a comparison of the microphones and transducers used to record these focus events is made in order to both verify the levels of pressure attained, and to ensure that microphone distortion is not the cause of the nonlinear features in the focusing.

10:35

3aSP7. Self-portrait of an ultrasonic wave propagating in a random scattering medium. Elsa Giraudat (Institut Langevin - CNRS, 1 rue Jussieu, Paris 75238 CEDEX 05, France, elsa.giraudat@espci.fr), William Lambert, Flavien Bureau, Mathias Fink, and Alexandre Aubry (Inst. Langevin - CNRS, Paris, France)

Synthetic beamforming for ultrasound imaging relies on a double focusing operation of back-scattered waves at transmission and reception on each point of the medium. Matrix imaging consists in decoupling the location of these transmitted and received focal spots .The response between those virtual transducers forms the focused reflection matrix that contains much more information than raw ultrasound images. Here, we aim to investigate the focused reflection matrix beyond the focusing time to fully leverage the available temporal information. In this time-dependent frame, each virtual source is associated with a focused incident wave while each virtual sensor enables to probe the propagation of this wave through the medium in the time domain. . This approach is the matrix equivalent of iterative time-reversal methods developed in speckle noise*(Montaldo, PRL 2011)*. However, the matrix formalism enables to perform this whole process in

post-processing and provides a much greater flexibility and efficiency. It constitutes a promising tool for an optimized local spatio-temporal focusing of ultrasonic waves, even in presence of reverberation and multiple scattering phenomena. It also paves the way towards a quantitative characterization of random scattering media, by providing, for instance, a high resolution map of integrated speed-of-sounds in soft tissues.

10:50

3aSP8. Reflection matrix approach in a multiple scattering regime: Application to the tracking of a sinking sphere in quicksand. Arthur Le Ber (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, 1 rue Jussieu, Paris 75005, France, arthur.le-ber@espci.fr), Flavien Bureau, Xiaoping Jia, Arnaud Tourin, Mathias Fink, and Alexandre Aubry (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, Paris, France)

Conventional ultrasound imaging generally relies on a single scattering assumption. Assuming a homogeneous speed of sound, a one-to-one correspondence is actually found between each scatterer position and the time-of-flight of the associated echoes. However, in strongly scattering media such as granular media, multiple scattering (MS) cannot be neglected beyond a few scattering mean free paths l_s, the typical distance between two scattering events. A conventional image is then no longer a satisfying estimator of the medium reflectivity. Here, we propose an original solution relying on a matrix formalism to cope with MS and image deeper than the conventional MS limit. As a proof-of-concept, our experiment consists in the acquisition of a reflection matrix associated with a 2D ultrasound probe (1024 elements -3 MHz central frequency) of a steel sphere sinking in fluidized glass beads immersed in water ($l_s \approx$ Ø600 μm). As expected, we show that the target echo vanishes into the MS background when it goes deeper than a few l_s. Interestingly, an iterative time-reversal analysis of the reflection matrix is proposed to compensate for the phase distortions induced by the gap between the effective medium properties and our beamforming model. The target can then be tracked at larger penetration depths than the usual MS limit.

11:05

3aSP9. Equivalent circuit model to study super resolution in time reversal focusing in a periodic network of Helmholtz resonators. Adam D. Kingsley (Phys. and Astronomy Dept., Brigham Young Univ., Provo, UT 84602, adamkingsley@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

This presentation will introduce equivalent circuit modeling techniques that have been used to simulate the focusing of acoustic signals in a 1-D waveguide. The presence of resonators in the near field of the focus location decreases the effective wavelength of the focus and leads to super resolution. A circuit is created that represents the complete waveguide with the sources. Circuit elements are used to represent both resonators and the propagation of the wave. The spatial extent is simulated at individual frequencies using additional circuit junctions. The wideband response is determined by the superposition of the individual frequency results. Important parameters such as resonator density and the quality factor of the resonators are examined and their influence on the focus will be presented. Methods for solving the circuit models and improving computational efficiency will also be discussed.

11:20

3aSP10. A finite-element based implementation of time-reversal and phase-conjugation acoustic source localization techniques in COMSOL multiphysics. Akhilesh Mimani (Dept. of Mech. Eng., Indian Inst. of Technol. Kanpur (IIT-K), Kanpur, Uttar Pradesh 208 016, India, amimani@iitk.ac.in) and Alok Sartaj (Mech. Eng., Indian Inst. of Technol. Kanpur, Kanpur, Uttar Pradesh, India)

In this work, we develop a general working procedure for a finite-element (FE) based implementation of time-reversal (TR) and phase-conjugation (PC) acoustic source localization techniques in COMSOL multiphysics commercial package using its in-built acoustics modules. To this end, a detailed step-by-step algorithm of the forward propagation is presented which includes creation of the computational domain and absorbing boundaries, defining acoustic source(s), meshing, solving, i.e., carrying out a time- or frequency-domain simulation followed by extracting data at virtual microphone array (nodes). The same procedure is used for implementing the TR simulations and frequency-domain PC except that source(s) inside the domain are now deactivated, and time-reversed or phase-conjugated data is enforced at the array nodes, followed by computation of the source map. The test-case of an idealized monopole in a 2D free-space is first considered whereby it is shown that both FE-based TR and PC methods deliver identical results which are in an excellent agreement with a finite-difference time-domain (FDTD) solver results, thereby validating the procedure. Next, the problem of localizing an idealized source in the presence of multiple scatterers and reflections which resemble an urban environment is considered. The COMSOL algorithms are shown to produce satisfactory results which demonstrate the efficiency of commercial FE solvers in handling complex geometries which otherwise tend to become tedious using the traditional FDTD solvers.

11:35

3aSP11. Cancellation of dolphin sonar click in communication signal. Donghyeon Kim (Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, Busan 49112, Republic of Korea, donghyeon.ual@gmail.com), Jiyoung Song, and J. S. Kim (Korea Maritime and Ocean Univ., Busan, Republic of Korea)

In long-range underwater communication, the conventional passive Time Reversal Processing (TRP) is widely used to mitigate the distortion due to the multipaths and temporal spreading. However, the signals produced by marine animals were present in the data acquired during the experiment, contaminating the communication signal and acted as an unwanted signal. In particular, because the impulsive signal such as a dolphin sonar click has a wide bandwidth and a short pulse duration, it is difficult to separate the communication signal from the impulsive signal. This overlapping problem affects critically the communication performance. In this study, we propose a method to successfully cancel click sound. The method involves combining the adaptive TRP and the estimation of Green's function of click sound. The adaptive TRP was successfully used in canceling crosstalk between interferers in a multiuser environment. Also, the estimation of Green's function utilized the characteristics of the impulsive signal with clear distinction between paths. The effect of spatially nulling click sounds without distorting the phase of the communication signal showed a remarkable improvement in communication performance in the seagoing experimental data of BLAC18.

Session 3aUW

Underwater Acoustics: General Topics in Underwater Acoustics: Modeling and Measurements I

Aijun Song, Cochair
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Advanced Mathematics Applications, Metron, Inc., 2020 SW 4th Ave., Suite 170, Portland, OR 97201

Contributed Papers

7:55

3aUW1. Mode-based models with realistic boundary conditions for a laboratory tank. Kaylyn N. Terry (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, kaylynterry@gmail.com), Cameron T. Vongsawad, and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

A new laboratory water tank at Brigham Young University is being characterized, and a suitable model is needed to describe the sound propagation. Recently, Novak *et al*. [*Acta Acust. Acust*., 2018] provided a model for low frequency sound propagation in small tanks with thin walls and Dirichlet boundary conditions. Our goal is to find a model that works at higher frequencies in a large water tank, which includes realistic boundary conditions. These models were specifically made for a rectangular acrylic tank of 3.6 m long by 1.2 m wide with 0.91 m as the maximum depth over a frequency band of 10 Hz to 100 kHz. Solutions to the 3D wave in Cartesian coordinates were used with Dirichlet and Neumann boundary conditions, with key differences noted between transmission loss estimates when realistic boundaries and ideal approximations are used. The models are then updated to include estimates of the characteristic acoustic impedance of the wall material to modify boundary conditions. Each model is sensitive to the number of modes used. The tank sound propagation model will allow us to model our tank with and without anechoic panels, simulate the sound field for optimization problems, and create accurate training data for machine learning applications.

8:10

3aUW2. Acoustic characteristics from an in-water down-the-hole pile drilling activity. Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, 45600 Woodland Rd., Sterling, VA 20166, guan@cua.edu), Tiffini Brookens (Marine Mammal Commission, Bethesda, MD), and Robert Miner (Robert Miner Dynamic Testing of Alaska, Inc., Manchester, WA)

Sound generated by pile installation using a down-the-hole (DTH) hammer is not well documented and differs in character from sound generated by conventional impact and vibratory pile driving. This paper describes underwater acoustic characteristics from DTH pile drilling during the installation of 0.84-m shafts within 1.22-m steel piles in Ketchikan, Alaska. The median single-strike sound exposure levels measured at 10 m were 138 and 142 dB re 1 μPa^2 s for each of the two piles, with cumulative sound exposure levels of 188 and 193 dB re 1 μPa^2 s at 10 m, respectively. The sound levels measured at Ketchikan were significantly lower than previous studies, and the sound is determined to be non-impulsive in this study as compared to impulsive in the previous studies. These differences likely result from fact that the DTH hammer used at Ketchikan did not make direct contact with the pile, as had been the case in previous studies. Further research is needed to investigate DTH piling techniques and associated sound-generating mechanisms, and to differentiate the various types of sound emitted.

8:25

3aUW3. Effects of nonlinear internal gravity waves on normal-incident reflection measurements of marine sediments. Tzu-Ting Chen (Woods Hole Oceanographic Inst., 86 Water St., WHOI, Woods Hole, MA 02543, pingpingting326@gmail.com), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, RI), and Linus Y.-S. Chiu (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-Sen Univ., Kaohsiung, Taiwan)

Sonar data acquired by sub-bottom profilers and multi-beam echosounder systems are widely used to estimate geoacoustic properties of marine sediments. However, the uncertainty of the seabed property estimates caused by water column variability may limit the application. In this study, we focus on the acoustic focusing and defocusing effects of nonlinear internal waves on normal-incident reflection measurements in the South China Sea. The field experiment was conducted using a transceiver mooring consisted of a sound source and a four channel vertical hydrophone array. The variation of acoustic signal reflections from sub-bottom layers during three days at the same site is observed. The effects of the nonlinear internal waves passing through the transceiver mooring are identified. The impact on the sub-bottom property estimates is investigated. [Work supported by the Office of Naval Research.]

8:40

3aUW4. Underwater acoustic intensity characterization of anechoic panels in laboratory tank. Corey E. Dobbs (Phys. and Astronomy, Brigham Young Univ., Provo, UT, coreydobbs205@gmail.com), Gabriel H. Fronk, Tracianne B. Neilsen, and Cameron T. Vongsawad (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Acoustic vector intensity contains more information about sound propagation than standard pressure measurements. This fact motivates the characterization of sound propagation in an acrylic rectangular water tank with dimensions of 3.6×1.2 m^2 and a maximum depth of 0.91 m, contained in the Underwater Acoustics Lab at Brigham Young University. With the acquisition of an M20-040 Particle Motion Sensor from GeoSpectrum Technologies Inc, comparisons are made between the p-p and the p-u methods of calculating intensity as a function of frequency and source-receiver range. In addition, the decrease in intensity is quantified when Apltile SF5048 anechoic panels made by Precision Acoustics are used to line the side of the tank. This work helps characterize the impact of panels on sound propagation in the tank. The continuation of research on acoustic vector intensity will also allow for future research collaboration with biologists at BYU to make important measurements relating to the effects of sound intensity on the auditory responses of fish.

8:55

3aUW5. Identifying the optimal k-mean clustering for ship noise spectrograms. Bethany Wu (Phys. & Astronomy, Brigham Young Univ., 112 W 1230 N Apt. #311, Provo, UT 84604, bwu98@byu.edu) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Unsupervised learning techniques have the potential to be useful for ocean acoustic data. For example, the k-means clustering algorithm has been trained on synthetic spectrograms from surface ships and then applied to measured data to try to identify probabilities of seabed type and details such as the speed and location of the ships. Although the algorithm is effective in grouping data into clusters, the variability of these clusters is heavily dependent on the number of clusters, "k." This paper aims to investigate the range of variability in clusters when different values of k are chosen for a given ocean acoustics dataset. To assess the variability of the results, different methods will be used to measure distances from the centroid of each cluster. These methods include measuring the average, maximum, and sum of squared distances from each data sample to its assigned centroid. The results when using the optimal number of clusters are shown and a discussion is provided about how to interpret the results.

9:10

3aUW6. Backscattering by broadside and tilted intermediate thickness empty cylindrical metal shells in water: Experiments and theory. Bernard R. Hall (Phys. and Astronomy, Washington State Univ., 430 NE Maiden Ln., Apt. 3, Pullman, WA 99163, bernard.hall@wsu.edu) and Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA)

In prior research the backscattering properties of empty metal shells in water were investigated over a wide frequency range [Morse *et al.*, *J. Acoust. Soc. Am.* **103**, 785–794 (1998); Morse and Marston, *J. Acoust. Soc. Am.* **111**, 1289–1294 (2002)]. The detailed high-frequency features of the observed and modeled enhancements depended on the shell thickness. In the present research backscattering is measured, and modified models are introduced for a steel shell intermediate in thickness between those considered in the aforementioned prior investigations. Some of the outcomes of this investigation may be summarized as follows: (a) for broadside scattering the contribution from a coincidence-frequency related guided wave is easily observed and is prominent within a ray-theory based model; (b) supersonic antisymmetric guided waves are evident at high frequencies; (c) such antisymmetric guided waves also especially result in easily noticed meridional wave backscattering enhancements when the shell is tilted as was previously demonstrated for other tilted shells. [Work supported by ONR.]

9:25

3aUW7. Mid-frequency sound transmission in a deep ice-covered ocean at short and medium ranges. Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu), John E. Joseph, and D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, Monterey, CA)

A model of propagation and reverberation in a deep Arctic ocean and a tentative interpretation of ICEX16 data are discussed. The data subset used for analysis was collected by a NPS team in the Beaufort Sea (~3650 m depth) on 12 March 2016 using mobile EMATT transmitters which were moving under the ice in circular patterns from Ice Camp SARGO that provided various combinations of source-receiver ranges (~1–10 km) and depths (45–183 m). Transmitted waveforms were sequences of 60 s-long segments comprised of 57 s-long CW pulses at various frequencies (2800, 2900, and 3000 Hz) and 3 s-long 2300–3000 Hz LFM sweeps. Sound speed profiles measured during the experiment showed two shallow ducts, at 0–75 m and 75–250 m water depths. The model developed considers multipath propagation including multiple reflections and scattering from rough interfaces. The model-based analysis allows examining the spatial and time-frequency structure of recorded arrivals and their sensitivity to variations in water stratification and acoustic parameters of ice and ocean bottom. Preliminary results of the ICEX16 data-model comparisons are presented. Suggestions are considered for environmental measurements to provide necessary inputs and ground-truth for the acoustic model. A potential for enhanced remote sensing capabilities and optimal configurations for future experiments are discussed. [Work supported by ONR.]

9:40

3aUW8. Evaluating applicability of an azimuthally symmetric model in a laboratory tank with absorptive panels. Scott P. Hollingsworth (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, sh747@byu.edu), Cameron T. Vongsawad (Phys. & Astronomy, Brigham Young Univ., Orem, UT), and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In our lab at Brigham Young University, a tank of water will be used to help validate machine learning algorithms that could be applied to ocean acoustics. A model for sound propagation in the tank is required, and we are exploring the possibility of using ORCA, a range-independent normal mode model [Westwood *et al.*, *JASA* (1996)]. Because ORCA assumes azimuthal symmetry in cylindrical coordinates, reflections from the tank walls must be significantly attenuated for ORCA to be a good model. To compare our measurements to ORCA, two approaches have been tried: (1) timegating and (2) absorptive panels on the walls. The layering used in ORCA is air on top, water with a maximum depth of 0.91 m, 3 cm thick acrylic bottom, a lower air layer, and then concrete on bottom. A simpler model excluding the air between the acrylic and the concrete was also tested. Both models were compared to measured data; both relative and absolute transmission loss were compared across a large frequency band and as a function of source/receiver distances at set depths. If ORCA is a good model for the tank, then we could easily generate training data for machine learning applications.

9:55–10:15 Break

10:15

3aUW9. Observation of variable-order Lévy statistics in atomizing microscale capillary wave dynamics. Jeremy Orosco (Mech. and Aerosp. Eng., Univ. of California San Diego, 13754 Mango Dr., Unit 306, Del Mar, CA 92014, jrorosco@ucsd.edu) and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

When excited with very small amplitude [$O(\leq 10^{-9})$ m], very high ultrasonic frequency [$O(\geq 10^{6})$ Hz] vibrations, an otherwise quiescent small volume [$O(10^{-6})$ μl] of water will exhibit oscillations at its ambient interface that are visible by eye [$O(10^{-2})$ m and $O(10^{1})$ Hz]. Recent experimentation has shown that this remarkable behavior occurs in the absence of any classically predicted originating mechanisms such as Faraday instabilities, so that the physics of ultrasonically vibrated microscale water volumes remains enigmatic, lacking detailed theoretical characterization. In this talk, we present recently acquired, high-speed holographic time-dependent measurements of the ambient interface of a microscale water volume subject to acoustic forcing in the MHz regime. We show that the statistical distribution of surface oscillations—capillary waves—is a continuously variable function of the input amplitude, varying from near Gaussian at small powers to Lévy α-stable at higher powers. The relationship between this statistical observation and near-atomization physics is discussed and potential modalities for low-dimensional mathematical characterization of the system are presented.

10:30

3aUW10. Mu-Net: Community-shared infrastructure for mobile underwater acoustic networks. Aijun Song (Elec. and Comput. Eng., Univ. of Alabama, 245 7th Ave., Tuscaloosa, AL 35487, song@eng.ua.edu), Xiaoyan Hong (Comput. Sci., Univ. of Alabama, Tuscaloosa, AL), Fumin Zhang (School of Elec. and Comput. Eng., Georgia Tech, Atlanta, GA), Zheng Peng (Dept. of Comput. Sci., City College of New York, New York, NY), and Zhaohui Wang (Elec. and Comput. Eng., Michigan Technol. Univ., Houghton, MI)

It is well recognized that there is a strong need for shared testbeds to promote research growth in underwater communities. We are developing an open-source, open-architecture, community-shared infrastructure for mobile underwater acoustic networks, namely mu-Net. The overall goal is to support research and education in the joint communities of underwater signal processing, communications, networking, and robotics. The mu-Net infrastructure consists of three main components: an indoor testbed, a lake testbed, and underlying open-architecture software. The mu-Net software uses a service-oriented architecture to interface with open-access autonomy suites, such as MOOS-IvP and ROS. Functions associated with sensing, communication, networking, vehicle behaviors, and navigation are compartmentalized as modular service units. The indoor testbed is developed for lab tests with a moderate number of nodes, for example, eight nodes, for deployments in a swimming pool. The lake testbed uses two commercial autonomous underwater vehicles (AUVs) that can be deployed in a lake. We will provide shared access to both testbeds for data collection and algorithm validation. We will present the software architecture, hardware instruments, and the obtained acoustic measurements from the testbeds.

10:45

3aUW11. Acoustic surface impedance of sandy shores. Faith A. Cobb (Eng., East Carolina Univ., Slay Hall 248, Greenville, NC 27858, cobbf18@students.ecu.edu), 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 work presents an effort to characterize the relationship between moisture content and acoustic surface impedance of sandy soils. This effort is part of a larger project seeking to model long-range acoustic transmission loss over coastal areas. Previous studies were done to measure moisture content along the shore and collect samples for bench-testing with an impedance tube. This work seeks to build on those experiments by performing *in situ* acoustic surface impedance measurements with an omnidirectional sound source and two microphone receivers. These measurements follow the protocol set forth by the ANSI/ASA S1.18 and are recorded at multiple distances from the water on the shore. At each measurement location, sand samples are collected for gravimetric moisture analysis. Atmospheric conditions and ambient noise level are also recorded in accordance with the standard. These results demonstrate the variation of sand surface impedance from the fully saturated swash zone to nominally dry sand.

11:00

3aUW12. Using neural networks to estimate the probability density function of transmission loss in ocean environments with a database-driven model of uncertainty. Brandon M. Lee (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu), Jay R. Johnson, and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The prediction of transmission loss (TL) in the ocean depends on the relevant environmental characteristics, such as sound speed field, bathymetry, and bottom properties. These characteristics are often uncertain as they are commonly not all measured *in situ*, and are instead estimated from available databases. Thus, the resulting predictions of TL are uncertain, and this uncertainty can be quantified via the probability density function (PDF) of TL. Given a model for the environmental uncertainty, Monte Carlo techniques and thousands of realizations of the environment can be used to estimate the PDF of TL, but this method is computationally costly. A variety of alternative uncertainty-estimation techniques have been proposed, including a machine learning approach which uses neural networks trained on large Monte Carlo-produced TL datasets. The resulting neural network inherits the limitations of its training datasets and their underlying model of environmental uncertainty. A new environmental uncertainty model is described here which uses higher-resolution databases. A revised neural network approach to predicting PDFs of TL is developed and tested on $\sim 4.5 \times 10^6$ examples across 300 environments with the new, more realistic model of uncertainty at ranges up to 100 km for source frequencies between 50 and 600 Hz. [Sponsored by an NDSEG Fellowship.]

11:15

3aUW13. Force chains, creep and acoustic reflection in marine sediments. Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, chotiros@utexas.edu)

Marine sediments are water-saturated granular media. A feature of granular media is that the stress is not uniformly distributed but concentrated in force chains. Several force chains acting in parallel transmit external stresses through the medium. A force chain is a chain of particles through which the external stresses are concentrated and transmitted. Given that the contact stiffness between particles may be approximated as a Maxwell element, i.e. a spring in series with a damper, then the stiffness of any force chain may be modeled as several Maxwell elements in series. It is known that any number of springs and dampers in series may be reduced to just one spring and one damper in series, i.e., one equivalent Maxwell element. Therefore, several force chains in parallel may be modeled as parallel Maxwell elements. However, parallel Maxwell elements may not be reduced to one Maxwell element. The stiffness of a distribution of parallel Maxwell elements falls under the generalized creep model. This model is used to represent the skeletal frame of a larger porous medium model. The inversion of model parameters from the reflection coefficient is explored in the context of a muddy sediment. [Work supported by ONR, Ocean Acoustics Program.]

11:30

3aUW14. Green's function of the rigid half cylinder with soft free surface for application to ship noise. Elina Cros (Naval Group, 199 Ave., Pierre-Gilles de Gennes, Ollioules 83190, France, elina.cros@ec-lyon.fr), Michel Roger (Ecole Centrale de Lyon, Écully, France), and Gilles Serre (Naval Group, Ollioules, France)

The present study addresses an experimental and analytical investigation of the scattering of the tonal noise of low-speed propeller by a half-cylinder and a soft free surface. The interest is marine propeller noise of surface ships at the first harmonics of the blade passing frequency. Indeed, installed marine propellers correspond to very compact configurations. Both sizes blade tip radius and the distance to the ship hull are small compared to the wavelength. Thus, in view of the extremely low Helmholtz numbers, some amplification due to an asymptotic scattering regime is expected. Measurements made in air in similar conditions, in anechoic chamber confirmed that noise from small-scale isolated propeller is dramatically intensified near a half cylinder with reflecting plane. The problem is formulated analytically by using the formalism of source-modes near a scattering half cylinder combined with the effect of the free surface in two dimensions. In this configuration, the diffraction is found to induce an amplification but with a different wave pattern to what the diffraction is with a cylinder only. The results show that the installation effect is crucial for the analysis of marine propeller noise at the very low frequencies.

11:45

3aUW15. Mode coupling in a coastal wedge due to quasi crossing of range-dependent modal eigenvalues and spatial sound field variability. Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), Yanyu Jiang, and Qianchu Zhang (Marine Geosciences, Univ. of Haifa, Haifa, Israel)

In the paper sound waves propagation in a coastal wedge with sound speed profile (the presence of thermocline) is studied. In down- and up-slope propagation each adiabatic waveguide mode is transformed from the "bottom—surface reflected" shape to the "bottom-bottom refracted" one at some distance from the edge, depending on mode number. Analysis of mode composition and the corresponding spatial variability of the sound field is carried out using Parabolic Equation with the following decomposition of the field over adiabatic modes. It is shown, that there is significant change of amplitudes of adiabatic modes which is interpreted as manifestation of mode coupling. It is shown that mode coupling has local character, at the definite distance from the source, where two range dependent eigenvalues convergent to each other. This phenomenon is analog to the so called quasi crossing of states in atomic physics (Landau–Zener theory of nonadiabatic transitions in two-level system). This leads to sequential excitation of more and more higher modes in down-slope propagation and even creation of higher modes which did not exist at the position of the source. Results of modeling are presented, possible experimental observations are discussed. [Work was supported by RFBR, Grant 20-55-S52005.]

Exhibit

An instrument and equipment exhibition will be located in the Ballroom near the registration area and meeting rooms.

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, 29 November, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary beverage.

Tuesday, 30 November, 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: 4 December, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. break serving coffee.

Session 3pAA

Architectural Acoustics: AIA CEU Presenters Course Training Session

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

Kenneth W. Good, Cochair
Armstrong World Industries, Inc., 2500 Columbia Avenue, Lancaster, PA 17601

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. Architectural acoustics short course presentation material. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects called "Architectural Acoustics." An architect can earn one continuing education unit (CEU) by attending this short course, if presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, finish treatments, and implementation of quality acoustical spaces. This paper will cover the course material to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this special session and membership in TCAA are required.

2:05

3pAA2. Architectural acoustics continuing education course—Presenter registration and reporting. Kenneth W. Good (Armstrong World Industries Inc., 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects called "Architectural Acoustics," for which attendees can earn one continuing education unit (CEU). This paper will cover the administrative requirements of the ASA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork so that AIA members may receive credit for the course. The manner in which the course is given is also dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course. Of course, anyone is free to register with the AIA to provide their own CEU program. However, the advantages of participating in this program are that the TCAA short course is already prepared, it is pre-approved by the AIA, and the registration fees are paid by the Acoustical Society of America.

Session 3pAB

Animal Bioacoustics: Animal Bioacoustic Signals Transmission and Environment

Contributed Papers

1:30

3pAB1. Acoustic propagation in heterogeneous environments and its implications on the active space of rock ptarmigan. Arthur Guibard (LMFA, Ecole Centrale de Lyon, 36 Ave., Guy de Collongue, Ecully 69134, France, arthur.guibard@ec-lyon.fr), Didier Dragna, Sebastien Ollivier (LMFA, Ecole Centrale de Lyon, Lyon, France), and Frédéric Sèbe (ENES/CRNL, Université Jean Monnet, Saint-Etienne, France)

Acoustic communication networks among birds are widely studied in homogeneous environment like tropical forest or open field. There is no study in heterogeneous environment to our knowledge. However, the propagation of an acoustic signal is strongly influenced by topography, meteorological and ground effects, especially at long distance. For bird species living in high mountains, these effects can have a prominent impact on how their vocalization spreads. Here we try to redefine the active space notion, using the rock ptarmigan as a model. We develop a sound propagation code dedicated to bioacoustic studies based on the parabolic equation method and taking into account the above mentioned effects. Propagation measurements were carried out at a mountain site to provide information on typical conditions encountered and to test the validity of our model in this context. Using this model, we describe how the mountainous conditions affect active spaces of communication compared to typical homogeneous situations. We further test how singing during display flights can be a good strategy for a transmitter to expand its active space in such a context. A fine modeling of the propagation of acoustic signals is likely to provide important cues for understanding communication networks of mountainous species.

1:45

3pAB2. Deep-learning exploration of the acoustic granularity of bat habitats. Liujun Zhang (Elec. and Comput. Eng., Virginia Tech, 1075 Life Sci. Circle ictas II, Blacksburg, VA 24060, sdujune@gmail.com), Andrew Farabow, Pradyumann Singhal (Comput. Sci., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Many bat species are able to find their way through densely vegetated habitats and select their microhabitats to suit their respective needs. This raises the question as to how granular the natural habitats of bats are when sensed through the echoes triggered by the animals' biosonar pulses. To investigate this question, a portable biomimetic sonar head has been used to collect about 40 000 foliage echoes across a natural forest site on the Virginia Tech campus that was approximately 250×140 m^2 in size. Each recorded echo was assigned a spatial location that was estimated from the outputs of a concurrently operated GPS receiver. These spatial locations were then used to cluster the echoes into a varying number of compact patches using a k-means clustering algorithm. To determine whether these spatial patches could be determined from the echoes, a convolutional neural network based on the ResNet50 architecture was trained to classify the echoes with respect to their spatial labels. Even based on just a single echo, up to 100 different spatial patches could be distinguished in this way with accuracies greater than 95%. These results demonstrate that bat biosonar can capture sensory information for small-scale navigation in natural forest habitats.

2:00

3pAB3. Fin whale localization and environmental inversion using modal arrivals of the 20-Hz pulse. Jennifer L. Amaral (Marine Acoust., Inc., 2 Corporate Pl., Ste. 105, Middletown, RI 02842, jennifer.amaral@marineacoustics.com), James H. Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, RI), and Kathleen J. Vigness-Raposa (Inspire Environ., Middletown, RI)

Fin whale doublet calls, described as two 20-Hz pulses recorded at different interpulse intervals, have been attributed to the whale's calling behavior, however they could also result from acoustic mode propagation effects. Modes travel with their own frequency-dependent group velocities. The dispersion of these modes results in the sound being recorded as multiple arrivals on a receiver. Multiple modal arrivals of 20-Hz fin whale calls were recorded, with mode one arriving after mode two. The time delay between modal arrivals varied throughout the recording and is range dependent. The range dependency of this time delay was used to estimate the range of the whale from individual hydrophones by determining the modal group velocities that resulted in the observed delays. A pair of hydrophones was used to localize the whale for the duration of time when two modes were detected. The KRAKEN normal mode model was utilized in an inversion scheme to determine the compressional wave speed and depth-to-bedrock in the study area that supported the estimated modal group velocities. The inversion resulted in a depth-to-bedrock of 200 m and compressional wave speed of 1735 m/s, which were supported by measurements reported in literature for nearby regions. [Work supported by BOEM.]

2:15

3pAB4. Investigating the acoustic directionality of mouse ultrasonic vocalizations. Yichong Ma (Psychol. and Brain Sci., Univ. of Delaware, 105 The Green, Rm. 108, Newark, DE 19716, yma@udel.edu), Megan R. Warren (Biology, Emory Univ., Newark, DE), and Joshua P. Neunuebel (Psychol. and Brain Sci., Univ. of Delaware, Newark, DE)

Across the animal kingdom, ultrasonic vocalizations (USVs) are used as a means of communication, navigation, and predation, and in some species (e.g., echolocating bats and whales), the directionality of the signal facilitates these processes. The utility and characteristics of mouse USVs, however, are less clear. Using an 8-channel microphone array, we recorded, localized, and assigned USVs to individual mice that were in isolation ($n = 28$) or socially interacting (male-female dyads, $n = 13$; 4-mouse, mixed-sex groups, $n = 11$). Males emitted a total of 3675 (100%), 22 268 (84.48%), and 73 488 (82.05%) of the assigned vocalizations when in isolation, interacting with a female, or socializing in a group. By examining the relationship between the USV sound intensity on the eight microphones and the head directions of the mice, we observed that male mice consistently produced directional USVs across each of the three behavioral contexts. Specifically, USV intensity was strongest in the direction that the mouse was facing and weakest in the opposite direction. Overall, these findings directly demonstrate that male mouse USVs are directional and may provide insight into the communicative function of mouse vocalizations.

Session 3pAOa

Acoustical Oceanography: Acoustical Oceanography Prize Lecture I

Grant B. Deane, Chair
Scripps Institute Of Oceanography, Code 0238, UCSD, La Jolla, CA 92093-0238

Chair's Introduction—1:00

Invited Paper

1:05

3pAOa1. A sound resolution to the food paradox in the sea. Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst. (MBARI), 7700 Sandholdt Rd., Moss Landing, CA 95003, kbb@mbari.org)

The average concentrations of biota in the ocean are generally low, presenting a critical problem for ocean consumers. In seminal experimental work in the 1970s, Lasker demonstrated that animals fed the average food concentrations measured using nets and pumps do not survive, presenting a paradox for life in the sea. Active acoustics, in contrast to more traditional tools, allow us to sample a wide range of animals at high spatial and temporal resolution. Employing a variety of platforms including ships, profilers, moorings, and autonomous vehicles to deploy acoustic sensors, often guided by the predators themselves, we find that instead of being relatively devoid of life, the ocean is peppered with narrow hot spots of activity. These features, often missed with traditional sampling tools, exist from the surface ocean to the deep sea, the tropics to the poles, in animals ranging from plankton and fish to squid and whales. These small patches of plenty have impacts on ecosystems disproportionate to their contribution to the total biomass. Small aggregations provide the key to solving experimentally demonstrated feeding paradoxes as well providing a mechanism for evolution in an apparently isotropic environment where there are no obvious barriers to gene flow, Hutchinson's "Paradox of Plankton."

Session 3pAOb

Acoustical Oceanography: Acoustical Oceanography Prize Lecture II

Grant B. Deane, Chair
Scripps Institute Of Oceanography, Code 0238, UCSD, La Jolla, CA 92093-0238

Chair's Introduction—2:15

Invited Paper

2:20

3pAOb1. Understanding marine life through long-term passive acoustic time series. Ana Sirovic (Trondhjem Biological Station, Norwegian Univ. of Sci. and Technol., Trondheim 7491, Norway, ana.sirovic@ntnu.no)

Sound allows us to study the ocean on a much finer time scales than other methods of study as it is feasible to collect continuous underwater recordings over months or years at a time. My work has been focused on using such long recordings to improve our understanding of the ecology of large whales and marine fishes. However, ecological insights can only be gained from data that are truly comparable across time and space. The initial steps to achieving comparable time series from long-term recordings collected at vastly different locations include: effective detection of signals of interest, full knowledge of the characteristics of those signals and interfering noise, and normalization of those detections by appropriate recording effort across time and space. I will highlight a few things we have learned about baleen whale and fish species through long-term passive acoustic monitoring effort, such as an improved understanding of call function for some baleen whales or a better sense of animal occurrence. This, in turn, will allow us to start coupling the occurrence of these animals into the larger patterns of ecosystem dynamics, ultimately leading to a fuller understanding of their roles in the ocean.

WEDNESDAY AFTERNOON, 1 DECEMBER 2021 COLUMBIA A, 1:00 P.M. TO 3:00 P.M.

Session 3pBAa

Biomedical Acoustics: Biomedical Acoustics Student Poster Competition

Kenneth B. Bader, Cochair
Department of Radiology, University of Chicago, 5835 South Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637

Kevin J. Haworth, Cochair
Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

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 and 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. The impact of lipid shell composition of perfluorocarbon nanodroplets on size distribution and acoustic droplet vaporization and cavitation dynamics
Student author: Phoebe J. Welch

1aBAa10. Bubble cloud progression in fibrous tissue mimicking hydrogels at different histotripsy sonications
Student author: Yashwanth Nanda Kumar

1aBAa11. Static magnetic fields dampen focused ultrasound-induced cavitation and blood-brain barrier opening outcome
Student author: Yaoheng Yang

1aBAa13. Reliable and safe blood-brain barrier opening by closed-loop feedback control of focused ultrasound
Student author: Chih-Yen Chien

1aBAb1. Ultrasonic intra-body communication using semi-guided waves through human body tissues
Student author: John O. Gerguis

1aBAb2. Micro-elastography on spheroids using a magnetic pulse as a shear wave source, and the impact of cavitation on their mechanical properties
Student author: Gabrielle Laloy-Borgna

1aBAb3. Effects of crosslinking density on the acoustic properties of hydrogel scaffolds
Student author: Megan Anderson

1aBAb5. Histotripsy bubble dynamics in anisotropic hydrogels
Student author: Jacob Elliott

1aBAb6. Atomization of a collagenase-treated tendon
Student author: Molly Smallcomb

1aBAb7. Three-dimensional echo decorrelation imaging of percutaneous microwave ablation in liver tumors
Student author: Elmira Ghahramani Z.

1pBAa5. Sonobiopsy increases release of circulating tumor DNA for sensitive molecular diagnosis of glioblastoma
Student author: Christopher Pham Pacia

1pBAb3. Image-guided focused ultrasound-mediated drug delivery and comparison between 2-D and 3-D therapeutic strategies
Student author: Ryan Margolis

2aBAa2. Elastic properties of human hematoma model and its sensitivity to histotripsy liquefaction
Student author: Ekaterina M Ponomarchuk

2aBAa3. Doppler ultrasound observations and metrics of boiling histotripsy treatment progression
Student author: Minho Song

2aBAa5. Low-intensity pulsed ultrasound induces metabolic lipolysis of adipocytes
Student author: Sangnam Kim

2aBAa6. Suppressing tissue clutter in the presence of motion and nonlinear propagation of ultrasound
Student author: Geraldi Wahyulaksana

2aBAa12. Demonstration of nonlinear interaction of crossed turbulent streaming jets
Student author: Jenna M. Cartron

2aBAa13. Nonlinear scattering of crossed ultrasonic beams in the presence of a turbulent bubbly jet flow in water
Student author: Katherine A. Haas

2aBAb3. Improving ultrasonic evaluation of osteochondritis dissecans using a cadaveric model
Student author: Philip M Holmes

2aBAb5. Color Doppler imaging of pure crystals in hyperbaric conditions
Student author: Eric Rokni

2aBAb7. Foam gratings as an alternative to customised acoustic lenses
Student author: Luke Alexander Richards

2pBAa2. Determination of nonlinearity parameter B/A of liquids by comparison with solutions of the three-dimensional Westervelt equation
Student author: Lonnie Chien

2pBAa3. Non-linearities under highly focused high-intensity ultrasound fields
Student author: Pradosh Pritam Dash

2pBAa4. Examining influence of primary frequency ratio on distortion product otoacoustic emissions generation using an alternating frequency/time domain cochlear model
Student author: Haiqi Wen

2pBAa6. Experimental measurement for assessing pseudo-sound effect in parametric array
Student author: Jiyoung Song

2pBAa7. Experimental investigation of parametric array for low frequency measurement system
Student author: Donghwan Jung

2pBAb2. Impact of phase-change contrast agent size during activation using ultrasound
Student author: Dominique James

2pBAb3. Sonopermeation to deliver topotecan using lipid-prodrugs, liposomes, and microbubbles
Student author: Mendi G Marquez

2pBAb5. C-mode passive cavitation images for predicting large volume FUS-mediated drug delivery outcome
Student author: Yan Gong

3aBAb8. Measurement of the arterial wall elasticity using the velocity of flexural waves in retinal blood vessels
Student author: Gabrielle Laloy-Borgna

4pBA11. Externally-induced shear waves in the right ventricular free wall during the cardiac cycle
Student author: Luxi Wei

5aBAa3. Cavitation is the primary mechanism for stone dusting in holmium: YAG laser lithotripsy
Student author: Junqin Chen

5aBAa7. Particle swarm optimization of the manipulation of acoustic waves through nonhomogeneous, anisotropic mediums for application in Shock Wave Lithotripsy
Student author: Joseph J Roemhildt

5aBAa10. Single-framework simulations of acoustic-wave–bubble cloud–stone interactions
Student author: Jean-Sebastien Alexandre Spratt

5pBA3. Breaking urinary stones to small size with burst wave lithotripsy
Student author: Shivani Ramesh

5pBA8. Acoustic radiation force on objects of arbitrary shape and composition
Student author: Blake E Simon

Session 3pBAb

Biomedical Acoustics: Biomedical Acoustics in Ophthalmology II

Jonathan Mamou, Cochair
Riverside Research, 156 William St., 9th Floor, New York, NY 10038

Xiaoming Zhang, Cochair
Mayo Clinic, 200 1st St. SW, Rochester, MN 55905

Invited Papers

1:00

3pBAb1. Plane-wave ultrasound assessment of ocular blood flow in preeclampsia and correlation with optical coherence tomography angiography. Ronald H. Silverman (Ophthalmology, Columbia Univ. Irving Medical Ctr., New York, NY, rs3072@cumc.columbia.edu), Srilaxmi Bearelly, Raksha Urs (Ophthalmology, Columbia Univ. Irving Medical Ctr., New York, NY), Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), and Ronald Wapner (Obstetrics and Gynecology, Columbia Univ. Irving Medical Ctr., New York, NY)

We examined both eyes of 58 post-partum subjects within 72 hours of delivery with plane-wave ultrasound and optical coherence tomography angiography (OCTA). 8 had preeclampsia (PE), 21 severe PE (sPE). 9 chronic or gestational hypertension and 21 were controls. We performed plane-wave ultrasound with a Verasonics Vantage-128 and L22-14vXLF 18 MHz linear array, acquiring 3-angle compound images of the region of the optic nerve and choroid at 6 kHz for 3 s. Data were post-processed to obtain pulsatile flow characteristics of the central retinal artery and vein, the short posterior ciliary arteries and the choroid. Ultrasound revealed significantly elevated blood flow velocity and reduced resistance in sPE compared to controls. Spectral domain OCTA images were acquired with a Zeiss Plex Elite immediately before or after ultrasound. We analyzed 6x6 images centered on the macula and optic nerve to determine vascular density of the superficial and deep retina, and 3 i 3 images centered on the macula to determine foveal avascular zone area and vascular density of the choriocapillaris. Comparison of ultrasound with OCTA findings revealed significant correlations between flow velocity and resistance with vascular density in the superficial retina and choriocapillaris. These findings demonstrate a relationship between functional (ultrasound flow) and structural (OCTA vascular density) parameters in the eye, and especially in the choriocapillaris which we hypothesized would play role in the reduced flow resistance observed in sPE.

1:20

3pBAb2. High-frequency quantitative ultrasound to assess chemical crosslinking induced changes in the posterior sclera. Cameron Hoerig (Riverside Res., 156 William St., Fl. 9, New York, NY 10038, choerig@riversideresearch.org), Sally McFadden (Vision Sci., Hunter Medical Res. Inst. and School of Psych., College of Sci., Eng., and Environment, The Univ. of Newcastle, New Castle, New South Wales, Australia), Quan V. Hoang (Singapore Eye Res. Inst., Singapore, Singapore), and Jonathan Mamou (Riverside Res., New York, NY)

Myopia alters the collagen microstructure of the sclera which changes the tissue biomechanical properties. Although most forms of myopia are mild, progression of the disease to high or pathologic myopia can lead to significant weakening of the sclera and greatly increase the risk for the formation of a posterior staphyloma, a sight-threatening condition. Potential treatments to counteract effects of myopia on the sclera include chemical crosslinking (CXL). Formaldehyde releasing agents like sodium hydroxymethylglycinate (SHMG) are an attractive option for the posterior sclera because they increase tissue stiffness and require no light activation. Previous research has demonstrated that high-frequency quantitative ultrasound (QUS) is sensitive to the microstructural changes in the sclera caused by myopia. In this study, we investigate the ability of QUS to monitor microstructural changes caused by CXL. *Ex vivo* pig eyes were immersed in a 26.5 mM SHMG solution and scanned by an 80MHz ultrasound transducer at regular time intervals. RF echo data were analyzed to compute QUS parameters from the backscatter coefficient, normalized power spectrum, and echo envelope statistics. Correlations between QUS parameters and immersion time were investigated to evaluate the efficacy of QUS for monitoring CXL treatment. Results of this study demonstrate the capability to high-frequency QUS to assess changes in tissue microstructure caused by CXL with SHMG.

1:40

3pBAb3. Non-invasive ultrasound stimulation on the retina and visual cortex for visual restoration. Gengxi Lu, Xuejun Qian, Runze Li (Biomedical Eng., Univ. of Southern California, Los Angeles, CA), Biju Thomas (USC Roski Eye Inst., Univ. of Southern California, Los Angeles, CA), Mark S. Humayun (USC Roski Eye Inst., Univ. of Southern California, Los Angeles, CA), and Qifa Zhou (Biomedical Eng., Univ. of Southern California, 1042 Downey Way, Los Angeles, CA 90089, qifazhou@usc.edu)

Nearly 30% of the U.S. population aged over 75 have age-related macular degeneration and retinitis pigmentosa. 10% of them may become legally blind. Currently, blindness cannot be cured and patients' living quality can be compromised severely. Electrical visual prostheses are used as the best near-term strategy. However, they have challenges including invasive devices, complex implantation surgeries, and limited resolution. Non-invasive ultrasonic neuromodulation can be a promising technology for non-invasive visual prostheses. Our work showed that direct ultrasound stimulation on the retina can evoke neuron activities in either normal-sighted or retinal degenerated blind rats *in vivo*, indicating a promising future of ultrasound stimulation as a novel and non-invasive visual prosthesis for translational applications in blind patients. For patients with damaged optic nerves, retinal prostheses are not effective. Thus, we also investigated the feasibility of transcranial focused ultrasound for non-invasive stimulation of the visual cortex to develop a non-invasive cortical visual prosthesis. Besides stimulations, ultrasound has been used to non-invasively monitor brain activity by imaging cortical blood flow. Our study shows that ultrasound-induced visual responses can be monitored by ultrasound flow imaging of the brain, constructing an ultrasound stimulation-recording closed-loop system.

2:00

3pBAb4. Novel 3D ultrasound imaging and automated deep learning analysis in ophthalmology. Mahdi Bayat (Case Western Reserve Univ., 10900 Euclid Ave., Cleveland, OH 44106, mxb871@case.edu), Ahmed T. Minhaz, David L. Wilson (Case Western Reserve Univ., Cleveland, OH), and Faruk Orge (UH Cleveland Medical Ctr., Cleveland, OH)

We present novel 3D ultrasound technologies for imaging of the eye. We computationally align and improve imaging volumes acquired by mechanical scanning of conventional ophthalmic ultrasound systems. We also present results of using emerging ultrahigh frequency array systems that increase volumetric image reconstruction speed by several orders of magnitude and allowing Doppler processing of blood flow. The easily interpretable 3D visualizations can greatly assist in automated analysis and reducing operator dependency. We present deep learning methods for volumetric segmentation and quantification. The 3D ophthalmic ultrasound combined with automated deep learning analysis allows non-expert users to easily use this technology for diagnosis, monitoring response to therapy and improving overall ocular disease management at low cost, versatility, and safety for repeated and timely scanning.

Contributed Paper

2:20

3pBAb5. Automated *in vivo* detection and localization of posterior staphyloma in ultrasound b-modes. Theresa H. Lye (Riverside Res., 156 William St., Fl. 9, New York, NY 10038, tlye@riversideresearch.org), Kazuyo Ito, Daryle Jason G. Yu, Yee Shan Dan, Isabella Q. Loh (Singapore Eye Res. Inst., Singapore, Singapore), Ronald H. Silverman (Dept. of Ophthalmology, Columbia Univ. Irving Medical Ctr., New York, NY), Quan V. Hoang (Singapore Eye Res. Inst., Singapore, Singapore), and Jonathan Mamou (Riverside Res., New York, NY)

In high and pathologic myopia eyes, staphyloma, an outpouching of the eyewall, is associated with risks of permanent vision loss. Automated detection of staphyloma in ultrasound B-modes could assist clinicians in quickly identifying patients at risk. In this study, an algorithm to detect the presence of staphyloma and locate the staphyloma apex was developed. A 10 MHz ultrasound probe was used to acquire 127 B-modes, from 82 eyes and 52 patients. Sixty-one of the B-modes included staphyloma, as determined by an experienced ophthalmologist who also marked the staphyloma apex in each image where present. In each B-mode, the vitreoretinal boundary was automatically detected by a thresholding method. The local radius of curvature (K) was computed, and the ability of the standard deviation of K to detect staphyloma was evaluated by receiver-operating characteristic analysis. The staphyloma apex was automatically detected as the point on the vitreoretinal boundary furthest from the center of the transducer. The area under the curve using standard deviation of K for staphyloma detection was 0.927. The average error in detected staphyloma apex location was 1.34 ± 1.36 mm. These results demonstrate the potential of automated staphyloma detection and localization from ultrasound, which could assist in quick diagnosis of staphyloma.

Session 3pCA

Computational Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, Acoustical Oceanography, and Signal Processing in Acoustics: Showcases of High Performance Computing in Acoustics II

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

Shung H. Sung, Cochair
Troy, MI

Ralph T. Muehleisen, Cochair
Energy Systems, Argonne National Laboratory, 9700 S. Cass Ave., Bldg 362, Lemont, IL 60439-4801

Chair's Introduction—1:00

Contributed Papers

1:05

3pCA1. The noise dose range of the X-59 estimated from propagation simulations. William Doebler (NASA Langley Res. Ctr., NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov) and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

NASA's X-59 aircraft will soon be flown in a series of low-boom community noise surveys across the USA. Community test experimental designs will include noise dose schedules, which require an estimate of the noise dose range that X-59 can produce. To determine the dose range, propagation simulations using NASA's PCBoom code were completed using CFD-generated X-59 C612A near-field pressure data for an on-design condition (Mach 1.4 at 53 200 ft) and two off-design conditions (Mach 1.4 at 46 000 ft and Mach 1.3 at 43 000 ft). Near-field data at four cardinal aircraft headings were propagated through one year of realistic atmospheric profiles from the Climate Forecast System Version 2 database at several USA locations. Statistics of the mean and range of the Perceived Level (PL) of the booms in the inner 40 km of each carpet as well as the carpet width were computed. Results indicate the typical inner carpet mean PL is approximately 70–87 dB depending on the aircraft condition. The typical PL range across the inner carpet is 6–8 dB (on-design) and 5–13 dB (off-designs). The carpet widths ranged from 32–55 km. Understanding these quantities for various flight and atmospheric conditions is important for planning X-59 community noise surveys.

1:20

3pCA2. A new hybrid method between equivalent source method and boundary element method for modeling diffraction. Vincent Roggerone (Pôle Onde, Laboratoire de Mécanique et d'Acoustique, 4 impasse Nikola Tesla, Marseille 13013, France, roggerone@lma.cnrs-mrs.fr), Régine Guillermin (Pôle Onde, Laboratoire de Mécanique et d'Acoustique, Marseille, France), and Sandrine T. Rakotonarivo (Aix-Marseille Univ. and Lab. of Mech. and Acoust., Marseille, France)

The Equivalent Source Method (ESM) allows to model the acoustical radiation of an object with a set of monopole sources inside the object. This method efficiently describes acoustical radiation of an object with smooth surface, but is not performant for addressing diffraction induced by sharp edge. In this study, we use the simple source formulation to establish ESM as an integral formulation. In this form, ESM belong to the Boundary Element Method (BEM) family, with an implicit mesh. Therefore, we propose an alternative method similar to ESM with sources on the surface object, as the singularities can now be handled with the integral formulation. This new method is more performant than ESM for modeling diffraction, and is easier to implement than standard BEM based on Gauss Quadrature while it still takes advantage of the ESM implicit mesh. Comparisons of this hybrid method with ESM and standard BEM approaches show that the developed method is a good compromise between the two other approaches: the new method is more accurate than BEM, but less than ESM for a smooth object; while it is equivalent to BEM and more accurate than ESM for an edgy object. Work Funded by AMIDEX foundation.

1:35

3pCA3. A three-dimensional multi-grid staggered finite-difference time-domain method for underwater target acoustic scattering. Roberto Sabatini (Embry-Riddle Aeronautical Univ., 1 Aerosp. Blvd, Daytona Beach, FL 32114, sabatinr@erau.edu), Yan Pailhas, Giorgio Urso (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy), Paul Cristini (Laboratoire de Mecanique et d'Acoustique, Marseille, France), Angeliki Xenaki, and Alessandra Tesei (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

Solving the linearized equations of continuum mechanics to model the acoustic scattering of targets close to the seafloor is a challenging task. Among the traditional algorithms, staggered second-order finite-difference time-domain (FDTD) schemes combine simplicity and ease of implementation and parallelization, especially on modern GPUs. They have been successfully applied in two-dimensional configurations. Their major drawback is their inherent low resolution, due to their formal low order and the stair-step representation of the interfaces between media (e.g., target/water). To avoid excessive computational costs in three-dimensional configurations, while maintaining high numerical accuracy, we propose a FDTD multi-grid method. The mesh is refined locally near the interfaces, while a coarse grid is employed in homogeneous regions. In the fine grids, the aforesaid FDTD is employed, whereas, in the coarse grid, higher-order staggered schemes are used to compensate for the decrease in resolution due to the larger spacing. High-order interpolation allows matching the solutions at the grid interfaces. In this presentation, the performances of the new method on classical benchmark tests are analyzed and discussed. The numerical and computational capabilities are then illustrated by applying the proposed technique to the scattering of buried objects.

1:50

3pCA4. Implementation and analysis of high-performance cloud computing for underwater acoustic modeling applications. Alex Higgins (Elec. & Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 25-01, Portland, OR 97201, higginja@ece.pdx.edu) and Martin Siderius (Elec. & Comput. Eng., Portland State Univ., Portland, OR)

In the last few years advances in compute speeds have plateaued and multi/parallel processing (MPP) has been the way forward. With the advent of multiple processor cores packaged within a single microchip the architectures that were restricted to large servers has crossed over into consumer workstations. Now, more than ever, an understanding of how to maximize computational work is needed. The design and implementation of a Cloud based parallel-compute cluster for very large-scale processing of underwater acoustic models is presented. Ray-based, parabolic equation and wave number integration models were evaluated. This type of system allows for the creation of real-world sized solution sets which were previously prohibitive for most researchers. This is mainly due to their excessively long runtimes or being restricted to gated hardware, such as super-computer systems. Amazon Web Services is used as the underlying technologies for the creation of the Cloud based parallel-computer cluster, but the principles discussed are universal and may be applied to any MPP system. [Work supported by the Office of Naval Research.]

2:05–2:20 Break

2:20

3pCA5. The ongoing sound change between /l/ and /n/ in Gan Chinese: Random forest classification. 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)

The contrast between /l/ and /n/ in syllable-initial position may be absent or not consistently maintained across sub-dialects of Gan Chinese. To see if the /l/ and /n/ contrast is acoustically differentiated among speakers of Gan Chinese, a set of acoustic measurements were extracted from their production of words beginning with /l/ and /n/ and submitted to random forest classification. The extracted acoustic measurements were: (1) consonantal duration; (2) frequency and formant intensity of F2 and F3; (3) H1*-H2*, H1*-A1, and H1*-A2; (4) A1-P0 and A1-P1 measurements (at the 25% point of the following vowel). Post vowel context binary: /i/ and non-/i/. All tokens (/l/: 73%; /n/: 27%) were partitioned into a training set (953 tokens) and a test set (318 tokens). After training with the training set, the final classifier was applied to the test set to evaluate the categorical prediction of /l/ and /n/ with a classification probability. The results obtained indicated that most variants of /n/ would be categorized as /l/, suggesting an ongoing sound change between /l/ and /n/ that may be influenced by the vowel context.

2:35

3pCA6. Dynamically orthogonal ray equations with adaptive reclustering. Aaron Charous (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02142, acharous@mit.edu), Michael J. Humara, Wael H. Ali, Manmeet S. Bhabra, Abhinav Gupta, and Pierre F. Lermusiaux (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

The vast size of the ocean coupled with the Nyquist criterion often makes high-frequency ocean acoustics computationally intractable without approximation. A common approach is to employ ray theory to develop an approximate Green's function for the acoustic wave equation. But due to the dynamic nature of ocean fields and the many sources of uncertainty in the temperature, salinity, density, sea surface, bathymetry, neither the sound speed nor the source and receiver locations are precisely known, and a single deterministic solution of the ray equations is often insufficient. In place of Monte Carlo simulations, we develop and numerically solve dynamically orthogonal ray equations to efficiently approximate solutions to the stochastic acoustic wave equation. The Lagrangian nature of the ray equations introduces a new computational challenge for reduced-order models; a naive approach requires realizations to be reconstructed at each time step. To remedy this, we employ a Taylor expansion of the sound speed around "mean rays" and propose a novel, adaptive algorithm to determine when the modes and coefficients of the reduced-order model must be reclustered and/or augmented to preserve accuracy. This offers a unique methodology for uncertainty quantification in the presence of non-Gaussian statistics which may be utilized in forward modeling as well as in acoustic inverse problems.

2:50

3pCA7. Optimal reduced-order solution to the 3D deterministic parabolic wave equation. Aaron Charous (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02142, acharous@mit.edu) and Pierre F. Lermusiaux (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

In weakly scattering media, the parabolic wave equation is often utilized to develop approximate solutions to the Helmholtz equation. However, in three dimensions, at high frequencies, and when the computational domain becomes very large, computational cost grows quickly. We propose using a reduced-order modeling approach for the deterministic 3D parabolic wave equation via the dynamical low-rank approximation. This approach is similar in spirit to the normal-mode technique; however, instead of using fixed modes for the entire computational domain, we instantaneously optimally evolve the modes in range according to the PDE dynamics. This dynamic order reduction provides a more accurate low-rank approximation to the parabolic wave equation. We demonstrate the efficacy of this technique on realistic ocean acoustic test cases.

3:05

3pCA8. Generative deep learning model for broadband acoustic metamaterial design. Feruza Amirkulova (Mech. Eng., San Jose State Univ., 1 Washington Sq, San Jose, CA 95192, feruza.amirkulova@sjsu.edu), Thang Tran, and Ehsan Khatami (Phys. and Astronomy, San Jose State Univ., San Jose, CA)

This study presents a promising data-driven approach for approximating and minimizing acoustic wave scattering in 2D acoustic cloak design, in which the optimal arrangement of cylindrical structures is found via a combination of generative deep learning models and optimization based on data generated by multiple scattering theory. In forward design, a fully connected neural network and convolutional neural network is trained with binary images of planar configurations of rigid cylinders to predict the total scattering cross section (TSCS) at discrete values of normalized wavenumber. In inverse design, generative deep learning models are developed that encode cylindrical structures into a continuous representation called a latent vector, and new structures are generated by decoding random latent values in this representation. In these models, the variational autoencoders, supervised and unsupervised learning, and Gaussian process are combined to predict optimal arrangements of scatterers for the desired minimal TSCS. Comparison of the scattered wavefield between the generated configurations and the numerical result shows the validity and effectiveness of the proposed method.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Aaron Gunderson, Chair
Applied Research Laboratories, The University of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758

Chair's Introduction—2:20

Invited Papers

2:25

3pID1. The effect of COVID-19 on underwater sound. David R. Barclay (Dept. of Oceanogr.,Dalhousie Univ., 1355 Oxford St., LSC Bldg., Halifax, NS B3H 4R2, Canada, dbarcl@gmail.com)

The sounds of the turning machinery and propulsion systems from commercial ships are ubiquitous in the temperate deep ocean basins. The vertical structure of seawater temperature, salinity, and pressure conspire to create nearly lossless underwater propagation conditions at 10's and 100's of Hz, the frequency band where diesel engines and generators, cavitating propellers, and turning gearboxes produce their peak acoustic energy. In the shallow waters of the continental margins, the human use of the ocean for transport, fishing, recreation, and construction has led to increasing levels of anthropogenic noise over the last century. As the COVID-19 pandemic spread across the globe, the reduction of economic activity and marine traffic resulted in a decrease in underwater noise, with observations made at a number of underwater listening stations and autonomous recording devices across the ocean basins. These measurements, along with ship track data provided by the automated identification system (AIS) and trade and shipping statistics, improve our knowledge of the relationship between acoustic energy, the marine acoustic habitat, and the various human uses of the ocean.

2:45

3pID2. Making music during a pandemic. Thomas R. Moore (Dept. of Phys., Rollins College, Box 2743, Rollins College, Winter Park, FL 32789, tmoore@rollins.edu)

Efforts to mitigate the spread of COVID-19 among musicians have recently produced a significant body of original research on the flow of the breath while making music. Percussion and string instrument players have no special considerations beyond those of the general population in reducing the spread of COVID-19, but the other musical instruments involve the exhalation of breath through the instrument toward other players and the audience. Similarly, singing produces both aerosol and ballistic droplets from the mouth and the louder one sings the greater the distance the exhaled particles travel. This presentation will review recent attempts to understand the flow of breath during musical performance, as well as the acoustical effects of some of the methods that musicians have used to reduce the spread of COVID-19 during rehearsal and performance.

3:05

3pID3. COVID-19 effects on social isolation for older persons with sensory loss. Peggy Nelson (Ctr. for Applied/Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, nelso477@umn.edu), Walter Yueh-Hsun Wu, Kristi Oeding, Elizabeth Anderson, Katherine Teece, and Gordon Legge (Ctr for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN)

Social isolation and loneliness are significant risk factors for overall health, especially for older adults with vision or hearing impairment. This study was designed to understand the impact of the COVID-19 pandemic restrictions on the social /emotional wellbeing in adults with sensory impairment. Three groups of older adults – vision loss (VL, $N = 13$), hearing loss (HL, $N = 12$ with cochlear implants and $N = 12$ with hearing aids), and controls (CTL, $N = 18$) – were recruited from the Twin Cities community. They were interviewed at 6-week intervals from April 2020 to July 2021 concerning their wellbeing and social isolation. Overall, results show the widespread impact of the pandemic on social interactions and emotional wellbeing of older adults with sensory loss. All three groups had worse scores on the patient health questionnaire (PHQ-9) after the start of the pandemic. Older people with visual impairments were especially vulnerable to changes that led to a loss of independence. Although facial coverings impaired speech intelligibility for older people with hearing loss, the quieter acoustic environment during stay-at-home restrictions of gatherings may have mediated the negative effects. Results highlight unique factors that have affected older persons with sensory loss.

Session 3pMU

Musical Acoustics: General Topics in Musical Acoustics II

Christopher Elmer, Chair
American Mathematical Society, 17231 Lands End, Chelsea, MI 48118

Contributed Papers

1:30

3pMU1. A computational method to determine transition times for soprano and bass clarinet players. Whitney L. Coyle (Rollins College, 1000 Holt Ave.—2743, Winter Park, FL 32789, wcoyle@rollins.edu), Evangelina Y. Wong, Jack D. Gabriel, and Connor N. Kaplan (Rollins College, Winter Park, FL)

This work defines transition times between articulated notes for reed instruments such as clarinets and bass clarinets. To determine these transition times, analysis of measurements of musician mouthpiece pressure, measured using sensor-equipped mouthpieces made especially for the clarinet and bass clarinet is required. The transition time (△T) can then be compared in different musical contexts (dynamics, tempos, etc.), playing regimes, and between various players. The data collection method and metric definition will be described, and results from different playing situations will be offered.

1:45

3pMU2. The lowest frequency modes of the mandolin modeled as a coupled oscillator system. Connor Robertson, Gretchen Schowalter (Phys., Lewis & Clark College, Portland, OR), and Steve Tufte (Dept. of Phys., Lewis & Clark College, MSC 15, 0615 SW Palatine Hill Rd., Portland, OR 97219, tufte@lclark.edu)

The mandolin is an acoustically unique musical instrument with elements similar to the guitar and violin. Inspired by a classic guitar study by Christensen and Vistisen, we tested if the mandolin's low-frequency response could be modeled as a simple 3-mass coupled oscillator system. We measured the response of the front plate, back plate, and air within the cavity from sinusoidal forcing of the front plate using microphones, accelerometers, and a force probe. We calculated the mobility (i.e., 1/impedance = v/F) and compared this to a theoretical model. To determine how the collective resonances are affected by the individual oscillators in the system, we added small masses to the front and back plates, and "collared" the f-holes to add mass to the moving air, thus lowering the resonances of the corresponding oscillator. We found clear evidence of coupling: by changing any one element, the resulting three resonant frequencies of the coupled system were affected. We also found the expected phase relations between the individual oscillators for each of the three resonances of the coupled system. From a detailed analysis, we found that a 3-mass coupled oscillator model can reasonably approximate the low-frequency behavior of the mandolin.

2:00

3pMU3. Algebraic method for shaping percussive bars. Greg Elliott (Phys., Univ. of Puget Sound, 1500 N Warner, CMB 1031, Tacoma, WA 98416, gelliott@pugetsound.edu)

In the tuning optimization problem for percussive bars, shape profiles are parameterized in terms of some appropriate set of functions, and parameters are sought that produce desired mode frequencies. For a specific bar shape, the bending mode frequencies can be accurately determined within the Timoshenko-Ehrenfest model using a finite element analysis. Under certain conditions, a useful set of functions that reduces the differential equation to a purely algebraic problem can be constructed from solutions to the corresponding uniform system problem. These constructed functions vary on the same scale as the uniform beam bending modes and hence couple effectively to shift the mode frequencies. This algebraic methodology is applied to the percussive bar problem, and a Monte Carlo optimization is performed to determine shape functions that produce bars with the frequency ratios of the marimba (1:3:6), the xylophone (1:4:10), and an octave bar with ratios 1:2:4:8. Bars were constructed according to these designs, and acoustical measurements are in good agreement with model predictions.

2:15

3pMU4. Finite element modeling of guitar top plates. Joely Caisse (Phys., Whitman College, 345 Boyer Ave., Whitman College, Walla Walla, WA 99362, caissejl@whitman.edu) and Kurt R. Hoffman (Phys., Whitman College, Walla Walla, WA)

Here we describe the results of finite element method modeling of the resonant vibrations of a guitar top plate in various stages of construction using COMSOL. These models were completed in parallel with electronic speckle pattern interferometry ESPI measurements on three different top plates made of Engleman spruce purchased from Stewmac. The modeling was first set up for a thin piece of wood in a rectangular shape. Subsequent modeling was done at each step as the bode shape was cut, then a tone hold cut, and finally bracing was added. Care was taken to rotate the stiffness matrices for the braces in the model. Normal mode eigenfrequencies and mode shapes were calculated at each step in the fabrication process. Good agreement between the model and the measured data was obtained at low frequencies, but the model deviated from the measured results at higher frequencies.

2:30

3pMU5. Finite element simulations of free reed instrument operation. Philip Kaufinger (Lehigh Univ., Lehigh University, Bethlehem, PA 18015, pgk222@lehigh.edu), Liam Wynne (Middlebury College, Middlebury, VT), and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

To facilitate research into the acoustical properties of free reed instruments, there is a need for a quick and inexpensive way to prototype new geometries. It is here that a computational model with easily adjustable parameters offers a substantial advantage over an experimental apparatus if it can sufficiently resolve the essential acoustic features. Simulations using the finite element method have been utilized to solve the fluid dynamics, pressure acoustics, structural mechanics, and the multi-physics couplings underpinning various free reed systems. These phenomena include attack transients and frequency-pressure dependence of Western free reeds, fluid-structure interaction of turbulent airflow in an Asian free reed, effects of air compressibility on reed vibration, and the acoustic impedance of a reed-driven khaen pipe. Results consistent with experimental and analytical models were obtained and methods of optimizing the computational cost while maintaining realism are proposed. The models fail, however, to capture the essential feature of free reed sound production via the periodic interruption of airflow through the reed plate. Ways for future research to

achieve this benchmark are suggested. [Research supported by National Science Foundation, Grant NSF-REU-1950337.]

2:45

3pMU6. Hybrid lattice-Boltzmann digital-waveguide simulation of wind instruments. Song Wang (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Rue Sherbrooke Ouest, Montreal, QC H3A 1E3, Canada, song.wang5@mail.mcgill.ca) and Gary Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., Montreal, QC, Canada)

A hybrid lattice-Boltzmann digital-waveguide model is proposed to improve the efficiency of the computational aeroacoustics modeling of wind instruments. The nonlinear sound generator and the assumed linear resonator of a wind instrument are modeled separately using the lattice Boltzmann (LB) method and digital waveguide (DWG) method, correspondingly. Their coupling is achieved through the characteristic boundary condition, which breaks down the lattice Boltzmann solutions at the junction into separated traveling waves along different characteristic lines. The lattice Boltzmann solver then sends the outgoing acoustic wave to the digital waveguide, and receives the incoming acoustic wave to complete the boundary condition. Two examples are tested to demonstrate the validity of the proposed method, including a hybrid cylindrical pipe composed of two coupled cylindrical segments, one represented by LB and the other by DWG, and a hybrid single-reed instrument that comprises an LB mouthpiece-reed system and a DWG pipe.

WEDNESDAY AFTERNOON, 1 DECEMBER 2021 — 301 (L)/304 (O), 1:00 P.M. TO 2:40 P.M.

Session 3pNS

Noise, Physical Acoustics, ASA Committee on Standards, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Military Jet Noise Measurements and Analysis

Caroline P. Lubert, Cochair
Mathematics and Statistics, James Madison University, 301 Dixie Ave., Harrisonburg, VA 22801

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

Chair's Introduction—1:00

Invited Paper

1:05

3pNS1. Microphone placement investigation for standard aircraft ground run-up noise measurements. Alan T. Wall (Sensory Systems Branch, Air Force Res. Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Steven C. Campbell (Ball Aerosp., Beavercreek, OH), and Frank S. Mobley (Sensory Systems Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

The correct placement (height, orientation, and mounting techniques) for microphones used in standard data collection practices for aircraft environmental noise models is a topic of frequent debate. Previous studies have demonstrated and quantified the dependence of spectral data recorded as a function of microphone placement variations and relative source origin. In this work, an in-depth investigation of microphone mounting effects is performed with an emphasis on (1) microphone height and (2) ground-mounting configuration for the purpose of quantifying ground run-up aircraft noise emissions for environmental noise model generation. Data for this investigation were collected on the T-7A Red Hawk in both standard (see ANSI S12.75-2012) and off-standard configurations for comparison.

1:25

3pNS2. Far-field characteristics of installed F404 engine noise. Matthew A. Christian (Dept. of Mech. Eng., Brigham Young Univ., ESC N284, Provo, UT 84602, matthew.christian@byu.edu), Logan T. Mathews, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Jon P. Johnson (Dept. of Phys., Brigham Young Univ. - Idaho, Rexburg, ID), and Steven C. Campbell (Ball Aerosp., Beavercreek, OH)

Noise measurements have been recently made of a T-7A-installed GE F404 engine in both the near and far-fields (Leete *et al.*, AIAA paper No. 2021-1638). This paper describes far-field analyses of data collected along a 76 m (250 ft) arc at inlet angles spanning 30 to 160 degrees, at engine conditions ranging from intermediate through full afterburner. From calculated overall and frequency-dependent directivity curves, far-field noise properties are obtained as a function of engine condition, including change in peak directivity angle and associated spectral properties, radiation lobe width, overall sound power level, and sound power spectra. The results guide physical insights regarding changes in jet characteristics with engine condition as well as serve as benchmark data for comparing with laboratory and numerical datasets. [Work supported by ONR.]

1:40

3pNS3. Similarity spectra-based decomposition of noise from an installed F404 engine. Cooper D. Merrill (Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, cdouglasmerrill@gmail.com), Kristi A. Epps, Kevin M. Leete (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Steven C. Campbell (Ball Aerosp., Beavercreek, OH)

Similarity spectra analysis has long been a popular method for identifying both the fine-scale and large-scale components of jet noise. For an imperfectly expanded supersonic jet, addition of a broadband shock-associated noise (BSN) model allows spectra to be better represented. A similarity spectra analysis of T-7A-installed GE-F404 engine noise has been performed at conditions ranging from intermediate to full afterburner. The BSN contributions are significant at military and afterburner conditions. The combined model captures the noise reasonably well with some caveats. First, spatiospectral lobes present in the measured noise are not well represented. Second, the measured low-frequency spectral slope is steeper at higher engine conditions in the region of maximum radiation. Third, the measured high-frequency slope is shallower across most of the radiation angles. Fourth, the noise at large inlet angles, beyond the peak radiation lobe, are not well represented by the combined model. The successes (and failures of the model) for different spatial regions and frequencies will aid in developing improved models for noise radiation. [Work supported by ONR.]

1:55

3pNS4. Near-field acoustical holography-based noise source analysis of an installed, afterburning F404 engine. Logan T. Mathews (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmathew3@byu.edu), Kevin M. Leete, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

Near-field acoustical holography has been used previously to study characteristics of jet noise radiated from tactical aircraft. This paper investigates frequency-dependent source and radiation characteristics of an installed GE F404 engine at afterburner, as determined using multisource statistically optimized near-field acoustical holography. Included in the analyses are the number of partial fields required to represent the source, the extent of the region of maximum sound power production, and directional characteristics. Comparisons are made with other source reconstructions of high-performance engines. [Work supported by ONR.]

2:10

3pNS5. Spatiospectral lobe characteristics for F404 engine noise. Tyce Olaveson (Phys. and Astronomy, Brigham Young Univ., N284 Eyring Science Ctr. BYU, Provo, UT 84602, tyceolaveson@gmail.com), Jacob A. Ward (Phys. and Astronomy, Brigham Young Univ., Salt Lake City, UT), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Jon P. Johnson (Phys., Brigham Young Universtiy-Idaho, Rexburg, ID), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

Spatiospectral lobes are a currently unexplained phenomenon seen in noise radiation from tactical aircraft. These lobes can be seen as either multiple peaks in the noise spectra at a given field location or multiple local maxima in the directivity when observing the radiated noise field at a particular frequency. Using hybrid beamforming of a 120-microphone near-field array, the characteristics of these lobes are studied for a GE F404 engine installed on a T-7A aircraft and at different engine conditions. Both the measured field and field reconstructions show multiple spatiospectral lobes. While the overall directivity of the noise shifts forward with increase in frequency, individual lobes appear, shift aft, and then disappear as frequency is increased. Individual lobes were also raytraced back to the jet centerline to determine an apparent acoustic source location. For frequencies where multiple lobes are present, they generally radiate from an overlapping maximum source region that moves upstream with frequency. This corroborates the characteristics of spatiospectral lobes observed in other aircraft. [Work supported by ONR.]

2:25

3pNS6. Optimized metal foam liners with spatially gradient properties for aircraft engine noise reduction. Amulya Lomte (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, axlomte@shockers.wichita.edu), Manish Mahajan, and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Recent studies conducted by NASA have identified over the rotor liners as one of the most promising avenues for aircraft noise reduction. However, conventional noise absorption technologies cannot withstand the harsh pressure and temperature conditions near the engine core. While open celled metal foams are an alternative material choice, their open porous structure results in comparatively lower absorption characteristics. In this talk, we summarize the results from our recent efforts on improving the acoustical properties of metal foams for aircraft noise reduction applications. We show that compressing the foams substantially improves their absorption coefficient because of the reduction in the effective pore size. Unfortunately, this improvement comes with an undesirable weight penalty. To overcome this, we propose an optimized layered configuration to create a step-wise relative density gradient that provides comparable absorption characteristics while being over 20% lighter than the benchmark compressed foam. Our results show that metal foam liners with spatially gradient property configurations can provide broadband engine noise reduction while minimizing the resulting weight penalty.

Session 3pPA

Physical Acoustics, Signal Processing in Acoustics, ASA Committee on Standards, Engineering Acoustics, and Computational Acoustics: Infrasound II

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

Philip S. Blom, Cochair
Earth & Environmental Sciences, Los Alamos National Laboratory, PO Box 1663, M/S F665, Los Alamos, NM 87545

Contributed Papers

1:00

3pPA1. On cataloguing the possible waveforms for impulse propagation along ground to ground thermospheric paths. Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., Los Alamos, NM) and Roger M. Waxler (Univ. of MS, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu)

Infrasound propagation in the thermosphere is characterized by very low mean density, a very steep mean temperature gradient, and tidal winds. The steep temperature gradient means that there is always a thermospheric duct and this always provides ground to ground propagation paths that pass through the thermosphere. These paths are modulated by the atmospheric tides, while the very low density in the thermosphere causes severe attenuation and non-linear distortion. For an impulsive signal initially in the form of an N-like wave, such as the signal produced by a large explosion, non-linear propagation effects cause the positive pressure phases to propagate more rapidly than the negative pressure phases, in turn causing the duration of the impulse to increase. This effect can be quite significant in the thermosphere and continues until a caustic is encountered transforming the N-wave into a U-wave. The severe attenuation causes the high frequency components of the signal to attenuate resulting in a long period and fairly clean waveform that propagates back to the ground. The combination of steep temperature gradient and atmospheric tides produce a canonical thermospheric propagation environment with a diurnal cycle. In this presentation, it is shown that non-linear ray theory (weak shock theory) can be used to catalogue the waveform shapes of ground returns from explosive signals traveling along thermospheric paths. Qualitative comparisons to observed thermospheric returns are made.

1:15

3pPA2. An investigation of dissipation processes in a multi-component gas-mixture: Application to acoustic-wave absorption in the thermosphere-ionosphere. Benedict Pineyro (Embry-Riddle Aeronautical Univ., 1 Aerosp. Blvd, Daytona Beach, FL 32114, pineyrob@my.erau.edu), Roberto Sabatini, and Jonathan Snively (Embry-Riddle Aeronautical Univ., Daytona Beach, FL)

Infrasonic waves generated in the lower atmosphere can propagate into the thermosphere and perturb the ionosphere. These disturbances are accompanied by a change in the ionospheric electron density, which can be detected via remote sensing methods, such as Global Navigation Satellite System derived measurements of integrated total electron content. The altitude-dependent decrease in density strengthens dissipative phenomena which affect these measurements by reducing the acoustic-wave amplitude. Sutherland and Bass [https://doi.org/10.1121/1.1631937] have described atmospheric absorption, but only up to 160 km, and neglected interspecies diffusion. However, the atmosphere above the mesopause is a mixture of three major gases, namely, N_2, O_2, and O, where processes associated with mass-fraction-density gradients could affect acoustic-energy dissipation. This work revisits absorption processes via numerical simulations of the equations of fluid mechanics for multicomponent-gas mixtures, under the assumption of a small Knudsen number (i.e., satisfying continuum approximation). More specifically, diffusion due to mole-fraction, pressure, and temperature gradients, and heat diffusion due to concentration gradients are included alongside classical thermo-viscous terms. Their impact is investigated on vertically-propagating acoustic pulses with varying frequencies, <1 Hz, and different amplitudes that match typical values observed from geophysical and anthropogenic sources (e.g., earthquakes, thunderstorms, explosions).

1:30

3pPA3. Infrasound from orbital space vehicle launches. Samuel A. Olausson (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, samolausson@gmail.com), Michael S. Bassett, Griffin Houston, Logan T. Mathews, Mark C. Anderson, J. T. Durrant, Grant W. Hart, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Launches of medium and heavy-lift space vehicles produce high-amplitude, broadband noise environments with significant infrasound. Measurements taken at Vandenberg Space Force Base during Falcon 9 and Delta IV Heavy launches allow for a comparison of infrasound between rockets and environments. For the Falcon 9, low surface wind and high signal-to-noise ratios allowed for the observation of infrasound all the way through main engine cutoff. In contrast, the Delta IV Heavy measurements featured much higher surface winds and lower signal-to-noise ratios. However, even during the Delta IV Heavy measurement, infrasound was recorded for several minutes after liftoff. Characteristics as a function of distance and time are discussed.

1:45

3pPA4. Infrasound signature measurements for U.S. Army infantry weapons during training. Alessio Medda (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, alessio.medda@gtri.gatech.edu), Walter Carr (Walter Reed Army Inst. of Res., Silver Spring, MD), Robert Funk (Georgia Tech Res. Inst., Smyrna, GA), Bradley Garfield (Walter Reed Army Inst. of Res., Silver Spring, MD), and Krish Ahuja (Georgia Tech Res. Inst., Smyrna, GA)

United States Military personnel are exposed to blast overpressure from a variety of sources during training and military operations. While it is known that repeated exposure to high-level blast overpressure may result in concussion like symptoms, the effect of repeated exposure to low-level blast overpressure is not well understood yet. Impulsive acoustic sources such as pressure waves generated by explosions, artillery launches, and rocket launches are typically characterized by a broadband energy distribution with

resulting pressure measurements that exhibit frequency components well into the infrasound range. This study focused on data acquired in the audible and infrasound range during infantry weapons training for machine gun, grenade launcher and rocket fire events. Time, frequency and time-frequency analysis are used to quantify infrasound intensity for each event type. High resolution time-frequency analysis using the synchrosqueezed wavelet transform is used to precisely quantify energy content in the lowest frequency range. Preliminary results indicate that data collected from these weapon systems exhibited a large amount of infrasound in the range between 2 and 20 Hz. This energy can couple directly with the human body and in case of repeated exposures this coupling can include altering of physiological processes.

2:00–2:15 Break

2:15

3pPA5. Designing and building low-cost infrasound sensors to enable a large-N sensing array. William (Chip) Audette (Creare LLC, Hanover, NH, wea@creare.com), Chris Brooks (Creare LLC, Hanover, NH), Carrick Talmadge, Garth Frazier (NCPA, Univ. of MS, Oxford, MS), and Roger M. Waxler (Univ. of MS, University, MS)

Industrial and commercial pressure sensing elements have continued to improve in performance while also dropping in price. In this paper, we present our approach to designing and building infrasound sensor nodes using these off-the-shelf sensor elements. We then built a 50-node array and demonstrated its use in the field. Calibration results for the sensor nodes and initial field results are presented. With the feasibility of this approach demonstrated, we iterated the design of the system based on lessons-learned from the field experience. We fabricated a 300-node array to the revised design. System performance and cost are presented.

2:30

3pPA6. Infrasound sound source localization for early tsunami detection. Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), Haris Ihsannur, Bernadeta N. Eka Rini, and Muhammad R. Radityo (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Early detection of tsunami needed for sufficient evacuation time, which the existing method using buoy can not accommodate. In this report, we investigated low-frequency signal (infrasound) as a cue for tsunami detection. We used Bayesian infrasound source localization (BISL) method for predicting events and direction of the tsunami based on prior data. The method consists of three processes such as detection, association, and localization. All processes are done to infrasound data that are recorded from the microphone array. The recorded infrasound data at Minahasa, Indonesia used for evaluating the proposed method. The result indicates that BISL can predict latitude, longitude, and incident time of the tsunami based on the true value of the real tsunami event.

2:45

3pPA7. Seismometers as infrasound sensors. Richard D. Costley (U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil), Chris Hayward (Southern Methodist Univ., Dallas, TX), and Luis De Jesus Diaz (U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS)

An experiment was conducted in West-Central Mississippi in which five explosive charges were detonated. The TNT equivalent sizes of the charges ranged from 0.57 to 10.91 kg (1.25 to 24 lb). Among the arrays of sensors deployed, seismometers were deployed near microphones at distances of 0.5, 2.1, and 8.4 km from the source. The blast wave at these distances had decayed in amplitude to an acoustic wave. The coherence between the seismometer and microphone signals showed that the seismometer provided reasonable representation of the acoustic wave over a limited frequency band. This band changed with distance from the source. In addition, two 3-component seismometers were deployed near each other 8.4 km from the source. These seismometers were of different types, one having a resonance frequency of 1 Hz and the other 4.5 Hz. The signals from the horizontal components of these seismometers were analyzed to determine their effectiveness as vector sensors. The results showed that the back-azimuth determined from the seismometers agreed reasonably well with ground truth for the first arrival of the acoustic wavefront, however the results degraded as the trailing part of the wavefront passed. Permission to publish was granted by the Director, Geotechnical and Structures Laboratory.

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

H. S. Colburn, Chair
Boston, MA

Chair's Introduction—1:00

Invited Paper

1:05

3pPP1. Behavior, brain, the ear and development, and back again. Edwin Rubel (Virginia Merrill Bloedel Hearing Res. Ctr., Univ. of Washington, VMBHRC Box 357923, Seattle, WA 98195, rubel@uw.edu)

I am humbled and immensely honored to be awarded the 2020 William and Christine Hartmann Prize in Auditory Neuroscience. This talk will be composed of three stories, designed to honor the dedicated people who have shared my enthusiasm for trying to more fully understand development of the ear, the brain and hearing during the past half century. Story 1 will trace some of the questions we've pursued and why; from fundamental studies on behavior and brain development in neonatal chickens, to development of cochlear frequency maps, to hair cell death and regeneration and to understanding new details of binaural processing. Story 2 will discuss the rationale and some of the results underlying the 40+ years of studies on the biological interactions between the ear and the brain during development; interactions whereby activity shapes the structural and functional architecture of auditory pathways in the brain. Finally, Story 3 will describe our foray into the world of translational pharmacology, developing a new model to more fully understand hair cell death and using it to discover a new drug that may prevent hearing and balance disorders due to ototoxic medicines.

Session 3pSA

Structural Acoustics and Vibration, Education in Acoustics and Noise: Perspectives from Senior Researchers in Structural Acoustics and Vibration

Jason Smoker, Cochair
724, NSWCCD, 9500 MacArthur Blvd, West Bethesda, MD 20817

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

Chair's Introduction—1:00

Invited Papers

1:05

3pSA1. The system loss factor in sound and vibration. James McDaniel (Mech. Eng., Boston Univ., Boston University, Dept. of Mech. Eng., Boston, MA 02215, jgm@bu.edu)

In 1930, Kimball wrote a paper entitled The damping factor in vibrations that appeared in the journal *Product Engineering*. That paper proposed a damping factor based on the decrement of a vibrating system with exponential decay. Kimball showed that the proposed damping factor was proportional to the ratio of the energy dissipated per cycle to the total vibrational energy. The total vibrational energy was defined as the maximum potential energy over a cycle. Since that time, the definition of total vibrational energy has been more thoroughly examined because the total vibrational energy varies within a cycle due to damping. In this presentation, a new definition is proposed that defines the total vibrational energy as the temporal average of the total vibrational energy over a cycle. This definition has the advantage of reducing the system loss factor to the modal loss factor under certain limits. Since 1930, Kimball's damping factor has appeared in various forms under various names, the most common of which is the system loss factor. This presentation discusses these appearances in order to understand energy losses in dynamic systems caused by a wide array of physical mechanisms. The presentation concludes with a discussion of the system loss factor in present-day numerical simulations, highlighting its independence from eigenvalue information. [Work supported by ONR under Grant N00014-19-1-2100.]

1:25

3pSA2. Have I learned anything? Sharing my perspectives from a career in acoustics. Scott D. Sommerfeldt (Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

After graduation, a career of 30–40 years can seem like a long time to many of us, with so many questions to resolve. Where do I want to work – both in terms of company and geography? How do I get ahead—in my career and in life? How do I best contribute to advancing and balancing my personal ambitions, the mission of my company, professional opportunities, and so forth? And what about non-work activities? How do I best balance those? I cannot answer these questions for anyone, but in this presentation, I will share some of my perspectives on some of these questions, and what I have personally learned over many years of striving to contribute in my discipline, while also striving to live a fulfilled life.

1:45

3pSA3. Bridging cultures with acoustical materials research. Michael R. Haberman (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

This talk will provide a short overview and insights associated with the education and early career of the author over the past two decades and how his experiences with different cultures, both social and professional, has influenced scientific advancements. This will include a short overview of formal education in both the United States and France and professional growth as a researcher at the Applied Research Laboratories at The University of Texas at Austin (UT) and then as a professor in the Department of Mechanical Engineering at UT. The talk will highlight research done in each context and how those experiences have improved outcomes in scientific and personal contexts.

2:05

3pSA4. Hard-fought “pearls of wisdom” (?) from a senior structural acoustician. Robert M. Koch (U.S. Navy, 304 White Horn Dr., Kingston, RI 02881, Robert.M.Koch@navy.mil)

Throughout the course of a relatively long career in science and engineering, including advanced degrees, teaching, and a host of other external technical activities, there are numerous lessons learned and perspectives gained that are frequently much more important than specific technical advice. Oftentimes, the most important and critical teachings to be passed on to fellow present and future researchers are not narrowly-focused but wide-ranging and generally applicable. The most essential of these “pearls of wisdom” are learned occasionally out of successes yet more often out of apparent failures and frustrations. Almost invariably, the most valuable information is wide-ranging and universal and independent of any particular scientific discipline or field. This paper presents numerous technical and life-/career-related lessons that are hopefully of value and interest to current and upcoming scientists, engineers, and students. In addition, time permitting, a particular specific technical case study is presented.

2:25–2:30 Break

2:30–2:50
Panel Discussion

Plenary Session and Awards Ceremony

Maureen Stone,
President, Acoustical Society of America

Annual Membership Meeting

Introduction of Recipients of ASA Scholarships

Presentation of Certificates to New Fellows

Kyle M. Becker – For leadership in ocean acoustics

Mark A. Bee – For contributions to understanding amphibian bioacoustics

Jonas Braasch – For interdisciplinary contributions to musical acoustics and psychoacoustics of spatial audio technology

Matthew J. Goupell – For advancing understanding of binaural processing in electric and acoustic hearing

Brian T. Hefner – For contributions to scattering and reverberation in underwater acoustics

Brent Hoffmeister – For contributions to the ultrasound characterization of bone

Adrian K.C. Lee – For contributions to our understanding of auditory attention

Subha Maruvada – For contributions to ultrasound metrology

D. Benjamin Reeder – For advancements in underwater acoustic propagation and scattering

Bradley E. Treeby – For contributions to computational modeling in biomedical ultrasound

Richard A. Wright – For contributions to understanding how phonetic variability impacts communication

Pavel Zahorik – For contributions to understanding auditory perception in natural environments

Pei Zhong – For contributions to shock wave lithotripsy

Presentation of Science Communication Awards

Why Music Makes Us Feel, According to AI, Caitlin Dawson, *USC Viterbi news website*, 31 October 2019

How to Restore the Legendary Acoustics of Notre Dame, Emily Conover, *ScienceNews*, 12 January 2020

Ocean Uproar: Saving Marine Life from a Barrage of Noise, Nicola Jones, *Nature*, 10 April 2019

Science of Voice Acting, Colette Feehan, *Tik Toc*, 12 December 2020

Long Rang Acoustic Devices (LRAD) and Public Safety, Tyler Tracy, *Acentech Blog*, 10 August 2020

Vibrotactile Technology to Support d/Deaf People in Music Education, Carl Hopkins, University of Liverpool website, 6 March 2020

Introduction of Prize Recipients

2020 Medwin Prize in Acoustical Oceanography to Kelly Benoit-Bird

2021 Medwin Prize in Acoustical Oceanography to Anna Širović

Hartmann Prize in Auditory Neuroscience to Edwin Rubel

Presentation of Awards

Pioneers of Underwater Acoustics Medal to Finn Jensen

Silver Medal in Animal Bioacoustics to Peter Narins

Silver Medal in Biomedical Acoustics to William D. O'Brien

Silver Medal in Psychological and Physiological Acoustics to Ruth Y. Litovsky

Silver Medal in Signal Processing in Acoustics to William S. Hodgkiss

Silver Medal in Speech Communication to Joanne L. Miller

WEDNESDAY AFTERNOON, 1 DECEMBER 2021 WILLAPA, 7:00 P.M. TO 8:00 P.M.

Session 3eID

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved! – Sounds to Astound

L. Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11747

Daniel A. Russell, Cochair
The Pennsylvania State University, 201-D Applied Science Building, University Park, PA 16802

The BYU student chapter of the Acoustical Society of America will put on a demonstration show for families with kids (age 12 -17). This workshop will cover connections between arts and physics. Properties of sound waves, including resonance, reflections, and frequency content, will be discussed in the context of musical instruments, singing, and concert halls. The primary goal of this acoustics demo show is to expose kids to opportunities in science and engineering and to interact with professionals in many areas of acoustics. Meeting attendees with kids who would like to attend must RSVP by sending an email to Keeta Jones at kjones@acousticalsociety.org.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:45 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 are as follows:

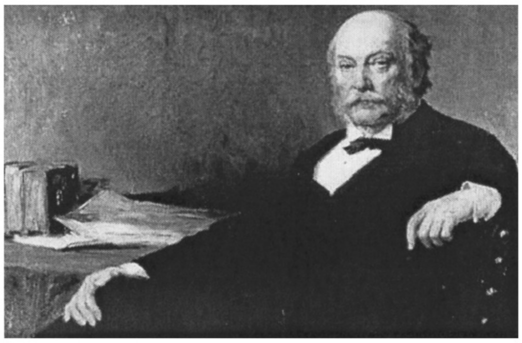
Engineering Acoustics (4:45 p.m.)	402
Acoustical Oceanography	Quinault
Animal Bioacoustics	302/303
Architectural Acoustics	Elwha A
Physical Acoustics	402/403
Psychological and Physiological Acoustics	Elwha B
Signal Processing in Acoustics	502
Structural Acoustics and Vibration	301/304

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Elwha A

Committees meeting on Thursday are as follows:

Computational Acoustics	306/307
Musical Acoustics	302
Noise	Elwha A
Speech Communication	Quinault
Underwater Acoustics	Elwha B



SCIENTIFIC PAPERS
Volumes I–VI
1869–1919
Lord Rayleigh
(John William Strutt)

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Animal Bioacoustics

Peter M. Narins

2021

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

Whitlow W. L. Au	1998
James A. Simmons	2005
Richard R. Fay	2012

CITATION FOR PETER M. NARINS

...for contributions to the sound production, hearing and neuroethology of anuran amphibians.

SEATTLE, WASHINGTON • 1 DECEMBER 2021

If you have spent time exploring the forests of Puerto Rico – or those of China, Costa Rica, Thailand, Madagascar, Cameroon, Nicaragua, French Guiana, Guyana, Brazil, Chile, or perhaps the lowlands of southern Brunei or the swamps of Indonesia or Malaysia – there is some chance you encountered this year's Silver Medal winner, Peter Martin Narins, with his microphone trained on the snout of a calling frog. Through his many adventures around the globe to explore hearing and sound communication, Peter has brought the study of animal bioacoustics to the world, and the world to the study of animal bioacoustics. Recognized for his groundbreaking studies on the unique auditory system of anuran amphibians, he has also contributed new knowledge on the auditory and communication systems of starlings, rufous-faced warblers, golden moles, Cape mole rats, and cats. Throughout his work, Peter has effortlessly integrated an evolutionary biologist's drive to discover why with an engineer's determination to understand how.

After receiving his Bachelor's (1965) and Master's (1966) degrees in electrical engineering from Cornell University, Peter taught for several years in Chile, as a member of the Peace Corps. Here, he met his wife of 52 years, Olivia Gubler. Returning to Cornell, Peter earned a Ph.D. (1976) in Neurobiology and Behavior, working with R. R. Capranica, a pioneer of comparative auditory neurobiology. After a two-year postdoctoral fellowship with E.F. Evans at the University of Keele (UK), he joined the faculty at UCLA, progressing rapidly through the ranks to earn a Full Professorship (1987). Peter is now Distinguished Research Professor of Integrative Biology & Physiology at UCLA. Over the course of his remarkable academic career, Peter has formed productive collaborations with scientists all over the world.

A fundamental thread running throughout Peter's research career is the investigation of basic auditory mechanisms in a comparative, evolutionary context. He developed a framework that integrates behavioral studies in the natural environment with laboratory neurophysiological experiments to ask how hearing works. His list of new discoveries is long. He was the first to document sex differences in the operation of the vertebrate ear. He discovered that frogs detect sounds not only via their eardrums but also via their lungs. This astonishing result paved new paths in our understanding of the evolution and mechanics of internally coupled ears. He found that seismic cues provide some frog species with important cues in acoustically noisy environments. In field studies of the functionally blind Namib golden mole, he showed that this animal relies solely on seismic cues to locate prey buried in sand.

Many of Peter's discoveries address how frogs deal with the considerable noise encountered when loudly calling males aggregate in dense choruses to attract females. His work is also at the forefront of understanding how anthropogenic noise and climate change impact frog communication. In field studies, he showed that male Puerto Rican Coqui frogs precisely time their own advertisement calls to avoid overlap with (and masking from) rival calling males. Using laser vibrometry, Peter demonstrated that these can frogs vocalize at intense levels (90-110 dB SPL at 1 m) for extended periods of time without damaging their own hearing because self-generated sounds arrive roughly in phase at both the inner and outer surfaces of the eardrum, thereby damping its movements. He was a leader of an international team that made the surprising discovery that stream-breeding frogs in China and Borneo produce and hear advertisement calls with ultrasonic components extending in frequency well above the background noise of fast-flowing water. Not only did the team record these species' acoustic signals and behavioral responses in the natural environment, but also, they documented the anatomical adaptations that make ultrasonic hearing possible. Using a 23-year database of field recordings made in the Puerto Rican rainforest, Peter documented changes in the advertisement calls of Coqui frogs as ambient temperature increased over this time period. Results from one of his field trips to Thailand helped launch investigations into the negative impacts of anthropogenic noise on frogs and other acoustically communicating animals. The practical significance of this line of research cannot be overstated.

Peter's contributions to training students in animal bioacoustics are unsurpassed. Even with 24 graduate students and 23 postdocs of his own, he served on the graduate committees of more than 100 additional students, including students from England, South Africa, Colombia, and Uruguay, while also teaching in several international summer schools devoted to bioacoustics and neuroethology. As an informal mentor to the two of us throughout our own careers, he has challenged us to think harder and deeper about science.

Peter's record of service to the discipline is exemplary. Within ASA, he served 3-year terms on the Animal Bioacoustics Technical Committee beginning in 1995, 2009, 2012, 2015, and 2018, and on the Psychological and Physiological Acoustics Technical Committee in 1991. He served as an instructor at the ASA School (2014), and he was an opening plenary speaker at the 5th Joint Meeting of the Acoustical Societies of America and Japan (2016). Peter was elected Fellow of the ASA in 1993. He has also advanced animal bioacoustics research and education through his service in the International Society for Neuroethology, where he served as Councilor (1995-1998), Treasurer (2004-2007), and President (2014-2016), and which elected him Fellow in 2016.

Beyond his contributions to animal bioacoustics, Peter is a long-time contributor to another form of communication – ham radio, his hobby since age 13. Peter remains an active participant in the ham radio world, with a contact record of 332 out of 340 active ham entities. He is also known for being a consummate storyteller, a fellow attendee at every scientific meeting you have ever been to, a skilled guitarist, and a devoted family man. Peter and Olivia are blessed with two children, Tom and Astrid, and three grandchildren, Nicole, Ryan, and Siena.

Peter Narins is an invaluable member of the animal bioacoustics community and the ASA. We are honored to contribute this encomium to his highly deserved receipt of this year's Silver Medal in Animal Bioacoustics.

MARK A. BEE
ANDREA MEGELA SIMMONS

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Biomedical Acoustics

William D. O'Brien, Jr.

2021

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

Silver Medal in Bioresponse to Vibration

Floyd Dunn 1989

Silver Medal in Biomedical Ultrasound/Bioresponse to Vibration

Ronald T. Verrillo 1999
James G. Miller 2004

Silver Medal in Biomedical Acoustics

Kullervo H. Hynynen 2013

CITATION FOR WILLIAM D. O'BRIEN, JR.

. . . for contributions to ultrasound bioeffects, dosimetry, and quantitative tissue characterization.

SEATTLE, WASHINGTON • 1 DECEMBER 2021

William D. O'Brien, Jr. has distinguished himself in the field of biomedical acoustics through an over 50-year career devoted to education, research, organizational leadership, and the mentoring of generations of trainees in biomedical ultrasound. Born in Chicago, Bill received B.S. (1966), M.S. (1968), and Ph.D. (1970) degrees from the University of Illinois at Urbana-Champaign (UIUC), conducting his doctoral research under ASA Gold and Silver Medalist Floyd Dunn. From 1971 to 1975 he worked as a Research Scientist at the U.S. Food and Drug Administration's Bureau of Radiological Health (now the Center for Devices and Radiological Health). He then joined the faculty at UIUC, where he now is the Donald Biggar Willett Professor Emeritus in Electrical and Computer Engineering. Since 1995, Bill has been the Director of the UIUC Bioacoustics Research Laboratory, a research center recognized internationally for its preeminence in biomedical acoustics.

Because medical applications of ultrasound range from imaging early first trimester embryos to tissue destruction via high-intensity focused ultrasound, determining safe levels for imaging applications always has been a concern of research institutions, regulatory bodies, user organizations, and industry groups. Bill has been at the forefront of research in the safety and dosimetry of diagnostic ultrasound, making its use in obstetrics safer for both mother and baby. His research has encompassed propagation of ultrasound in the uterus, including modeling tissue layers between the ultrasound source and fetus (in collaboration with the University of Cincinnati's Department of Obstetrics and Gynecology), and estimation of temperature rise and thermal dose from diagnostic ultrasound. Bill led a joint industry-government-professional society task group that assessed thermal risk from ultrasound exposure resulting in an international consensus standard (known informally as the "Output Display Standard") for displaying the Thermal Index and the Mechanical Index. These bioeffects indices are incorporated on virtually all clinical ultrasound imaging devices sold worldwide and provide the basis for both safety guidance and governmental regulation.

Bill also conducted pioneering research in applications of quantitative ultrasound (QUS) imaging in biology, agriculture, and medicine. He made early contributions by investigating basic interactions between acoustic waves and tissue. His work led to better understanding of the relationship between measurable acoustic properties of tissue, such as the speed of sound and attenuation coefficient, and the underlying tissue constituent materials, such as collagen, water, and fat content. His QUS studies have resulted in significant new diagnostic information for characterizing normal and abnormal tissue. His collaboration with the Department of Women, Children and Family Health Science at the University of Illinois Chicago led to the first use of QUS in pregnancy to detect cervical tissue changes for risk assessment of premature labor, allowing early intervention and improvement in pregnancy outcomes. In another application, he developed QUS imaging systems for objective assessment of quality and quantity in beef grading.

Further, Bill's work in assessing pulse-echo quantitative parameters (attenuation coefficients, backscatter coefficients, and their frequency dependence) for both diagnosis and therapy monitoring is cited often in a recent, national effort to establish "Pulse-Echo Quantitative Ultrasound" (PEQUS) biomarkers. The PEQUS work is being undertaken by a large group of physicians, scientists, engineers, and other medical professionals as part of the Quantitative Imaging Biomarkers Alliance (QIBA) led by the Radiological Society of North America. Bill's UIUC group has worked with researchers at the University of San Diego to develop quantitative ultrasound criteria to stage nonalcoholic fatty liver disease (NAFLD), the most common cause of chronic liver disease in the U.S. Developing these criteria is the current focus of the PEQUS-QIBA group. As a result of this long history of research productivity, Bill has authored over 400 publications, including over 280 original peer-reviewed articles, 58 of which were published in the *Journal of the Acoustical Society of America*.

Bill's intellectual vision, organizational skills, and drive over decades have created a legacy of influence in our field. An outstanding example is his years leading the Ultrasonics in Biophysics and Bioengineering Conference at UIUC's Allerton House retreat. This annual conference provided a unique forum for the exploration of current topics in an informal and intimate setting. Attendees that Bill assembled ranged from international experts to graduate students. The impact on the latter is especially noteworthy, as legends in the field would sit with the youngest of graduate students and talk as peers. This endeavor represents just one of the numerous ways Bill gave of his time, talent, and energy to help yield generations of life-long devotees to biomedical acoustics.

Bill's exemplary contributions to professional organizations are too many to mention here, but to name just a few, he has served as President of both the American Institute of Ultrasound in Medicine (AIUM) (one of only four non-physicians in the AIUM's 70-year history) and the Institute of Electrical and Electronics Engineers (IEEE) Ultrasonics, Ferroelectrics, and Frequency Control (UFFC) Society. For 16 years he was Editor-in-Chief of the Society's *Transactions on Ultrasonics, Ferroelectrics, and Frequency Control*. He has been an Advisor to the Society of Diagnostic Medical Sonography, a society founded to promote, advance, and educate its members and the medical community in the science of medical diagnostic ultrasound.

In recognition of his considerable contributions, Bill has achieved Fellow status in the ASA, AIUM, and IEEE. Further, he is a Founding Fellow of the American Institute of Medical and Biological Engineering. From the AIUM he has received the Presidential Recognition Award (twice), the Joseph H. Holmes Basic Science Pioneer Award, and the William J. Fry Memorial Lecture Award. From the IEEE, he was recipient of the Centennial Medal, Millennium Medal, and the Outstanding IEEE Student Branch Counselor Award for IEEE Region 4, which comprises 10 states in the American Midwest. Within the IEEE UFFC Society, he was the 8th recipient of the Rayleigh Award, which represents the Society's highest honor.

Bill's exceptional commitment and influence in the field of biomedical acoustics make him particularly deserving of the ASA Silver Medal, and we congratulate him on this well-deserved recognition.

GERALD R. HARRIS
J. BRIAN FOWLKES
JAMES A. ZAGZEBSKI

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Psychological and Physiological Acoustics

Ruth Y. Litovsky

2021

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PREVIOUS RECIPIENTS

Lloyd A. Jeffress	1977	Neal F. Viemeister	2001
Ernest Glen Wever	1981	Brian C. J. Moore	2002
Eberhard Zwicker	1987	H. Steven Colburn	2004
David M. Green	1990	William A. Yost	2006
Nathaniel I. Durlach	1994	Roy D. Patterson	2015

von Békésy Medal

The von Békésy Medal is presented to an individual, irrespective of nationality, age, or society affi liation, who has made an outstanding contribution to the science of psychological and physiological acoustics, as evidenced by publication of research results in professional journals or by other accomplishments in the field.

PREVIOUS RECIPIENTS

Jozef J. Zwislocki	1985
Peter Dallos	1995
Murray B. Sachs	1998
William S. Rhode	2010
M. Charles Liberman	2012

CITATION FOR RUTH Y. LITOVSKY

… for contributions to understanding binaural hearing and the perceptual consequences of providing bilateral cochlear implants

SEATTLE, WASHINGTON • 1 DECEMBER 2021

If you asked random members of the hearing science community to name the people who elevate the field through their both their science and their leadership, Ruth Y. Litovsky would undoubtedly be near the top of the list. And that is the key to Ruth. She has made important contributions scientifically, without question; but she also has worked tirelessly throughout her career to create an equitable, welcoming, and well-functioning auditory science community. She sets high standards for herself and her trainees, while also helping every individual to achieve their best work. She contributes both her extraordinary organizational gifts and her warm personality so that the Acoustical Society of America (ASA) – and sister scientific organizations – run smoothly, and better.

Scientifically, Ruth's work has revealed fundamental insights into the perceptual capabilities of human listeners. She has studied sound localization, source separation, intelligibility, development, and the precedence effect. More recently, her work has expanded to explore the impact of experience and critical periods on perception. Not only important in basic science, her work is also of great clinical significance, especially in the area of cochlear implants. She has a long record of influential publications, including the definitive review paper on "the precedence effect," published in the *Journal of the Acoustical Society of America* (JASA) in 1999 and a plethora of studies on spatial hearing in both normal-hearing listeners and listeners with cochlear implants. Over the course of the last three decades, she has consistently identified critical gaps in our knowledge and then developed creative experiments addressing those gaps.

Ruth's academic trajectory followed a focused path. As an undergraduate at Washington University in St. Louis, she earned a bachelor's degree in Psychology and a Master's in Neuropsychology. She moved to Amherst, Massachusetts for her Ph.D. in Developmental Psychology under Rachel Keen (formerly Clifton), where the lab demonstrated that the precedence effect exhibits interesting dynamics, an observation that has guided thinking in the field ever since. Importantly, it was this work that brought Ruth to the ASA community: it was the topic of both her first ASA presentation in the spring of 1989 and her first JASA publication, three years later. Driven by a desire to learn more about the mechanisms underlying psychoacoustic phenomenon, she then did postdoctoral work at the University of Wisconsin with Professor Tom Yin. She moved to Boston in 1994 to continue her research in spatial hearing as a Research Associate at Boston University, the Massachusetts Institute of Technology, and the Eaton-Peabody Lab of the Massachusetts Eye and Ear Infirmary. She moved to the University of Wisconsin - Madison in 2001 to join the faculty of the Department of Communication Sciences and Disorders, where she established her own productive, independent laboratory and rose through the professorial ranks to her current position of Department Chair.

Roughly at the time that she moved to Wisconsin, she turned her attention to cochlear implants. This choice has had a profound impact not only on Ruth's career, but on the research community. The shift brought Ruth's deep knowledge of spatial hearing to the field at a time when bilateral implants were unusual and underappreciated. She quickly carved out a niche, first demonstrating that bilateral implants provided perceptual benefits, and later probing the mechanisms supporting these benefits. At the same time, she built a parallel line of research on the development of spatial hearing in typical children. Ultimately, these two lines of work came together in studying spatial hearing in children with bilateral cochlear implants, and later, developmental aspects of spatial hearing in both children and adults using cochlear implants.

Ruth has achieved wide recognition for her scientific achievements and her sustained record of research excellence. She is a highly sought-after speaker with over 100 keynote and invited talks to date. She was elected a Fellow of the ASA in 2009 and received a Fulbright Scholar award in 2014 to support a sabbatical in Australia.

Many scientists with scientific contributions as fundamental and important as Ruth's achieved this fame by focusing strictly on their own careers. However, the energy and passion that Ruth puts into her science is matched by her commitment to diversity, equity, and inclusion (DEI), as well as mentorship. She has long been a strong advocate of DEI efforts, even before the recent increase in awareness of these issues and has helped ASA (and other parts of the hearing community) adopt best practices to address systemic problems. She has established numerous mentoring programs within every organization of which she is a part. In her own laboratory, she has trained many graduate students and postdocs, and has excelled in helping them excel; many are now independent scientists, adding to her legacy.

One of the key factors informing Ruth's approach to mentoring is that she has, from the start, balanced the demands of family and work with grace. In fact, it is a good bet that anyone who met Ruth in the mid-1990s can conjure an image of her at some reception talking with senior leaders in the field about spatial hearing, all while toddler Leora clutched her leg and baby Micah balanced at her hip (Gaby, the youngest, came a little later). As she would be the first to tell you, she could never have done it without the support and equal efforts of her husband David Baum. But she also has been open about not just the joys, but also the challenges of juggling being a mother and a scientist and a wife. By example, she has served as a role model to those around her. Moreover, she has worked to normalize being dedicated to both family and to science and has been instrumental in creating policies and procedures that help young parents be successful, such as providing childcare benefits to conference attendees.

Ruth has also been selfless in her more general service to the profession, taking on leadership both scientifically and administratively in the ASA as well as other organizations. Within the Acoustical Society, she has been elected to the P&P Technical Committee an astonishing four times, starting in 2001 (serving 12 out of 19 years), served as an Associate Editor for JASA for 13 years (2006-2019), and organized numerous special sessions. The field of hearing science, and ASA more generally, are better for Ruth being such an engaged, energetic, and positive presence.

BARBARA G. SHINN-CUNNINGHAM
H. STEVEN COLBURN

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Signal Processing in Acoustics

William S. Hodgkiss

2021

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

Edmund J. Sullivan	2010
Theodore G. Birdsall	2011
Brian G. Ferguson	2015

CITATION FOR WILLIAM S. HODGKISS

… for contributions to at-sea experimentation and ocean acoustics signal processing

SEATTLE, WASHINGTON • 1 DECEMBER 2021

The dates between Bill Hodgkiss' BSEE from Bucknell to his Ph.D. in EE from Duke, 1/72-8/75 is early evidence that his understanding of time dilation and Doppler in signal processing spills over to his everyday activities. He then spent two years as an engineer at the Naval Ocean Systems Center in San Diego and another year as an Assistant Professor at Bucknell. His subsequent career trajectory after joining the University of California, San Diego (UCSD) Graduate Department of the Scripps Institution of Oceanography (SIO) and its Marine Physical Laboratory (MPL) was Assistant to Distinguished Full Professor. Over the years as primarily a professor and seagoing researcher, Bill almost always occupied other senior university positions, simultaneously; an incomplete list is SIO Deputy Director of Scientific Affairs, Chair of the UCSD Academic Senate, and UCSD Associate Vice Chancellor for Academic Planning and Resources. Most of the time during his senior professorship he was also Deputy Director of MPL. In his spare time, he also earned an MBA from UCLA by commuting 1 day a week during his early days when he was being considered for tenure. Among all this activity and responsibilities, he has continuously confounded his colleagues with his ability to appear, almost instantaneously, at sequential meetings on diverse matters at locations separated by typically large UCSD campus distances. Additionally, his students, colleagues and staff rave about his infinite patience and his generosity of time. Fortunately, Bill's talent for space-time manipulation also allows him to spend much time with his wife Paula and their two daughters who live nearby with their families.

His initial collaborations provided a solid foundation for his subsequent research. A list of mentors he provided me includes: his thesis advisor Loren Nolte (a student of Ted Birdsall), followed by Fred Spiess, Vic Anderson and Fred Fisher, an exceptional seagoing triumvirate of researchers influencing his early days at MPL. The evolving central theme of Bill's research was to understand the interaction between the typically inadequately known dynamic ocean medium and signals embedded in ambient noise for the purpose of extracting and using these signals. Much of his effort over his research career, together with his graduate students, involved carrying out a massive measurement program. Combining his deep knowledge of signal processing theory and data analysis with more than 50 research cruises (of greater than 2 weeks) collecting acoustic and ocean data (while fending off his well-known vulnerability to motion sickness), Bill is clearly one of the world's premier sea-going, environmental ocean acoustics signal processing researchers.

Early work included the development of both software and hardware implemented adaptive array processing of acoustic signals, noise, and reverberation. This included work on the localization of autonomous sensors and the development of autonomous vector sensors deployed in the ocean (Swallow Floats). His next venture was the beginning of intense matched field processing (MFP) research starting with the study of ambient ocean noise. His 1992 paper on fluctuations of signals at 1000 km range involved one of the earliest large vertical arrays deployed in deep water. His subsequent series of MFP papers, mostly in shallow water were based around a series of "SWellEx" experiments that also provided data for a generation of ocean acoustic signal processors to test their new methods. Here, he pioneered and experimentally validated MFP work including narrowband, broadband and bispectral processing. His signal processing contributions also included some of the earlier, experimentally verified, shallow and deep-water inversion methods using MFP. Bill, in subsequent years made it a practice to always upgrade the storage media that held the data he collected allowing him to distribute it as requested by many researchers—an extremely valuable service to our research community.

In the next decade or so, Bill pioneered research in ocean phase conjugation and time reversal mirror acoustics in shallow water. With his engineering staff that he mentored over the years, he designed and successfully deployed the first ocean acoustic time reversal mirror (TRM)—a twenty-nine element split-cylinder source-receiver transducer array centered around 450 Hz that was successfully deployed multiple times in the Mediterranean. Subsequently, he built and deployed 3.5 and 20 kHz TRM's. This research not only demonstrated TR in the ocean, but it showed that there were important signal processing concepts and applications to ocean acoustics centered about its intrinsic coherence that could be applied to reverberation nulling, inversion, and most importantly, underwater communications (ACOMMS). This research included results on the stability of the TRM focus as a function of the ocean medium and a method to shift the location of the focus using waveguide invariant theory. The latter was the signal-processing analog of mechanically and adaptively varying the focus of a telescope. All this work appeared in a series of important TR papers and TR based acoustic communication papers that combined equalization, vertical and horizontal arrays, and synthetic aperture and even demonstrated the feasibility of ultra-long distance, deep-water ACOMMS out to a range of 3,250 km. Continuing his research in MFP related topics, Bill added statistical estimation and nonlinear optimization to his research in matched field inversions, and together with his at-sea experiments significantly advanced the field of signal processing in a complex and poorly sampled ocean medium.

As his university administrative responsibilities increased in the last ten years, his research contributions accelerated further, emphasizing very complex, signal processing problems in which there is great uncertainty in all the input parameters. Advanced statistical methods, random matrix theory and moving array signal processing have been successful subjects of his research. He successfully utilized adaptive processing to the noise/passive fathometer, and he designed and conducted some of the earliest shallow water acoustic noise experiments that demonstrated that the acoustic transfer function between two points in the sea can be measured passively and that information about the ocean bottom can be extracted. His most recent work includes the application of compressive sensing to array processing and an experimental demonstration of moving source tomography.

With his many years of combining state-of-the-art signal processing with seagoing experiments including directing multi-institutional expeditions, classroom teaching in all areas of signal processing and mentoring 26 Ph.D. students, Bill Hodgkiss clearly has earned the Silver Medal in Signal Processing in Acoustics.

WILLIAM A. KUPERMAN

ACOUSTICAL SOCIETY OF AMERICA

Silver Medal in

Speech Communication

Joanne L. Miller

2021

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Franklin S. Cooper	1975	Katherine S. Harris	2005
Gunnar Fant	1980	Ingo R. Titze	2007
Kenneth N. Stevens	1983	Winifred Strange	2008
Dennis H. Klatt	1987	David B. Pisoni	2010
Arthur S. House	1991	Sheila E. Blumstein	2014
Peter Ladefoged	1994	John J. Ohala	2015
Patricia K. Kuhl	1997	Anne Cutler	2020

CITATION FOR JOANNE L. MILLER

… for contributions to the understanding of phonetic and lexical processing in speech perception

SEATTLE, WASHINGTON • 1 DECEMBER 2021

Joanne L. Miller was born and raised in Minneapolis, Minnesota, and completed her B.S., M.A., and Ph.D. degrees in the Department of Communication Sciences and Disorders at the University of Minnesota. As a graduate student, Joanne published studies on fluency disorders and central auditory processing, but her interests soon focused on the perception of speech. In 1974, Joanne accepted an NIH postdoctoral fellowship at Brown University with Peter D. Eimas, a psychologist whose breakthrough studies focused on the perception of speech by infants. The research collaboration between Miller and Eimas on adult and infant speech perception flourished. They were married in 1977. By that time, Joanne was Assistant Professor of Psychology at Northeastern University where she quickly rose through the ranks to become Associate Professor, then Professor, and in 1997 was named Matthews Distinguished University Professor of Psychology. Joanne became Chair of the Psychology department in 2010, and she remained in that position for a decade until her retirement in 2020. Joanne is now Matthews Distinguished University Professor Emeritus.

Joanne Miller's studies on speech perception and lexical access are known for their precision, both in terms of theory and methods. She foregrounded a basic scientific problem in speech perception, specifically, the relation between perception of the phoneme and its highly varied surface expression. The relation between linguistic elements and their physical manifestations (whether these are articulatory, acoustic, or auditory) defies simple accounts. Combining acoustic analysis, speech synthesis, and perceptual studies, Joanne's elegant experiments show that perceivers adjust the standards governing perception of phonetic categories in relation to the contexts in which they occur. In critical experiments conducted with a host of collaborators, many of which were published in the ASA's *Journal of the Acoustical Society of America*, Joanne brilliantly demonstrated key features of the context effects in phonetic perception.

Phonetic features distinguish the consonants and vowels that make up words, and characterize aspects such as 'manner of articulation,' which distinguishes the phoneme /b/ (as in bag) from the phoneme /w/ (as in wag). An early and especially influential study in Joanne's large portfolio of published work, conducted with Alvin Liberman of Haskins Laboratories, is emblematic. A major acoustic property distinguishing /b/ from /w/ is the duration of the initial formant transitions, with shorter transitions specifying /b/ and longer transitions specifying /w/. Miller and Liberman demonstrated that the mapping between transition duration and phonetic category is not absolute but systematically varies as a function of later-occurring information in the syllable. They interpreted this effect as an adjustment for speaking rate. A crucial test examined perception under two conditions: when consonant-vowel syllables were lengthened by increasing the duration of the vowel, reflecting a slower rate of speech, versus when they were lengthened to the same extent by adding a final consonant, reflecting a faster rate of speech. Even though overall syllable duration was the same in the two cases, the /b/-/w/ boundary shifted with the change in speaking rate, providing evidence that speaking rate, and not simple overall syllable duration, was the critical factor driving the perceptual effect. In a subsequent study with Peter Eimas published in *Science*, Joanne found evidence that young infants also process transition duration in relation to later-occurring information in the syllable. This finding provides a first step in mapping out infants' perception of speech under complex contextual manipulations. Joanne went on to investigate numerous aspects of rate-dependent perception, as well as other contextual factors that influence phonetic and lexical processing. She also turned her attention to the fine-grained internal structure of phonetic categories, characterized by some exemplars along a phonetically relevant acoustic continuum being perceived as better category exemplars than others. Joanne showed that internal category structure is itself context-dependent, systematically influenced by such factors as speaking rate, phonetic context, and dialect, and that internal category structure reflects talker-specific phonetic variation.

The ability to win federal funding for one's ideas is a measure of success. Joanne Miller was Principal Investigator of a National Institutes of Health (NIH) grant on speech perception that was continuously funded for more than 30 years, from 1978 to 2009, with an accompanying NIH Research Career Development Award from 1981-1986. Joanne's history of funding from NIH is impressive and serves as an indicator of the esteem her peers had for her research questions and the systematic way in which she addressed them.

Beyond her scientific contributions, Joanne Miller has contributed greatly to the Acoustical Society of America (ASA) and was elected Fellow in 1992. She served as Associate Editor for Speech for the *Journal of the Acoustical Society of America* (1985-1988), as a member of the Speech Communication Technical Committee (1984-1990), the Books Committee (1988-1994), the Committee on Public Relations (1993-2002), and as ASA representative to the Linguistics and Language Science Section of the American Association for the Advancement of Science (1998-2010). Importantly, Joanne was the Editor-in-Chief of a highly respected three-volume series on *Speech Communication* published by the Acoustical Society of America in 1991. Working with Bishnu Atal and Ray Kent, Joanne co-edited one volume on *Speech Perception*, one volume on *Speech Production*, and one volume on *Speech Processing*. Each volume offered a selection of the classic papers in these three areas and the three-volume set was very well received.

Joanne Miller's contributions to the next generation of scholars of human communication are also noteworthy. She trained and mentored many excellent students and junior colleagues and exposed them to her rigorous methods and precise theoretical reasoning. Her students have gone on to successful careers in a variety of settings.

Awarding the Silver Medal in Speech Communication to Joanne L. Miller recognizes her highly significant contributions to the scientific study of speech perception as well as her many important contributions to the Acoustical Society of America.

PATRICIA KUHL
ROBERT REMEZ
DAVID PISONI

ACOUSTICAL SOCIETY OF AMERICA
PIONEERS OF UNDERWATER ACOUSTICS MEDAL

Finn Jensen

2021

The Pioneers of Underwater Acoustics Medal is presented to an individual irrespective of nationality, age, or society affiliation, who has made an outstanding contribution to the science of underwater acoustics, as evidenced by publication of research in professional journals or by other accomplishments in the field. The award was named in honor of five pioneers in the field: H. J. W. Fay, R. A. Fessenden. H. C. Hayes, G. W. Pierce, and P. Langevin.

PREVIOUS RECIPIENTS

Harvey C. Hayes	1959	Ivan Tolstoy	1990
Albert B. Wood	1961	Homer P. Bucker	1993
J. Warren Horton	1963	William A. Kuperman	1995
Frederick V. Hunt	1965	Darrell R. Jackson	2000
Harold L. Saxton	1970	Frederick D. Tappert	2002
Carl Eckart	1973	Henrik Schmidt	2005
Claude W. Horton, Sr.	1980	William M. Carey	2007
Arthur O. Williams	1982	George V. Frisk	2010
Fred N. Spiess	1985	Michael B. Porter	2014
Robert J. Urick	1988	Michael J. Buckingham	2017

ENCOMIUM FOR FINN B. JENSEN

… for contributions to ocean acoustic modeling and model data validation

SEATTLE, WASHINGTON • 1 DECEMBER 2021

Finn Jensen was born in Aalborg Denmark and studied Engineering at the Technical University of Denmark (TUD) where he gravitated toward more basic physics, selecting fluid mechanics for his M.S. specialization courses. There he met a new faculty member, Leif Bjorno, who became his mentor and close friend. Leif encouraged and secured funding for Finn to do a Ph.D. under his guidance. From there, both went forward to lead extremely productive careers as research scientists.

In 1973, Finn accepted a 3-5 year appointment at the NATO SACLANT ASW Research Centre (now CMRE). Finn, always preferring brevity, saw no need for the dash thereby simplifying the length of his tenure to 35 years. The high impact of his research, well known in the broad, international underwater acoustics (UA) community has Centre specific roots. Throughout his career, mostly in his position as the head of the Environmental Modeling Group, he strived to make the Centre an important focus of international research by attracting and mentoring scientists that would also go on to be key researchers in the UA community. The extent of the Centre's subsequent broad involvement in UA under his leadership is clearly revealed by Finn being the first author of *Computational Ocean Acoustics* (1993, revised 2011), the standard UA text for more than a quarter of a century. The other authors, who worked with and/or for him up to about 1992, did not overlap with each other during their individual 4-5 years tenures at the Centre. A partial list of scientists he mentored includes two of us (Henrik Schmidt, Michael Porter), as well as Peter Gerstoft and Mario Zampolli. Another important example of Finn making the SACLANTCEN a center of the UA community is that by 1988, his group had distributed extremely useable versions of normal mode, parabolic (PE) and spectral codes to 46 institutions in 10 countries. Below, we present a selection of his contributions that demonstrate his pioneering activities.

Among the first very important contributions he made was the analysis of transmission loss data in shallow water as a function of the environment and frequency—he not only elucidated the required measurement process to determine the "optimum frequency" of propagation, but he quantitatively identified the major environmental features which determined the optimum frequency. His seagoing experimental work has been extensive, characterizing diverse areas such as the Baltic Sea, Strait of Sicily, GI-UK Gap and Iceland-Faroe Front areas. At the earliest (1976), he noticed one-third octave frequency dependence of near-surface transmission loss indicated acoustic properties of the ocean bottom in shallow water data. This research evolved over a period of years culminating in a seminal 1983 *Journal of the Acoustical Society of America* (JASA) "Optimal Frequency" paper. This work was not only of importance to the UA crowd, but also to marine mammal acousticians relating the environment to the properties of marine mammal calls as well as more recently, the environmental community concerned with "soundscapes."

At about the same time, he was also applying the PE method together with normal mode theory to study range-dependent environments—explaining PE results in terms of modal coupling to the continuous spectrum. Many acousticians are familiar with his iconic result of upslope propagation radiating into the bottom mode by mode. Regarding the PE, after its initial development by Tappert, Jensen's research in that area included a careful error analysis of the numerics which then led him to study the consequences of energy-conservation in the numerical procedure for changing environments. This led to rigorous methods of segmenting two and three-dimensional environments for accurately propagating acoustic fields and to a series of benchmark examples that the modeling community used to verify new developments—over the next decades. Leo Felsen, an icon in analytic wave theory (as opposed to computer model development), said that Finn was keeping modelers "honest'. Involvement with benchmarks is a recurring theme over the years. He has played leadership roles in developing procedures to validate not only acoustic propagation models, but also structural acoustic models for predicting scattering from complex elastic objects as discussed below.

Because a large component of his research has been concerned with measurements in bottom limited regions, he has made major contributions to modeling interactions with the seabed. His contributions in this area include full wave solutions of reflection off a multi-layered viscoelastic half-space followed by a study of beam penetration in ocean bottom sediments. His research on using ocean bottom surface-waves to determine sediment elastic properties was a precursor to recent research on the latter using ambient noise fields.

It was only natural for Jensen to extend his seismo-acoustic research to include scattering from objects lying on the bottom, combining his ocean acoustic research with elastic medium physics. Since finite elements are generally too costly to employ for water-column-propagation, he turned his group's attention to developing hybrid finite element/waterborne propagation methods dealing with some of the fundamental issues arising from the hybrid methods. This included boundary and radiation conditions in range-dependent settings. As explained by Mario Zampolli, a key researcher in this program, this research became a major line of Centre work because of Finn's leadership, eventually covering theory, numerics, experiment and the development of benchmark models to assist other researchers in the field.

All the work mentioned here was carried out in Italy where Finn lives with his wife Patrizia. In their lovely home at the foot of the Carrara Mountains, they especially enjoy entertaining their three young grandchildren who always bring along their parents. These include his two daughters, Liza who is a software engineer at the European Central Bank in Frankfurt, Germany, and Stella who is a medical doctor at the Neurology Dept. of the main hospital in Carrara. When not in Italy, Patrizia and Finn are often globetrotting around the world, having visited, to date, 85 countries on all 7 continents.

Throughout his research there was a continuity of modeling, experimental validation, quantitative error analysis, and the sharing of an extensive set of research products. His publication list indicates co-authorship with two categories—those he mentored and other world class contemporaries (often with the former migrating to the latter). Dr. Finn Jensen is clearly a Pioneer in Underwater Acoustics.

WILLIAM A. KUPERMAN
MICHAEL B. PORTER
HENRIK SCHMIDT

Session 4aAA

Architectural Acoustics: Architectural Acoustics Potpourri

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

Chair's Introduction—8:20

Contributed Papers

8:25

4aAA1. Perception and task performance under two simulated office background noise conditions. Zhi Zhou (Graduate Program in Acoust., The Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, zmz5181@psu.edu), Michelle C. Vigeant (Graduate Program in Acoust., The Pennsylvania State Univ., University Park, PA), and Andrew Dittberner (GN Hearing, Bloomington, MN)

The acoustics of an open-office environment play an important role in people's working performance. This study was conducted to investigate how task performance is affected by two simulated office noise environments: (1) recorded HVAC noise and (2) a series of intermittent noises, e.g., conversations, phone ringing, etc. These conditions were reproduced over a 32-loudspeaker array in an anechoic chamber to include realistic spatial placement of sound sources. Each subject was seated at the center of the array and asked to perform five cognitive tasks to evaluate effects on memory, reasoning, planning, and attention. The tasks were digit span, grammatical reasoning, tower of London, attention network test, and mental arithmetic. After each noise condition, the participants rated several items including annoyance and perceived performance. Heart rate variability (HRV) was also recorded using ECG sensors as a physiological measure of arousal. Baseline measurements were first obtained in silence, followed by a task training set in silence, and an actual test set in silence. A statistical analysis was used to investigate possible correlations between the acoustic conditions and the cognitive task results, perceptual ratings, and HRV data, respectively. Study details and the findings will be presented.

8:40

4aAA2. Simulation of sound field in a reverberation chamber using the diffusion equation model. Ryan Hao (School of Architecture, Rensselaer Polytechnic Inst., 175 Troy Schenectady Rd., Apt. 2, Watervliet, NY 12189, ryanhao15@gmail.com), Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), and Juan Navarro (Polytechnic Sci., San Antonio's Catholic Univ., Murcia, Spain)

The acoustic diffusion equation model has been applied within a wide range of enclosure types to simulate diffuse sound fields. Recently, focus has been drawn to its applicability in reverberation chambers to simulate sound fields among a heterogeneous distribution of absorbing materials, particularly those that are highly absorptive. Where quasi-diffuse or non-diffuse sound fields are generated in the presence of highly absorptive material, the acoustic diffusion equation model has convenient capabilities to simulate sound energy flow. Furthermore, it has continued to show promise in characterizing the absorption coefficients of sound absorbing material. Challenges for reproducibility in reverberation room measurements have been well-documented and where traditional chamber-based measurement have shown to be inconsistent, this work will evaluate the acoustic diffusion equation model's suitability as an alternative and accurate method to characterize absorption coefficients.

8:55

4aAA3. Toward adaptive auralization of dynamic immersive virtual environments in room-centered virtual reality systems. Mincong Huang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, huangm5@rpi.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Existing auralization frameworks for interactive virtual environments have found applications in simulating acoustic conditions for binaural listening and real-time audiovisual navigation. This work, situated at the Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab), extends elements of these frameworks for human-scale interactive audiovisual display with movement of virtual sound sources and non-static virtual representation of the facility's footprint. The work involves the development of an adaptive acoustic ray-tracing prototype that is capable of generating impulse responses for individual virtual loudspeaker representation based upon changes of room orientation in virtual space at runtime. Through the integrated use of game engines (i.e., Unity and Unreal Engine), the prototype is presented in the context of dynamic audiovisual display, and actively analyzes the virtual scene geometries using a multi-detailed rendering approach. With both reconstructed high-resolution 3D models of existing spaces, and automatically generated virtual landscapes from geo-spatial data, the developed system is evaluated both in terms of computational efficiency, and in terms of conventional room acoustics parameters using model-based acoustic energy decay analysis across listening region. [Work supported by the Cognitive Immersive Systems Laboratory (CISL) and NSF IIS-1909229.]

9:10

4aAA4. The art of stereo reproduction—A test engineer's perspective. James Weir (Hottinger Bruel & Kjaer, 3079 Premire Pkwy, Ste. 120, Duluth, GA 30097, jim.weir@bksv.com)

Stereophonic reproduction is an art form often at odds with analytical testing. The author presents a breakdown of the aspects of stereo reproduction, including musical acoustics and the electroacoustics, to room acoustic assessment and the psychoacoustics used in the assessment and enjoyment of this common art form. This goal is to provide a holistic overview and encourage discussion and improvements in the processes used in the production of the art, as well as the analysis and dialog of the consumers of the art.

9:25

4aAA5. Threshold and the consultation process. Angel Castañon (Ventura College, 4667 Telegraph Rd., Ventura, CA 93003, angelc0428@gmail.com)

Here I had the opportunity to take part in the Acoustical Society of America's (ASA) Summer Undergraduate Research or Internship Experience in Acoustics (SURIEA) Program. The program was designed for underrepresented minority groups in order to introduce them to the field of

acoustics. After a two-week crash course I was able to experience what it was like working in an architectural acoustic consultation firm, and learn the extensive consultation process at Threshold. Not only that, but learning some of the programs, the methodology, skills, and purpose of Threshold as well. The impact of the program has certainly opened my eyes to the field of architectural acoustics, and how much work has to go into a project for the best possible outcome. In essence, I found the experience to be one of the most impact I had in my life.

9:40

4aAA6. The effects of acoustical ceiling panel type and penetrations for services on vertical sound isolation inside buildings. Gary Madaras (Acoust., Rockfon, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com)

Attenuation of sound transmitting between rooms oriented over one another inside buildings is studied. Transmission loss and sound transmission class were measured by an independent, accredited, acoustics laboratory with and without a variety of modular acoustic ceilings suspended under a baseline concrete floor structure. Ceiling panel material types include stone wool, fiberglass and mineral fiber. Ceilings were tested with and without the presence of service penetrations for supply air diffusers, return air grilles and light fixtures. Some ceilings were also scanned with a sound intensity probe and the resulting color sound maps are used as supplemental method of evaluating both isolation and absorption performance of the individual components of the ceiling systems. Results show that while the effects of ceiling panel type on absorption performance, and thus room acoustics, is substantial, the material type and weight of the ceiling panels do not substantially affect the overall isolation performance of the floor-ceiling assembly.

9:55

4aAA7. The Seattle Space Needle renovation: Acoustical design considerations and challenges at 600 ft above ground level. Daniel C. Bruck (BRC Acoust., 1932 1st Ave., Ste. 620, Seattle, WA 98101, danb@brcacoustics.com) and Jaimie Penzell (BRC Acoust., Seattle, WA)

Built for the 1962 Seattle World's Fair as a symbol of humanity's Space Age aspirations, the Seattle Space Needle is one of the most recognizable landmarks in North America. In September 2017, construction commenced for "The Century Project," a renovation project designed to reveal the tower's internal structure while expanding and improving its views. This presentation will discuss the acoustical design criteria for the two levels of all-glass, circular observation levels, as well as design approaches for reverberant sound control, sound isolation for key areas, and mechanical system noise control for a proposed fine dining restaurant at the "Loupe" level. The presentation will also include post-construction measurements and several photos during construction and at project completion.

10:10–10:25 Break

10:25

4aAA8. Elevator machine room noise. Zohreh Razavi (NDY, #608 1166 Alberni St., Vancouver, BC V6E 3Z3, Canada, z.razavi@ndy.com)

While elevators are sources of noise and vibrations in high- and low-rise buildings, this problem is overlooked and not enforced by many jurisdictions, including in Canada. Although the STC rating requirements of the demising walls between the elevator shafts and dwelling units are mandatory in Building codes, the noise limit of the elevator machine room is not addressed. The noise levels from elevators are not high, however they can be bothersome, especially for the residences at higher levels. In this paper, a summary of a noise review from an elevator machine room due to the low-rise occupants' complaints along with noise mitigation recommendations are reviewed. A sound control review requirement is mandated by most bylaws in Canada. However, it is at the architect's discretion to decide whether acoustical engineering services are required or not. Guidelines promoting occupant well-being and comfort should be included in the sound control requirements, and enforced in bylaws, and it is the duty of acoustic consultants to promote this.

10:40

4aAA9. Concrete and void: Understanding the acoustics of a parking garage. Nicholas DeMaison (Montclair State Univ., Montclair, NJ) and Zackery Belanger (Arcgeometer LC, 1111 Bellevue St., Ste. 360, Detroit, MI 48207, zb@arcgeometer.com)

The COVID-19 pandemic has converted musicians at all levels of professional accomplishment into improvising buskers, brocantuers of stray sound in *ad-hoc* market-places. Lacking the resources to design and build new open-air venues quickly, and facing the possibility of putting group music-making activity on hold, musicians have instead repurposed available spaces. In particular, the pandemic forced the first author's university orchestra from their dedicated performance space into a nearby parking garage. It performed unexpectedly well as a space for acoustic music, in spite of its singular brutalist material and large voids to the outside air. This study, undertaken by the conductor and an acoustician, draws insight from acoustic measurements and wave-based simulations in an attempt to unravel why a space designed for the limited function of supporting cars could give a respectable acoustic showing. The satisfying nature of the shared group experience may compel a widening of our perception of what constitutes an appropriate space for acoustic music making, and a reimagining of the historical line between outdoor and indoor music.

10:55

4aAA10. Acoustic effects of face masks on speech: Impulse response measurements between two binaural mannikins. Victoria R. Anderson (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE 68182-0816, vranderson@unomaha.edu), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

The COVID-19 pandemic has resulted in widespread use of facial masks that cover the nose and mouth areas to prevent viral transmission. While masks have helped provide protection from spreading illness, they have also had an impact on how people communicate, with many finding it more difficult to understand speech from those who are wearing masks. Facial masks affect the directivity and sound levels of the wearer's verbal communication, and if opaque, they also cover visual cues that aid understanding. Studies have documented the effect of different mask types on speech intelligibility, speech transmission index, talker directivity, and resulting speech features. This talk presents the first of measurements towards composing a corpus of impulse responses gathered between two head and torso simulators (HATS) in an anechoic chamber at varying directional positions and distances. One HATS with a mouth simulator serves as the talker and is outfitted with different mask types, including cloth, N95, and surgical paper masks, in combination with face shields. The resulting impulse response data base may be used by researchers to simulate masked verbal communication scenarios such as in virtual soundfield environments, allowing systematic testing of how masks impact speech perception in numerous situations.

11:10

4aAA11. A sound absorber made of copper to reduce virus spread in architectural applications. Jorge P. Arenas (Inst. of Acoust., Univ. Austral of Chile, Edif. 6000, Campus Miraflores, Valdivia 5090000, Chile, jparenas@uach.cl)

For many years, copper and copper alloys materials have been recognized as an efficient antimicrobial. Experimental studies have reported that solid copper surfaces can kill 99.9% of microorganisms within two hours of contact, mainly attributed to the release of ions. However, copper materials in acoustical applications have not been usual, mainly because of their costs. The COVID-19 outbreak has boosted the use of copper in the most frequently touched surfaces to help reduce the spread of the virus. In addition to being malleable, ductile, and an extremely good conductor of electricity, copper exhibits other interesting material properties. The characteristic impedance of copper is 80% of the characteristic impedance of stainless steel and it is slightly twice the characteristic impedance of aluminum. Meanwhile, the critical frequency by unit thickness (Hz-m) of a copper panel is about 50% greater than that of an aluminum or steel plate. In this

communication, a sound-absorbing device completely made of copper is presented, which makes use of a copper perforated panel and a copper fibrous infill material. The device would be useful to provide sound absorption in those environments where the protection of people's health is important.

11:25

4aAA12. Measurement verification of reverberation time in seating areas close to the lateral wall in the auditorium. Changqing Xia (BSE, The Hong Kong Polytechnic Univ., ZS801,Block Z, Hung Hom, Hong Kong, 20041829r@connect.polyu.hk) and Shiu Keung Tang (BSE, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

This study started with building a database for existed Jockey Club Auditorium. Using impulse response method to measure numerous seats which cover the whole auditorium. Reverberation time distributed in four range parameters simply look up the hall acoustic performance visualizing by chromatogram. With the hall geometric data and the first lateral reflection path difference calculated from the point of source and receiver in seating areas close to lateral wall, The measurement shows in bass and middle frequency, Early delay time and T_{20} and T_{30} exists large fluctuation which reveals seat-dip effect in the front seat area. While in mid and high frequency, T_{30} presents stable values indicating the direct sound and lateral reflection are predominant in early sound attenuation. Computer simulation is using Pachyderm Acoustical Simulation plug-in based on parametric architectural design software Rhino. Sound attenuation can be calculated with surface shapes defined surface transform efficiency by diffracting energy. However, Reverberation time differences still exist by comparing actual measurement and computer simulation results.

11:40

4aAA13. Sound properties in pre-Roman Etruria: an archaeoacoustic analysis of the Etruscan tomb space. Jacqueline K. Ortoleva (Classics, Ancient History and Archaeology, Univ. of Birmingham, Edgbaston, Arts Bldg., Birmingham B15 2TT, United Kingdom, Jxo644@bham.ac.uk) and Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., Houghton, MI)

The field of Archaeoacoustics has grown exponentially over the past decade providing clarity regarding sound propagation across multiple cross-cultural settings in the ancient world. However, many subfields of archaeology continue to overlook acoustic constructs in the archaeological record. Nowhere is this clearer than with ancient Etruria, a Pre-Roman Italic culture from Central Italy notable for their reverence of sounds generated from natural phenomena. This paper presents acoustic data collected for the first time inside a series of 5th–3rd century chambered tombs in Etruria. The relationship of architectural constructs and specific acoustic characteristics are noted across chronologies and regions. The study demonstrates the value of acoustical fieldwork when considering the visual and spatial properties of Etruria and provides the groundwork and recommendations for future exploration of sound propagation in Pre-Roman Italy.

Session 4aAB

Animal Bioacoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics: Applications of Bioacoustics in Killer Whale Conservation I

Tina M. Yack, Cochair
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Marla M. Holt, Cochair
Conservation Biology Division, NOAA/NMFS Northwest Fisheries Science Center, 2725 Montlake Blvd East, Seattle, WA 98112

Chair's Introduction—8:30

Invited Papers

8:35

4aAB1. Effect of interactive threats on foraging effort by endangered killer whales (*Orcinus orca*). Marla M. Holt (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd East, Seattle, WA 98112, marla.holt@noaa.gov), Jennifer Tennessen (Lynker, Inc. (under contract with NOAA NWFSC), Seattle, WA), M. Bradley Hanson, Candice Emmons (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), Deborah Giles (Friday Harbor Labs., Univ. of Washington, Davis, CA), Jeffrey Hogan (Cascadia Res. Collective, Olympia, WA), Eric J. Ward, Michael J. Ford (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), and Sheila J. Thornton (Ecosystem Sci. Div., Fisheries and Oceans Canada, West Vancouver, BC, Canada)

Killer whales face many anthropogenic threats including vessel traffic, noise, and reduced prey availability. Here, we use high-resolution suction-cup Dtags to study the behavior of endangered Southern Resident killer whales that rely on biosonar to hunt salmon, and investigate how proximate vessels affect foraging behavior. From tag data, we identified subsurface behavior, including foraging and prey capture events. We then tested several vessel and associated sound, demographic, and environmental variables on behavioral state occurrence, time spent within each state, and foraging effort involving prey capture. Whales made fewer prey capture dives and spent less time in these dives when vessels had an average distance <400 yard (366 m). Lower prey abundance and higher vessel speed reduced prey capture probability, documenting the interplay between these effects. Finally, whales dove to depth more slowly while increasing dive duration to capture prey in the presence of vessel-emitted sonar, but descended more quickly with higher noise levels and closer vessels. Current efforts investigating foraging behavior and noise exposure over the diel cycle aim to better quantify foraging rates and activity budgets in this endangered population. These findings advance awareness of vessels and noise consequences on killer whales to inform conservation and management actions.

8:55

4aAB2. Sound and movement data from two overlapping populations of killer whales reveal noise effects on foraging behavior. Jennifer B. Tennessen (NOAA Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd E, Seattle, WA 98125, jennifer.tennessen@noaa.gov), Marla M. Holt (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), Brianna Wright (Fisheries and Oceans Canada, Nanaimo, BC, Canada), M. Bradley M. Hanson (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), Candice Emmons (Northwest Fisheries Sci. Ctr., NOAA, Seattle, WA), Deborah Giles (Univ. of California at Davis, Davis, CA), Jeffrey Hogan (Cascadia Res. Collective, Olympia, WA), Sheila J. Thornton (Fisheries and Oceans Canada, West Vancouver, BC, Canada), and Volker Deecke (Sci., Natural Resources and Outdoor Studies, Univ. of Cumbria, Ambleside, United Kingdom)

Understanding ambient noise effects on foraging ecology of endangered killer whales is a necessary step toward predicting population-level consequences of acoustic disturbance. However, this has been limited by the difficulty of identifying prey capture which typically occurs out of sight, and by the challenge of obtaining a sufficient range of noise conditions. We addressed these problems using sound and movement data from suction cup-attached archival tags deployed on 52 Northern resident and endangered Southern resident killer whales sampled during overlapping field efforts. We measured broadband ambient noise levels above low-frequency cut-offs that minimized flow noise, and quantified foraging behavior using established acoustic and movement signatures to detect prey capture events. Statistical models revealed significant effects of noise level on searching, pursuit and capture outcomes, including the likelihood that a prey pursuit dive was aborted. We discuss the utility of broadening the range of noise level conditions available by sampling from multiple populations within overlapping geographic regions, and highlight the implications of noise-induced foraging interference on two killer whale populations with different population trajectories. Finally, we suggest future applications of this comparative approach to foster a greater understanding of the population consequences of acoustic disturbance in killer whales.

9:15

4aAB3. Masking escape in killer whales, stereotypical call design as a noise impact reduction strategy. Harald Yurk (Ecosystem Sci. Div., Fisheries and Oceans Canada, Vancouver, BC, Canada, harald.yurk@dfo-mpo.gc.ca), Caitlin O'Neill, Rianna Burnham, Christie Morrison, and Svein Vagle (Dept. of Fisheries and Oceans, Sidney, BC, Canada)

Killer whale call repertoires appear to remain relatively stable over generations because call design is considered socially transmitted between generations. Transmission errors during learning may be responsible for new call designs and repertoire size increases. Call and repertoire divergence may also be the result of adaptation to noisy environments. Temporal and spatial ambient noise variation occurs naturally in killer whale habitats and spans across the auditory frequency ranges of calls. Call components that vary in peak energy across frequency bands appear to escape auditory noise masking because peak amplitude variation per frequency is part of the stereotypical design of calls. As a result level variations in the noise spectrum rather than broadband noise dose may be the more relevant noise impact metrics for calls. Killer whale call design and repertoire divergence appears to reflect the naturally occurring variability in propagation of different frequencies. This may have led to repertoire variation among different populations allowing them to explore different acoustic niches. Presented are preliminary results from an ongoing study into call design influences on signal propagation under varying ambient noise conditions at different times of the year and different geographic locations and water depths within resident and Bigg's killer whale habitat.

9:35

4aAB4. Auditory masking in killer whales (*Orcinus orca*) with realistic signals and noise. Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Mike Felice (SeaWorld San Diego, San Diego, CA), Todd Robeck (SeaWorld Parks and Entertainment, Orlando, FL), Alyssa W. Accomando (National Marine Mammal Foundation, San Diego, CA), and Marla M. Holt (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., Seattle, WA)

Killer whales are the largest delphinid odontocete, a top predator and are globally distributed. Southern resident killer whales, which inhabit the Pacific coastal region of the northwest United States and Canada, are listed as endangered. Anthropogenic noise has been identified as a major threat to this populations of killer whales. This paper will cover recent results from killer whale auditory masking experiments using a variety of signals and noise types. Signal types included pure tones and recordings of killer whale calls. Noise types included Gaussian noise and recordings of seismic surveys and continuous active sonar. Potential auditory mechanisms that govern the resulting masking patterns, including dip listening, off-frequency listening, and comodulation masking release are discussed. These data will help to better understand and predict how killer whales can acoustically communicate, navigate, and locate prey in an increasingly noisy environment.

Contributed Papers

9:55

4aAB5. Passive acoustic monitoring of Southern Resident Killer Whales: Occurrence, residency and movement data in support of conservation. Katherine Gavrilchuk (Fisheries and Oceans Canada, 4160 Marine Dr., West Vancouver, BC V7V 1H2, Canada, katherine.gavrilchuk@dfo-mpo.gc.ca), Rianna Burnham, Svein Vagle, Jesslynn Shaw, Lynn Rannankari (Fisheries and Oceans Canada, Sidney, BC, Canada), Dylan Smyth, Erin Woodley, and Sheila J. Thornton (Fisheries and Oceans Canada, West Vancouver, BC, Canada)

Passive acoustic monitoring (PAM) of cetaceans offers valuable spatiotemporal information on presence and residency time to support population management and recovery strategies. Acoustic detections can be used to evaluate diel and seasonal occurrence patterns when visual detection opportunities are limited (e.g., night or winter), and acoustic encounter duration can inform habitat use patterns (e.g., travelling/shorter duration *vs* foraging/longer duration). Additionally, detection data from strategically placed recorders may also be used in the development of predictive movement models. Here we use PAM data to evaluate critical habitat use by the Endangered Southern Resident Killer Whale (SRKW) population in Juan de Fuca Strait and the Salish Sea, near the southwest coast of Vancouver Island, British Columbia. Continuous recordings were collected at three locations from May to October, 2018–2020, and used to assess presence and acoustic encounter duration. As this area experiences significant commercial vessel traffic and recreational vessel use, mitigation of risk from acoustic disturbance is a high priority for recovery. These PAM detection and encounter duration data were used to provide advice for management actions in support of SRKW recovery in Canadian Pacific waters.

10:10–10:25 Break

10:25

4aAB6. Soundscapes experienced by southern resident killer whales (*Orcinus orca*) in the Salish Sea and implications of vessel noise. Rianna Burnham (Inst. of Ocean Sci., Inst. of Ocean Sci., Sidney, BC V8L 5T5, Canada, rianna.burnham@dfo-mpo.gc.ca), Svein Vagle (Inst. of Ocean Sci., Sidney, BC, Canada), Pramod Thupaki (Hakai Inst., Victoria, BC, Canada), and Caitlin O'Neill (Inst. of Ocean Sci., Victoria, BC, Canada)

Endangered southern resident killer whales (SRKW, *Orcinus orca*) use waters of the Salish Sea around southern British Columbia and northern Washington State to forage, following the in- migration of prey. However, these waterways experience heavy vessel traffic, with important SRKW foraging areas near to international shipping lanes. Here we use recordings from six passive acoustic moorings to characterize the soundscape, and determine spatiotemporal patterns on daily to annual scales. Particular focus is given to the noise levels in frequencies used by SRKW to communicate and echolocate. A numerical model is used to map the sound field between the mooring locations and examine the anthropogenic inputs in SRKW critical habitat in the Salish Sea by commercial traffic. Extrapolation of the model output to higher frequencies allows an examination of the impact this may have on the range of effective acoustics use, with implications for SRKW navigation, foraging success and group cohesion. Finer scale analysis of vessel signatures as they transit the study area aids in identifying areas of increased risk of disturbance for whales. In addition, differences in modelled to observed data will help the characterisation of other sonic inputs to the soundscape, including smaller recreational vessels.

10:40

4aAB7. Managing vessel-generated underwater noise to reduce acoustic impacts to killer whales. Krista Trounce (Vancouver Fraser Port Authority, 100 The Pointe, 999 Canada Pl., Vancouver, BC V6C 3T4, Canada, krista.trounce@portvancouver.com), Alexander MacGillivray (JASCO Appl. Sci., Victoria, AB, Canada), and Jason Wood (SMRU Consulting, Friday Harbor, WA)

Launched in 2014, the Enhancing Cetacean Habitat and Observation (ECHO) Program, led by the Vancouver Fraser Port Authority, is a regional, collaborative program designed to better understand and reduce the cumulative effects from commercial shipping activities on at-risk whales. The program advances a number of research projects and operational initiatives, with a key focus on reduction of vessel-generated underwater noise to support recovery of the endangered Southern Resident Killer Whale (SRKW). To reduce vessel noise in key killer whale habitat, the ECHO Program manages seasonal voluntary slowdowns of large commercial traffic and lateral displacement of inshore traffic, and advances projects to better characterize vessel-generated noise. Determining the efficacy of vessel noise reduction efforts requires careful consideration of confounding factors to assess acoustic reductions, and estimate the resulting benefits to SRKW. This presentation will highlight results of noise mitigation efforts conducted to date, and identify some of the technical challenges faced in the acoustic analysis, including decisions on data filtering techniques. Methods used by the ECHO Program for evaluating potential effects of noise exposure on killer whale behavior and foraging will also be discussed.

11:00

4aAB8. Benefits of voluntary vessel slowdowns to acoustic space reduction for killer whales. Alexander O. MacGillivray (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, alex@jasco.com), Zizheng Li, David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Krista Trounce (ECHO Program, Vancouver Fraser Port Authority, Vancouver, BC, Canada)

Underwater noise from marine vessels can reduce the acoustic space that is available to marine mammals for performing critical life functions. Likewise, measures such as speed reductions that decrease vessel radiated noise can limit the extent to which acoustic space is affected by marine traffic. To investigate the benefits of voluntary summer vessel slowdowns in Haro Strait and Boundary Pass, the Enhancing Cetacean Habitat and Observation (ECHO) program commissioned a modelling study of acoustic space reduction which focused on two frequency bands relevant to resident killer whales: a communication band (0.5–5 kHz), using a metric termed Listening Space Reduction (LSR) and an echolocation band (15–100 kHz), using a metric termed Echolocation Space Reduction (ESR). The LSR and ESR metrics calculate a relative percent change from the maximum listening or echolocation space under natural ambient conditions to the reduced space caused by anthropogenic noise. Percent LSR and ESR was calculated in the study area at fine spatial (200 m) and temporal (1 min) resolution for scenarios representing baseline and slowdown traffic conditions. Results of this study demonstrate the benefits of voluntary vessel slowdowns on lost listening and echolocation space for resident killer whales.

11:20

4aAB9. Southern Resident killer whale acoustic monitoring and noise measurements to inform noise mitigation and potential management strategies. Jason Wood (SMRU Consulting, SMRU Consulting, PO Box 764, Friday Harbor, WA 98250, jw@smruconsulting.com), Kaitlin Palmer (SMRU Consulting, Vancouver, BC, Canada), Krista Trounce (ECHO Program, Vancouver, BC, Canada), Tina M. Yack (EcoSound Bioacoustics, Santee, CA), Frances Robertson (Environ. Stewardship & Marine Resources Committee, San Juan County, Friday Harbor, WA), and Harald Yurk (Ecosystem Sci. Div., Fisheries and Oceans Canada, Vancouver, BC, Canada)

SMRU Consulting, in collaboration with the ECHO Program, the San Juan County MRC and DFO, has been monitoring Salish Sea underwater soundscapes for the presence of Southern Resident killer whales (SRKW), to measure ambient noise levels, evaluate the efficacy of noise mitigation measures and inform potential management strategies. This talk will highlight results from four projects across the years 2019–2021. This includes measurements from a cabled observatory at Lime Kiln Point State Park in support of the ECHO Program's Haro Strait commercial vessel slowdowns; measurements from autonomous recorders at five locations in Burrard Inlet for the ECHO Program and the Tsleil-Waututh Nation to monitor trends in ambient noise and marine mammal presence; measurements from Coastal Acoustic Buoys at eight locations south of San Juan Island in support of San Juan County's management efforts, and a Coastal Acoustic Buoy for Offshore Wind deployment near Point Roberts to allow DFO to evaluate acoustic technology. This talk will discuss the pros and cons of these four different acoustic systems and highlight trends in the data that may help inform mitigation and management strategies.

11:40

4aAB10. Acoustic survey of Southern resident killer whale occurrence at the Lime Kiln hydrophone observatory, Haro Strait, Washington. Tina M. Yack (EcoSound Bioacoustics, 9423 Saint Andrews Dr., Santee, CA 92071, tina.yack@ecosoundbioacoustics.com), Jason Wood, and Dominic Tollit (SMRU Consulting North America, Friday Harbor, WA)

The aim of this study is to explore temporal and diurnal variation in Southern Resident Killer Whale (SRKW) acoustic activity and their relationship to other marine mammal events and to anthropogenic activity in Haro Strait, Washington. In support of the Vancouver Fraser Port Authority-led ECHO (Enhancing Cetacean Habitat and Observation) Program, six years (2016–2021) of summer/fall acoustic data from the Lime Kiln live hydrophone system (range: 10 Hz–100 kHz) in Haro Strait has been analyzed using PAMGuard software (64-bit Version: 1.15.11; Gillespie *et al.*, 2008). Automated whistle detectors and echolocation click detectors and classifiers were parameterized to detect marine mammals, ships, and echosounders in the dataset. PAMGuard Viewer Mode was then used to review the data and mark all marine mammal and anthropogenic events. The dataset provides important long-term information on marine mammal occurrence and anthropogenic activity in a well described core area of SRKW critical habitat. Daily event logs compiled using custom R script have been used by the ECHO program to better understand the effectiveness of multi-year voluntary commercial vessel slowdowns through Haro Strait and considerable year-to-year and monthly variation in SRKW detections have been identified. Coupled with long-term day-time visual observations of SRKW, the dataset is also useful for assessing the efficacy of PAM detection mitigation methods and night-time activity levels.

Session 4aAO

Acoustical Oceanography, Signal Processing in Acoustics, ASA Committee on Standards, Engineering Acoustics, and Computational Acoustics: New Data Sources and Processing Techniques in Fisheries Acoustics: Revolutionary or Incremental Progress?

Orest Diachok, Cochair
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Alex De Robertis, Cochair
NOAA Fisheries, NOAA Alaska Fisheries Science Center, 7600 Sand Point Way NE, Seattle, WA 98115

Christopher Bassett, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, NJ 98105

Chair's Introduction—7:50

Invited Papers

7:55

4aAO1. Decades of progress in active fisheries acoustics: Many challenges remain. Timothy K. Stanton (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., MS11, Woods Hole, MA 02543, tstanton@whoi.edu)

The technology in this field has evolved significantly over the past 6 + decades to one involving operational scientific echosounders (single-, split-, and multi-beam; broadband) used across a multitude of types of platforms. In parallel with hardware development has been significant development of various methods of data processing, scattering models, and associated interpretation, yielding new understanding of behavior of fish and zooplankton. Overall, this field is very active and productive as well as one that is fertile for new growth and understanding. Trends of development, key milestones, and current challenges are briefly reviewed.

8:15

4aAO2. Promises and challenges of fisheries acoustics: When biology collides with physics. John Horne (School of Aquatic and Fishery Sci., Univ. of Washington, Box 355020, Seattle, WA 98195, jhorne@uw.edu)

Active acoustic technologies have been used in fisheries resource management and ecological research since the 1920s. Since the first applications, both technological and analytic advances have been developed, vetted, and then rapidly adopted once a science or data need has been identified. In parallel, increased availability of off-the-shelf data acquisition and processing tools has broadened the traditional fisheries acoustics community from engineers, mathematicians, and physicists to include biologists and statisticians. The resulting wider availability of acoustic technologies, integration of instrument packages, and a more diverse user community has increased the number of acoustic-based studies. But like any other sampling tool, acoustics in isolation does not provide a definitive solution to mapping, counting, and/or identifying distributions and dynamics of interacting aquatic populations. Challenges of identifying spatial and temporal population distribution boundaries in response to evolving environmental conditions are amplified as management mandates and research scopes expand. Despite the challenges, integration of technological and data processing advances applied to study design and analytic workflow continues to increase the range of acoustic applications that address both research and management goals.

8:35

4aAO3. Riding the wave of oceanography's robot revolution to advance fisheries acoustics. Kelly Benoit-Bird (Monterey Bay Aquarium Res. Inst. (MBARI), 7700 Sandholdt Rd., Moss Landing, CA 95003, kbb@mbari.org)

A proliferation of uncrewed surface and underwater platforms are providing new ways to obtain acoustic data in the ocean, offering increased spatial and temporal coverage and access to unexplored areas. There are challenges to effectively exploiting robotic platforms for fisheries acoustics including difficulties in integrating relatively large, high power draw scientific echosounders into these platforms and the need to provide alternative evidence for target identification in the absence of trawling vessels. Recent experiments that integrated a constellation of robotic platforms and a cabled observatory to describe the abundance, distribution, and behavior of pelagic animals illustrate both the opportunities and challenges. Echosounders were employed from a variety of uncrewed platforms, letting us examine the tradeoffs of active acoustic sensing from each platform. Echosounder sampling was complemented by targeted environmental DNA sampling and quantitative video transects from autonomous underwater vehicles to provide information on target taxa and size classes. Together, these data revealed new details on the daily vertical reorganization of life in the pelagic. Creatively exploiting the

strengths of these new platforms rather than attempting to (poorly) replicate ship-based sampling is key to leveraging the robot revolution in ocean sciences.

8:55

4aAO4. Information extraction, scalable processing, open-source development, and community engagement: Data science in fisheries acoustics. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu)

The past decade has seen an explosion of the availability of fisheries acoustic data, driven primarily by the successful integration of echosounder on a variety of ocean observing platforms and a broader adoption of active acoustics as a tool for ocean sensing. However, we are just at the cusp of creating data processing techniques and workflows that can truly transform these data into information. In this talk, I will discuss recent advancements in multiple areas that I believe when combined, harbor the potential to accelerate research progress in fisheries acoustics toward making revolutionary changes: integration of physics-based models and data-driven methods, including machine learning; cloud computing; open-source software development; interoperable data and data standards; and community engagement, which plays a crucial role in driving rapid exchange of ideas and spurring innovation in not only fisheries acoustics, but also the broader oceanography and geoscience fields.

Contributed Papers

9:15

4aAO5. Fisheries acoustics in Norway—Wide band data, autonomous platforms and deep learning. Nils Olav Handegard (Inst. of Marine Res., Nykirkekaien 1, Bergen 5004, Norway, nilsolav@hi.no), Forland N. Tonje, Espen Johnsen, Geir Pedersen, Maria Tenningen, Rolf Korneliussen, and Egil Ona (Inst. of Marine Res., Bergen, Norway)

This presentation gives an overview of recent achievements in fisheries acoustics associated with the Center for Research Based Innovation in Marine Acoustic Abundance Estimation and Backscatter Classification (CRIMAC), Norway. We present a data processing pipeline from raw acoustic data to survey indices of abundance for fisheries assessments as a test bench. This includes the raw data as well as manually worked up labels for acoustic target classification (ATC). We also collect data from dedicated CRIMAC surveys and organize broadband data across a range of species, using different echo sounder setting, to find optimal settings for various standard surveys. We have trained deep neural networks on labelled acoustic data for ATC and tested this in the pipeline. Semi-supervised methods are also developed, requiring only 10% of the training data compared to the supervised models. IMR has a developed a strategy for phasing in unmanned platforms in our surveys. The process is stepwise, where we first augment existing surveys to maintain time series integrity; this approach has been tested on our sand eel and sprat surveys. We also embed our deep learning models in containers to facilitate simple deployment. This allows adaptive survey strategies, which is a step towards fully autonomous acoustic surveys.

9:30–9:45 Break

9:45

4aAO6. The use of deep learning techniques in acoustic tag signal identification. Tracey W. Steig (Innovasea, 711 NE Northlake Way, Seattle, WA 98105, tracey.steig@innovasea.com), Patrick A. Nealson (Consulting Services, Innovasea, Seattle, WA), Jean Quirion, and Frank Smith (Res. and Development, Innovasea, Bedford, NS, Canada)

Acoustic telemetry is routinely employed to detect and quantify behaviors of aquatic animals. The ability to acoustically tag and release fish and aquatic animals allows researchers to monitor their presence/absence and fine-scale spatial movement. One method of identifying individual tags employs a period signal encoding (HTI signal) method, which utilizes the full transmitted acoustic energy for both detection and identification, providing significantly greater detection ranges than encoded BPSK signal types. However, identifying period encoded tags typically requires manual data review by trained human analysts, resulting in increased processing cost and a lack of scalability. Innovasea conducted a project to determine the efficacy of applying modern Deep Learning techniques to automatically identify animals implanted with period encoded acoustic tags. Specifically, the well-known U-Net model was adapted and trained to detect period encoded acoustic tag signals from raw acoustic telemetry data. The raw data is first transformed into a spectrogram-like image representation and the U-Net learns an image mask that separates actual tag detections from noise and interference. The trained U-Net achieved greater than 95% acoustic tag identification accuracy when compared to tag identification performed by human analysts on the same test data. Numerous examples of the U-Net performance will be presented.

10:00

4aAO7. Where do the fish come from, and where do they go? Seasonal observations of pelagic fishes from echosounder moorings in the Chukchi sea. Robert Levine (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98195, leviner@uw.edu), Alex De Robertis (NOAA Alaska Fisheries Sci. Ctr., Seattle, WA), Daniel Grunbaum (School of Oceanogr., Univ. of Washington, Seattle, WA), and Christopher Wilson (NOAA Alaska Fisheries Sci. Ctr., Seattle, WA)

Recent summer surveys of the Chukchi Sea determined that pelagic fishes were dominated by large numbers of age-0 Arctic cod and walleye pollock, while adult fishes are comparatively scarce. Modeling based on regional currents indicates that these age-0 fishes are likely advected to the north in fall; however, the source and fate of these fishes remains unclear as this region is seasonally ice-covered. To determine the movement and variability of this age-0 gadid population, bottom-moored multifrequency echosounders were deployed at three locations in the northeastern Chukchi Sea from 2017-2019. These observations indicate that the abundance and composition of the pelagic community on the Chukchi Sea shelf is highly variable over seasonal time scales. Fish abundance was very low in winter, increased in May, and reached peak abundance in late summer. Target strength and diel vertical migration of fishes increased in summer, indicating that this is a key period for growth. Age-0 gadids were displaced to the northeast, consistent with the dominant advection on the shelf. Fish speeds and headings were strongly correlated with local currents, providing evidence that these small age-0 fishes are primarily being passively transported and behavior plays a limited role in population distribution.

10:15

4aAO8. Uncrewed surface vehicle survey of walleye pollock, *Gadus chalcogrammus*, in response to the cancellation of ship-based surveys. Alex De Robertis (National Oceanic and Atmospheric Administration, NOAA Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Seattle, WA 98115, alex.derobertis@noaa.gov), Mike Levine, Nathan Lauffenburger, James Ianelli, Cole Monnahan, Sarah Stienessen, and Denise McKelvey (National Oceanic and Atmospheric Administration, Seattle, WA)

In 2020, the developing COVID-19 pandemic disrupted fisheries surveys to an unprecedented extent. Many surveys were cancelled, including those for walleye pollock (*Gadus chalcogrammus*) in the eastern Bering Sea (EBS), the largest fishery in the United States. To partially mitigate the loss of survey information, we deployed three uncrewed surface vehicles (USVs) equipped with echosounders to extend the ship-based acoustic-trawl

time series of pollock abundance. Trawling was not possible from USVs, so an empirical relationship between pollock backscatter and biomass established from previous surveys was developed to convert USV backscatter observations to pollock abundance. The EBS is well-suited for this approach since pollock dominate midwater fishes in the survey area. Acoustic data from the USVs were combined with historical surveys to provide a consistent fishery-independent index in 2020. This application demonstrates the unique capabilities of USVs and how they could be rapidly deployed to collect information on pollock abundance and distribution when a ship-based survey was not feasible. We note the limitations of this approach (e.g., higher uncertainty relative to previous ship-based surveys), but found the USV survey to be useful in informing the stock assessment in a situation where ship-based surveys were not possible.

10:30

4aAO9. Target density estimation of mesopelagic fish using a towed, broadband, split-beam, acoustic scattering system. Zhaozhong Zhuang (Appl. Ocean Phys. & 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. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The ocean mesopelagic zone (200–1000 m) features a complex temporal and spatial distribution of organisms which may play an important role in ocean biogeochemical cycles. Yet a quantitative account of the abundance, biomass and migration behaviors of mesopelagic inhabitants is still lacking. Shipboard echosounders are limited by acoustic attenuation at higher frequencies constraining the usable frequency band for investigating deep sound scattering layers, as well as large acoustic sampling volumes, a result of the large depths associated with the deep scattering layers, combined with the inherent complexity of mesopelagic communities. Deep-See, a towed platform carrying multiple broadband sonars, focuses on individual targets in the deep scattering layers, to help interpret shipboard data. Target density estimation is the first step towards determining biomass in the mesopelagic zone. Two methods are investigated and applied to a summer 2019 dataset collected using two broadband split-beam echosounders (5.5–18.5 kHz, 18–47 kHz) on Deep-See in the slope waters off the New England Continental Shelf. The echo-counting method utilizes echo magnitude and phase information to determine singe targets while echo-statistics involves comparison of data to a physics-based numerical simulation to predict the statistics of broadband echoes from which target density can be extracted.

10:45

4aAO10. Assessment of the impact of broadband acoustic backscattering techniques for understanding the abundance, biomass, and distribution of the Ocean Twilight Zone. Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA, alavery@whoi.edu)

Mesopelagic fishes, which inhabit the Ocean Twilight Zone spanning ocean depths from 200-1000 m, are some of the most abundant vertebrates on earth, and are thought to play an important role in global ocean biogeochemical cycles. And yet there remain large uncertainties in their abundance, biomass, distribution, and behavior. Mesopelagic fishes are difficult to sample, or harvest for the purpose of commercial fisheries, due to their often remote and inhospitable habitats, and their ability to efficiently avoid trawl nets and cameras. As a result, there has been increased interest in using shipboard sonar systems to survey mesopelagic fishes, though little is known about their acoustic properties and target strengths. Recent advances in broadband acoustic scattering systems, post-processing analysis and approaches, and *in situ* platforms are being used to fill the technology gap and infer *in situ* target strengths of these elusive mesopelagic fishes. This talk presents an overview of the work done in this area to date, discusses the key challenges that still remain, and assesses the impact of broadband acoustic techniques in advancing our knowledge of the abundance, biomass, and distribution of mesopelagic fishes in the Ocean Twilight Zone. [Work supported by NSF and the WHOI OTZ Project.]

11:00

4aAO11. Scalable, probabilistic echo inversion for multifrequency and broadband acoustic surveys. Samuel S. Urmy (NOAA Fisheries, NOAA Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Seattle, WA 98115, sam.urmy@noaa.gov) and Alex De Robertis (Alaska Fisheries Sci. Ctr., NOAA Fisheries, Seattle, WA)

Identifying biological scatterers is a perennial challenge in fisheries acoustics. Most practitioners classify backscatter based on direct sampling and frequency-difference thresholds, then integrate at a single frequency. However, this approach struggles with species mixtures, and discards multifrequency information when integrating. Inversion methods do not have these limitations, but are seldom used, because their species identifications are often ambiguous and their algorithms complicated to implement. We address these shortcomings with a probabilistic, Bayesian inversion method. Like other inversion methods, it handles species mixtures, uses all available frequencies, and extends naturally to broadband signals. Unlike prior approaches, it leverages Bayesian priors to rigorously incorporate information from direct sampling and biological knowledge, constraining the inversion and reducing ambiguity in species identification. Because it is probabilistic, it can be trusted to run automatically: it should not produce solutions that are both wrong and confident. Unlike some data-driven machine learning models, it is based on acoustical scattering processes, so its inferences are physically interpretable. Finally, the approach is straightforward to implement using existing Bayesian libraries, and is easily parallelized for large datasets. We present examples using simulations and field data from the Gulf of Alaska, and discuss possible extensions and applications of the method.

11:15

4aAO12. Data-mining previous surveys to improve walleye pollock target strength estimates. Nathan Lauffenburger (Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, NOAA Alaska Fisheries Sci. Center, 7600 Sand Point Way NE, Seattle, WA 98115, nathan.lauffenburger@noaa.gov), Alex De Robertis, and Kresimir Williams (Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Seattle, WA)

Acoustic-trawl surveys are widely used to measure the abundance and distribution of pelagic fishes. The echo integration method used in these surveys requires estimates of the target strength (TS, dB re 1 m^2) of individuals to estimate abundance. *In situ* TS measurements are attractive to establish length-to-TS relationships because they are obtained under realistic conditions utilizing the same tools as those during surveys. However, these measurements are usually made in restricted circumstances, are limited in number, and data processing is often subjective. Here, we present a novel method to estimate TS from a large volume of previously collected survey data recorded at trawl sites. By applying a series of published filtering methods to TS data based on frequency response, packing density, and degree of co-location in overlapping beams, single fish echoes can be reliably isolated from existing survey data. The automated approach produced measurements representative of survey conditions (e.g., collected at survey speeds on free-swimming fish) from existing surveys. We applied this method to 30 surveys of walleye pollock (*Gadus chalcogrammus*) in Alaska and estimated a new length-to-TS relationship. The results were largely consistent with the relationship currently used for walleye pollock; depth and geographic area were significant covariates.

11:30

4aAO13. Applying deep learning to imaging sonar for the automated detection, classification and counting of untagged fish in fish passages. Vishnu Kandimalla (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Matt Richard, Frank Smith (Res. and Development, Innovasea, Bedford, NS, Canada), Tracey W. Steig (Innovasea, 711 NE Northlake Way, Seattle, WA 98105, tracey.steig@innovasea.com), Chris Whidden (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), and Patrick A. Nealson (Consulting Services, Innovasea, Seattle, WA)

The Ocean Aware project, led by Innovasea and funded through Canada's Ocean Supercluster, is developing the next generation of underwater observation systems to transform fishing, aquaculture, and marine energy. One facet of this project is developing innovative methods for real-time tracking of untagged fish and species at risk around man-made infrastructures, such as hydropower dams, that present barriers to fish passage. The system is based on applying modern Deep Learning techniques to automatically detect and classify fish and their species from high resolution imaging sonar. In this paper, we present our results from applying adaptations of the widely used YOLO machine learning model to detect and classify multiple species of fish from a public dataset containing eight distinct species of fish from the Ocqueoc River captured by a high resolution DIDSON imaging sonar. Our results demonstrate that deep learning models can indeed be used to detect, classify species, and track fish using high resolution imaging sonar. Although there has been extensive research in the literature identifying particular fish, such as eel versus non-eel and seal versus fish, to our knowledge this is the first successful application of deep learning for classifying multiple fish species with high resolution imaging sonar.

11:45

4aAO14. Classification with U-Net neural networks of herring, salmon, and bubbles in multifrequency echograms. Alex L. Slonimer (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, slonimer@uvic.com), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Alexandra B. Albu, Melissa Cote, Tunai P. Marques, Alireza Rezvanifar (Elec. and Comput. Eng., Univ. of Victoria, Victoria, BC, Canada), Stephane Gauthier (Fisheries and Oceans Canada, Victoria, BC, Canada), and Steve Pearce (ASL Environ. Sci., Saanichton, BC, Canada)

Convolutional neural networks were used to accurately identify biological and physical phenomena in multifrequency echograms. The primary classes of interest for this study were herring and juvenile salmon. Secondary classes included gas bubbles and the air-sea interface. A collection of U-Nets were trained to apply pixel-level classification. In addition to recorded acoustic data at four frequencies (67.5, 125, 200, 455 kHz), simulated data were created to provide context for the interpretation of the echograms. This contextual data included a channel for water depth and another channel for solar elevation angle. It is demonstrated that the inclusion of these context channels improve the efficacy of the classifier. After the pixel-level classification was performed, traditional school detection methods were applied to classify distinct schools of herring and aggregations of salmon. The pixel-level classification results were refined using the mean classification score of each school. To ensure broad applicability, this network was trained to classify echograms with noise left intact. It is also demonstrated that this method is scalable and effective at classifying data with a lower sampling rate than that used for training. The best performing model classified herring, salmon, and bubble classes with F_1 scores of 93.0%, 87.3%, and 86.5%, respectively.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics III

Christopher Elmer, Chair
American Mathematical Society, 17231 Lands End, Chelsea, MI 48118

Contributed Papers

8:00

4aMU1. On relieving aging vocalizations of famous trot singers by sound massage. Myungjin Bae (Commun. Eng., Soongsil Univ., 369 Sangdoro, Dongjakgu, Seoul 156-743, Republic of Korea, mjbae@ssu.ac.kr) and Kwang-Bock You (Commun. Eng., Soongsil Univ., Seoul, Republic of Korea)

Most of the songs of famous trot singers sing folk songs of sorrow, and their popularity has continued for decades. However, aged singing voices feel flat and monotonous compared to the loud singing voices of their prime. In order to improve aging vocalizations, it is sometimes used to calm the mind and body through ingestion of a tonic or neck massage. However, as the singing voice ages, the spectrum bandwidth gradually decreases, and the chords of the voices become simpler, so listeners do not feel the magnificence of the songs well. In this paper, a new method of relieving the aging characteristics of famous singers through sound massage of sound mug cups is proposed. In other words, based on the past vocalizations of famous trot singers, the singing spectrum was restored in its prime by reflecting the singing waveform from the mug cup and massaging the inner epithelium of the vocal cords. Although the bandwidth of a famous trot singer's voice has been reduced by –27% for 30 years when the proposed sound cup massage is applied, the aging spectrum is restored by more than 90% compared to the heyday, and most of the chords in the song come to life.

8:15

4aMU2. The effect of resonator shape on air flow in flue organ pipes. Elizabeth Buzbee (Dept. of Phys., Rollins College, Winter Park, FL 32789, ebuzbee@rollins.edu) and Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL)

The effect of changing the shape of the resonator of a flue organ pipe was studied in a replica of an organ pipe that was printed using transparent media. The pipe was designed so that the rectangular resonator could be replaced by a cylindrical one of the same cross sectional area, and the airflow in the pipe was imaged using high-speed transmission electronic speckle pattern interferometry. The resulting images reveal that there is a fundamental difference in the flow when the resonator shape is altered. Differences in the airflow and sound attributable to the shape of the resonator will be discussed.

8:30

4aMU3. Acoustics and temporality in an Indian classical monsoon raag. Ahona Palchoudhuri (Dept. of Anthropology and Music, Brown Univ., 128 Hope St., Giddings House, Providence, RI 02912, ahona_palchoudhuri@brown.edu)

Indian Classical Music is structured such that every raag, or scalar formation, elicits the atmosphere of a season or time of day. Yet, musicologists and theorists of sound have primarily studied the form within urban concert halls—often dislocated from the raag's originary temporal context and its relation to the earth. My paper studies the monsoon raag "Malhar" within the seasonal soundscapes of drought-prone rural India. It focuses in particular on how Malhar is sung at a village shrine during the course of a rain-making-ritual, conducted annually at the brink of the monsoon season. My paper examines how the musical strategies of Malhar transform when it moves from concert hall to shrine. Specifically, it compares how each Malhar renders the call of the hawk-cuckoo, known across the Indian sub-continent to usher in the monsoons. My research shows that while concert hall audiences decry singers' renditions of the cuckoo's call as "gimmicky" and "mere mimicry," followers at the shrine ascribe to it a ritual potency, and rely on it to evoke rain. I study this difference in musical perception as a way in which to better understand how acoustical and temporal context intersect, in concert hall and shrine, respectively, to transform the aesthetic experience of Indian Classical Music.

8:45

4aMU4. Investigating the origins of pitch modulation in flute vibrato. Sarah R. Smith (Dept. of Elec. and Comput. Eng., Univ. of Rochester, 617 Comput. Studies Bldg., 160 Trustee Rd., RC 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

Like most musicians, flute players often add vibrato to their playing for expressive effect. This is accomplished by a periodic modulation of the supplied air pressure resulting in significant amplitude and timbre variations of the resulting sound. However, a sizeable modulation in pitch also results in most cases with a width ranging from 20 to 50 cents of total variation. This work investigates the physical origins of this pitch variation based on an analysis of multiple databases of flute vibrato tones. Detailed tracking of the amplitude and pitch trajectories of each overtone is used to characterize the extent of amplitude and frequency modulation of each partial as well as their relative correlation. A high degree of correlation between amplitude and frequency modulation tracks suggests that both variations are controlled by the same physical parameter. However, the mechanisms for pressure-controlled pitch modulation are not well understood and many physically modeled synthesis algorithms do not produce this pitch variation when the input pressure is modulated. As such, this work systematically modifies existing physical models to reproduce the characteristics of recorded vibrato tones and evaluate potential mechanisms that may underlay the observed pitch modulations.

9:00–9:15 Break

9:15

4aMU5. Do lyrics help individuals to sing in tune? Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, 901 South Sixth St., Champaign, IL 61820, simin.soleimanifar@gmail.com), Hannah E. Staisloff (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL), and Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Normal hearing (NH) individuals may have difficulty in accurately recreating a familiar song by manipulating recordings of a sung vowel. This could indicate that lyrics play a key role in accurately producing familiar contours. This research investigated if that is the case. Five NH adults participated in this research. Participants were recorded singing *Happy Birthday* once with the lyrics and once when the lyrics were replaced with the vowel /a/. The preliminary results indicated that, at the group level, there was no significant difference in the accuracy of the produced contour when

singing *Happy Birthday* with and without lyrics. Additionally, the notes produced both with and without lyrics were significantly and highly correlated with each other. However, four out of five subjects produced more accurate songs when singing with lyrics. The results suggest that, if lyrics improve singing accuracy, the effect is small.

9:30

4aMU6. Untangling the effect of the reed resonance and tonehole lattice cutoff frequencies on the internal and external spectra of a clarinet-like instrument. Erik A. Petersen (Lab. de Mecanique et d'Acoustique, 4 Impasse Nikola Tesla, Marseille 13013, France, erikalanpetersen@gmail.com)

The spectrum characteristics of reed instruments are determined by many factors, some of which are related to musician control and others that are intrinsic to the physics of the instrument. Two such factors are the reed resonance frequency and the tonehole lattice cutoff frequency. The former can be manipulated to some degree by the musician, while the latter is fixed by the geometry of the air column. Although both of these variables are known to impact the spectrum of a woodwind instrument, it is complicated to evaluate the relative importance of their individual contributions, which may present in similar manners. Here, digital synthesis coupled with a simple radiation model is used to untangle the dual effects of these two characteristic frequencies on the internal and external spectra of a clarinet-like instrument. The amplitudes of the even harmonics in the internal and external spectra are boosted at the reed resonance frequency. While the tonehole lattice cutoff frequency also boosts the even harmonics in the internal spectrum, there is an additional overall increase in radiation at the cutoff which emphasizes both the even and odd harmonics in the external spectrum, implying that it may have a larger impact on the perceived timbre.

9:45

4aMU7. Preliminary report on the effect of room acoustics on choral performance in Notre-Dame and its pre-Gothic predecessor. Sarabeth Mullins (Inst. Jean le Rond d'Alembert, CNRS, Sorbonne Univ., 4 Pl. Jussieu, Paris 75005, France, Sarabeth.Mullins@sorbonne-universite.fr), Valérie Le Page (Inst. de Recherche en Musicologie, Sorbonne Univ., Paris, France), Julien De Muynke (I), Elliot K. Canfield-Dafilou (Inst. Jean le Rond d'Alembert, CNRS, Sorbonne Univ., Paris, France), Frédéric Billiet (Inst. de Recherche en Musicologie, Sorbonne Univ., Paris, France), and Brian F. Katz (Inst. Jean le Rond d'Alembert, CNRS, Sorbonne Univ., Paris, France)

The study of acoustics as intangible cultural heritage is an active field of research with implications beyond the scientific community. This study examines the influence of the acoustics of the new Gothic cathedral of Notre-Dame de Paris on the medieval singers who pioneered a style of plainchant known as the School of Notre-Dame using interactive virtual acoustic environments (VAEs) and real-time auralization. The experiment compared the approximated acoustics of the prior cathedral (torn down in 1163 CE to make room for Notre-Dame) with Notre-Dame which partially opened in 1182 CE. A speculative VAE of the pre-1163 cathedral was created by modifying a calibrated model of an extant and contemporaneous Roman basilica, while the early Notre-Dame VAE was generated by time-regressing a calibrated model of the modern cathedral to match its earlier state. A choir specializing in historically informed performance practices was studied as they sang *Organum Purum* and *Organum Notre-Dame* inside the VAEs of the pre-1163 cathedral and the early cathedral of Notre-Dame de Paris. Musical parameters were extracted from the recordings to examine what influence the different architectures may have had on musicians' performances.

10:00

4aMU8. Neural plasticity for music processing in young adults: The effect of transcranial direct current stimulation (tDCS). Eghe Adodo (St. Johns Univ., Bayside, NY), Cameron Paterson (St. Johns Univ., Jamaica, NY), and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Transcranial direct current stimulation (tDCS) has increasingly been proposed and utilized as a unique approach to enhance various communicative, cognitive, and emotional functions. However, it is not clear whether, how, and to what extent, tDCS influences nonlinguistic processing such as music processing. The purpose of this study was to examine cortical responses to music as a result of noninvasive brain stimulation. We used a passive listening multiple oddball paradigm consisting of six types of music pattern changes (rhythm, intensity, slide, location, pitch, and timbre) and recorded the event-related potentials (ERP) using 65-electrode sensor nets. Healthy young adults were tested before and after transcranial direct current stimulation (tDCS). We hypothesized that short-duration one-session tDCS would enhance some aspect of music processing. Our results indicated that the differences between pre- and post-tDCS ERPs were only evident in some conditions. Noninvasive brain stimulation, including tDCS, has the potential to be used as a clinical tool for enhancing auditory processing, but further studies need to examine how experimental parameters (dosage, duration, frequency, etc) influence the neural responses for auditory processing.

Session 4aNSa

Noise and ASA Committee on Standards: Aircraft Sonic Boom and Community Noise Assessment

Alexandra Loubeau, Chair
NASA Langley Research Center Hampton, VA 23681

Chair's Introduction—8:15

Invited Papers

8:20

4aNSa1. Investigating the impact of ambient noise on candidate sonic boom metrics. Mark C. Anderson (Phys. and Astronomy, Brigham Young Univ., D-70 ASB, Provo, UT 84602, anderson.mark.az@gmail.com), Kent L. Gee, J. T. Durrant (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Alexandra Loubeau, and William Doebler (NASA Langley Res. Ctr., Hampton, VA)

One goal of NASA's X-59 flight campaign will be to collect low-amplitude sonic boom data over several communities. The combination of lower-amplitude booms with greater ambient noise could pose challenges in measuring and calculating accurate sonic boom perception metrics intended to be correlated with human annoyance. This study uses two NASA sonic boom flight test and simulated X-59 datasets to investigate the impact of limited measurement bandwidth due to ambient noise for multiple candidate metrics. An estimate of the signal-to-noise ratio spectrum allows for the comparison of metrics calculated with full bandwidth, limited bandwidth, and those calculated by applying a low-pass filter that approximates a noise-free full-bandwidth spectrum. The low-pass filter is chosen to approximately match the expected high-frequency roll-off of the sonic booms. Results indicate that Perceived Level and A-weighted sound exposure level are strongly affected by ambient noise and bandwidth limitations, but that B-weighted sound exposure level is much less affected. These results contribute to community noise test planning by demonstrating the ambient noise impact on candidate sonic boom perception metrics. [Work supported by NASA Langley Research Center through National Institute of Aerospace.]

8:40

4aNSa2. Preliminary lasso regression analysis of environmental effects on sonic boom metric variability. J. T. Durrant (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, taggart.durrant@gmail.com), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Mark C. Anderson, Mylan R. Cook (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

NASA's Low-boom Flight Demonstration mission is a step toward commercial, overland supersonic flight. Certifying low-boom aircraft will require accurate measurement and understanding of uncertainty due to variations in meteorology, aircraft trajectory, and measurement environment. This work describes an analysis of the variability present in the Quiet Supersonic Flights 2018 (QSF18) dataset collected in Galveston, Texas, and an initial effort to correlate this variability with source and environmental factors. Although these data are for an F-18 executing a low-boom dive, no other low-boom dataset over a widespread populated area exists, and this study has the potential to guide future X-59 testing and data interpretation. Although several aircraft and meteorological variables are investigated for possible correlation, no single variable consistently explains the large variation in the measured booms. This has led to the use of lasso regression (LR) to identify the most relevant factors. Thus far, the fact that LR correlates boom Perceived Level (PL) with ambient PL is a strong indication of dataset contamination by significant ambient noise. Additionally, although several meteorological factors can contribute to boom metric variability, low-altitude wind speed is particularly important. [Work supported by NASA Langley Research Center through National Institute of Aerospace.]

9:00

4aNSa3. The effect of modeling dose uncertainty on low-boom community noise dose-response curves. William Doebler (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov), Aaron B. Vaughn (NASA Langley Res. Ctr., Provo, UT), Kathryn Ballard, and Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA)

In logistic dose-response modeling, failing to account for uncertainty in estimated doses can cause an artificial flattening or attenuation of the slope of the summary curve. In Lee *et al*. [*J. Acoust. Soc. Am.* **147**(4), 2222–2234 (2020)], data from two NASA low-amplitude sonic boom community noise survey tests were modeled using a Bayesian multilevel logistic regression (MLR) statistical model that assumed there was no uncertainty in the noise dose estimates. However, in these community tests, the noise dose uncertainty was estimated by Page *et al*. (NASA/CR-2014-218180 and NASA/CR-2020-220589/Volume I) using a leave-one-out method. In the current work, a term was added to extend the Bayesian MLR model to account for the estimated noise dose uncertainty quantified in the analyses of Page *et al*. This uncertainty term was included in two ways, either as classical or as Berkson uncertainty, and yielded similar results. When the uncertainty is accounted for in the Bayesian MLR model, the dose-response curves become 5%–10% steeper, but the difference in the noise dose that elicits a 5% highly annoyed response is small (less than 1 dB). This result is encouraging for future X-59 community tests whose survey area will be sparsely populated with noise monitors.

9:20

4aNSa4. Comparing two statistical models for low boom dose-response relationships with correlated responses. Aaron B. Vaughn (NASA Langley Res. Ctr., Eyring Sci. Center N201, Provo, UT 84602, aaron.b.vaughn@nasa.gov), William Doebler, Kathryn Ballard, and Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA)

This study compares two statistical modeling approaches to construct summary dose-response curves for low boom community noise surveys. NASA field survey data were used that consist of multiple responses from a survey participant. These data require an approach that accounts for the correlation among repeated annoyance observations from the same participant. A multilevel model accounts for the correlation by allowing estimated parameters to vary with each survey participant. On the other hand, a population average model utilizes generalized estimating equations and accounts for the correlation via a user-specified within-subject correlation structure. A visual comparison of the dose-response curves for these two methods reveals similar results. When comparing specific points along the summary curves, the multilevel model yields more precise confidence bounds than the population average model. The similarity between the summary curves derived from each model lends validity to both approaches for approximating a population representative summary curve.

9:40

4aNSa5. Experiment design considerations for longitudinal community noise surveys. Kathryn Ballard (NASA Langley Res. Ctr., Hampton, VA), Aaron B. Vaughn (NASA Langley Res. Ctr., Eyring Sci. Center N201, Provo, UT 84602, aaron.b.vaughn@nasa.gov), William Doebler, and Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA)

NASA is planning community noise surveys in the coming years to collect data that might enable the return of overland supersonic flight. Community noise surveys are typically cross-sectional, with only a single response per respondent. However, to determine community response to individual supersonic over flights, longitudinal surveys with repeated responses per respondent are more practical. The downside of repeated responses is the potential for response bias, such as a carryover effect in which previous test conditions affect current behavior. Subjective sonic boom listening test data collected at NASA Langley Research Center were reanalyzed to look for evidence of carryover effects. Initial results do not suggest strong carryover effects. This presentation will provide results and suggestions for experiment design of future community surveys.

THURSDAY MORNING, 2 DECEMBER 2021 402 (L)/403 (O), 10:15 A.M. TO 12:00 NOON

Session 4aNSb

Noise, Psychological and Physiological Acoustics and ASA Committee on Standards: General Topics on Noise-Induced Hearing Loss and Protection

Hanna Kurdila, Chair
U.S. Food and Drug Administration Silver Spring, MD 20993

Contributed Papers

10:15

4aNSb1. Infant ear canal modeling. Hannah Kurdila (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Silver Spring, MD 20993, hannah.kurdila@fda.hhs.gov), Joseph Vignola (Catholic Univ., Washington, DC), and Subha Maruvada (U.S. Food and Drug Administration, Silver Spring, MD)

MRI scanners produce significant sound pressure levels (SPLs), sometimes in excess of 120 dBA. While the effect of these SPLs has been explored for adult patients, it is currently unknown how they affect neonatal patients. One approach to understanding the effect of this noise exposure on infants is to use existing experimental data that associates a given sound level to a hearing outcome in adults and re-interpret it with respect to infants. This approach requires modeling the differences between the adult and infant hearing apparatus—outer (pinna + concha + ear canal), middle and inner ear. In this presentation, we will present simulations of the infant ear canal SPL gain and compare it to the expected acoustic exposure of the neonate during MRI brain procedures to better understand the effects of MRI noise on the neonatal ear.

10:30

4aNSb2. Non-lethal flashbang effectiveness pilot study: Acoustic effects. Caylin R. McCallick (Otolaryngol., Univ. of Colorado, Anschutz Medical Campus, 13001 E 17th Pl., Aurora, CO 80045, caylin.mccallick@cuanschutz.edu), Rebecca Welles, Carol Sammeth (Otolaryngol., Univ. of Colorado, Anschutz Medical Campus, Aurora, CO), Gregory Rule (Appl. Res. Assoc., Littleton, CO), Ted Argo (Appl. Res. Assoc., Aurora, CO), and Nathaniel Greene (Otolaryngol., Univ. of Colorado, Anschutz Medical Campus, Aurora, CO)

Flashbang grenades distract and temporarily disable an adversary, help gain surprise advantage, or disrupt communication in physical confrontations. Such devices are designed to emit high-amplitude acoustic signals, a bright light, and/or a concussive force. This pilot study evaluated the acoustic effects that accompany the detonation of these devices, namely temporary threshold shift (TTS), the acoustic startle reflex, and temporary noise-induced tinnitus. In this study, human subjects were evaluated on two tasks: (1) target localization and (2) speech-in-noise (SIN) testing. During these tasks, subjects underwent a random sequence of acoustic startle, simulated TTS, and simulated tinnitus conditions. While response accuracy and latency was monitored during these tasks, subjects' physical startle was also evaluated with electromyography (EMG) sensors, measuring muscle activity of the eyelid, jaw, neck, and bicep. Results suggest that each independent condition affects performance on localization and SIN tasks, with maximum suppressive effect occurring with all three main acoustic effects combined. As anticipated, the acoustic startle had a profound impact on the performance of at least some of subjects' listening ability in this pilot. Further work is necessary to fully characterize these effects in pursuit of creating a non-lethal flashbang device.

10:45

4aNSb3. Detection thresholds with active hearing protection devices in quiet and noise. Eric R. Thompson (711th Human Performance Wing, Human Systems Directorate, Air Force Res. Lab, 2610 7th St., Bldg. 441, Wright-Patterson AFB, OH 45433, eric.thompson.28@us.af.mil) and Nina J. Pryor (711th Human Performance Wing, Human Systems Directorate, Air Force Res. Lab, Wright-Patterson AFB, OH)

The Real-Ear Attenuation at Threshold (REAT) method is a standardized (ANSI/ASA S12.6) method of assessing the passive attenuation of hearing protection devices (HPD), including electronic HPDs with the electronics turned off. Characterizing HPDs with electronics turned on is performed with the Microphone in Real Ear (MIRE) method, or with an acoustic test fixture, as described in ANSI/ASA S12.42. It has been proposed that the REAT method may be useful for characterizing the effect of HPDs on detection of ambient sounds for HPDs with hear-through microphones, including the effects of electronics noise from the HPD. In this study, detection thresholds were measured with three electronic earmuff-type HPDs using a method similar to the REAT method. Detection thresholds were obtained for warble tones presented from a loudspeaker placed in front of the subject in a sound treated room. Thresholds were measured with unoccluded ears and with three HPDs with hear-through capability in quiet and in noise. In addition, thresholds were measured with one of the HPDs with foam earplugs inserted into both ears to simulate a conductive hearing loss. The results point to the interaction between device noise, ambient noise, and human absolute thresholds as limitations on the detectability of ambient sounds.

11:00

4aNSb4. Effects of mandibular motion on earplug function. David A. Eddins (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu), Steve Armstrong (Soundsgood Labs, Burlington, ON, Canada), Robert Budinsky, Nathan Higgins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), Roger Juneau (SoftTouch Labs, Harahan, LA), and Craig Formby (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL)

Earplug attenuation depends on developing and maintaining an acoustic seal between the earplug and the walls of the ear canal. During use, earplugs may lose some of their effectiveness by loosening over time and with activity, thereby compromising the acoustic seal in the ear canal and allowing more sound to enter the ear canal from the sound field. Mandibular motion (as during talking, singing, chewing, breathing) provides a nearly continuous source of ear canal movement resulting from forces applied by the condyle of the mandible. Individual differences in ear canal geometry also may impact the effects of mandibular motion on earplug function. The effects of mandibular motion on earplug function were investigated using an acoustic test fixture that combined (1) individualized pinna and ear canal structure beyond the second bend, inclusive of the mandibular bump, with (2) a standardized ear simulator (GRAS RA0045) with ear canal extension and microphone (GRAS 40AH 1/4"). Mandibular motion was simulated using a computer-controlled drive-pin with a silicone tip that articulated with the mandibular bump of the individualized silicon ear canal. Results indicated that as few as 50 cycles of mandibular motion could significantly compromise earplug function and that this compromise was highly subject-dependent.

11:15

4aNSb5. Analysis of multisource non-linguistic sound recognition among cochlear implant subjects. Ram-Charan M. Chandra-Shekar (CRSS-CILab: Ctr. for Robust Speech Systems—Cochlear Implant Processing Lab, Univ. of Texas—Dallas, 800 W Campbell Rd., Richardson, TX 75080, RamCharan.ChandraShekar@utdallas.edu) and John H. Hansen (CRSS-CILab: Ctr. for Robust Speech Systems—Cochlear Implant Processing Lab, Univ. of Texas— Dallas, Richardson, TX)

Naturalistic sounds carry rich information related to situational context or subject/system properties that contribute towards an acoustic awareness of the ambient environment. Sensorineural hearing loss reduces the functionality of the cochlea, auditory nerve, or central auditory pathways leading to degradation in auditory processing. Cochlear Implants (CIs) have been widely used to restore impaired auditory function and research advancements have focused on improving speech recognition and overall hearing-related quality of life. However, relatively few studies have investigated non-linguistic Sound Recognition (SR) among CI subjects. In this study, the recognition of sounds in CI users is assessed in a competing condition involving at least two non-linguistic sound sources. Furthermore, an end-to-end audio source separation neural network, SuDoRM-RF, with negative and permutation invariance training, is used for audio source recovery and comparatively assessed for potential identification improvement of non-linguistic sounds among CI users. Objective metrics such as classification accuracy, scale-invariant signal-to-distortion-ratio, and other audio quality related measures are used for assessment and subjective evaluation and compared against listener testing with CI subjects. The proposed study can model multi-source non-linguistic sound problems, the cocktail party effect and potentially provide an effective simulation for realistic listening test scenarios. [Study supported by NIH DC010494-01A.]

11:30

4aNSb6. A construction for the prediction of noise-induced annoyance in the presence of auditory masking. Andrew Christian (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., M/S 463, Hampton, VA 23681, andrew.christian@nasa.gov)

There is a large amount of evidence to suggest that people's reactions to annoying sounds are moderated by the presence of other background sounds. One of the mechanisms of this moderation is auditory masking: when the annoying sound simply cannot be heard over the background. This presentation outlines a construction for computing a predictor of annoyance in the presence of masking. The construction is demonstrated by producing an augmentation of the common sound exposure level noise metric using one-third octave band data sampled at 2 Hz. These data allow an estimation of the probability of masking within each band at each time step. The acoustic energy within a masked band can be 'discounted' by this estimation before being summed in the usual equal-energy way and with the typical A-weighting corrections. In this way, backgrounds that only provide partial masking in frequency and time will produce predictions of annoyance that fall between the unmasked sound exposure value and negative infinity (the result if there was a certainty of complete masking). The application of this concept to the noise of multirotor air vehicles is discussed.

11:45

4aNSb7. On a measurement of noise-induced hearing considering the age of both ears. Myungjin Bae (Commun. Eng., Soongsil Univ., 369 Sangdoro, Dongjakgu, Seoul 156-743, Republic of Korea, mjbae@ssu.ac.kr) and Seonggeon Bae (Commun. Eng., Soongsil Univ., Youngin Kyunggido, Republic of Korea)

The noise-induced hearing loss suffered in the multimedia living environment is difficult to detect because there are few symptoms in the early stages. Existing pure tone audiometry detects the response to whether pure tones are well heard for each frequency to 250–8000 Hz sound, which is mainly used in everyday conversation, but it is inaccurate and takes a long time to measure noise-induced hearing loss. Therefore, in this paper, we

propose a new measurement method that can easily and quickly self-measure noise-induced hearing loss in both ears through a smartphone APP. In this method, the age-compensated 9 pure tones for each voice subband are alternately heard in both ears at 10 kHz or higher, thereby rapidly determining one's own hearing loss. When 18 tone pulses were heard for 27 s per person, 68 out of 100 subjects in their 20 s were identified as suspected of partial hearing loss, and 27 including them were identified as having severe noise-induced hearing loss. And when the subjects were instructed on how to discriminate the number of hearings for noise-induced hearing loss, the participants themselves immediately determined whether their hearing was impaired. In the end, the noise-induced hearing loss measurement method in both ears, which took more than 10 minutes with the existing manual method, could be easily and accurately measured with our proposed method.

THURSDAY MORNING, 2 DECEMBER 2021 QUINAULT, 8:00 A.M. TO 12:00 NOON

Session 4aPA

Physical Acoustics and Biomedical Acoustics: Session in Honor of David T. Blackstock I

Mark F. Hamilton, Cochair
Walker Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1591

Thomas G. Muir, Cochair
Applied Research Labs, University of Texas, P.O. Box 8029, Austin, TX 78713

Chair's Introduction—8:00

Invited Papers

8:05

4aPA1. David T. Blackstock: Scholar, teacher, mentor, and gentleman. Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, hamilton@mail.utexas.edu)

David T. Blackstock was born in Austin, Texas in 1930. After earning B.S. and M.S. degrees in physics at University of Texas at Austin he spent two years at Wright-Patterson Air Force Base, where he developed an interest in acoustics working under Henning von Gierke on hearing protection. He then joined F. V. Hunt's group at Harvard University and earned a Ph.D. in applied physics in 1960 with a doctoral thesis in nonlinear acoustics, ultimately his lifelong passion. Following three years at General Dynamics and seven years at University of Rochester, David returned to UT in 1970 to join its Applied Research Laboratories, and in 1987 he became Professor of Mechanical Engineering. Among David's legacies is his formulation of a convenient framework for implementing weak-shock theory. Combining analytical, computational, and experimental approaches, he proceeded to investigate acoustic saturation, finite-amplitude noise, suppression of sound by sound, parametric array in air, sonic boom, therapeutic ultrasound, and other fundamental problems in nonlinear acoustics. David was widely known as an exceptionally caring man, whether bringing Soviet scientists and their research to the attention of the West, teaching and mentoring graduate students, or through extensive service in the ASA and International Commission for Acoustics.

8:25

4aPA2. David T. Blackstock: Friend, husband, father. Stephen P. Blackstock (Appl. Res. Lab., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, spb@arlut.utexas.edu)

David T. Blackstock, my father, was an exceptional and esteemed scholar, scientist, and teacher. He was an even more exceptional human being: a pillar of integrity, a kind and generous friend, a devoted husband, and a loving father. Like many in the audience, I learned so much from him. I have never understood how he had the energy to be so engaged in his research and teaching and, also, so attentive to his family. One of my older brother's childhood friends remarked, shortly after David's passing: "What made him so special was that he remained a good dad despite all the pressures of his professional life." In this talk, I'll explore this mystery, and I hope that, along the way, you'll get to know David Blackstock, the person, a little better. David taught many students how to be good scientists. He taught me how to be a good human being.

8:45

4aPA3. David Blackstock and the Applied Research Laboratories at The University of Texas at Austin (ARL:UT): Laying the Academic Foundation for an Applied Laboratory. Karl Fisher (Appl. Res. Labs., Univ. of Texas, 10000 Burnett Rd., Austin, TX 78758, kfisher@arlut.utexas.edu) and Clark Penrod (Appl. Res. Labs., Univ. of Texas, Austin, TX)

David T. Blackstock took a leave of absence from the University of Rochester in 1969 and returned to the University of Texas in his hometown of Austin. ARL:UT was extremely fortunate to have David accept a full-time appointment at the lab as a Faculty Research Scientist in 1970. Over the next fifty years, David nurtured and strengthened the bond between the academic and teaching environment on UT's main campus and the applied work conducted at ARL:UT. He helped establish coursework that has educated numerous ARL:UT staff members, many of whom had no prior knowledge of acoustics. David increased the recognition of ARL:UT's acoustic research by attracting internationally renowned scholars and by encouraging our staff to publish their work and take part in the ASA. David was a tireless mentor to our staff and students and created an enduring relationship between ARL:UT and the UT faculty. A truly fitting legacy left for a University Affiliated Research Center.

9:05

4aPA4. David Blackstock and the Acoustical Society of America: Sixty-five years of commitment. Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnett Rd., Austin, TX 78758, misakson@arlut.utexas.edu) and Elaine Moran (Acoust. Society of America, Melville, NY)

David Blackstock and the Acoustical Society of America (ASA) have had a long history together. David attended his first ASA meeting in 1955 and became a member in 1959. Throughout his career, the ASA provided David with a place to publish and form collaborations. David more than repaid the ASA with service as the chair of meetings, technical and administrative committee service and as the Society's vice president and president. In 1993, he was awarded the Gold Medal, the ASA's highest honor. However, David's true service to the ASA is to the student and early career members of the Society. David coordinated the Students Meet Members for Lunch (SMMfL) initiative from 2002 to 2019. This program has paired numerous students with members of the Society. David's students from the University of Texas at Austin, the University of Rochester and Pennsylvania State University have been recognized by the Society with numerous awards and medals. In 2004, David was also recognized with the first ASA student council award for mentorship. In 2019, the ASA student council officially recognized David's commitment to students by renaming the mentorship award after him. Because of David's commitment to the Society and its future, his ASA legacy will never be forgotten.

9:25

4aPA5. David Blackstock and the University of Rochester. Diane Dalecki (Univ. of Rochester, 210 Goergen Hall, P.O. Box 270168, Rochester, NY 14610, dalecki@bme.rochester.edu)

This presentation provides perspective on David T. Blackstock at the University of Rochester in Rochester, NY. Dr. Blackstock's interactions with the University of Rochester began early in his academic career. In the early 1960's, he was an associate professor in the Department of Electrical Engineering at the University of Rochester for several years. Beginning in 1970, he moved to The University of Texas at Austin for his long academic career with the Applied Research Laboratories and the Department of Mechanical Engineering. Then in 1987, enabled through collaborations with Dr. Edwin L. Carstensen, Dr. Blackstock returned to Rochester to spend his summers at the University of Rochester. That same year, he became a visiting member of the Rochester Center for Biomedical Ultrasound directed by Dr. Carstensen. Dr. Blackstock continued to spend summers at the University of Rochester for several decades. Summers with Dr. Blackstock were filled with engaging scientific discussions, research collaborations, and interactions with many students and faculty in various departments. During the summer, he regularly taught his Fundamentals of Acoustics course and, during select years, his Nonlinear Acoustics course. Dr. Blackstock's longtime engagement with the University of Rochester greatly impacted the careers and lives of many students and faculty.

9:45–10:00 Break

10:00

4aPA6. Early works of David Blackstock in nonlinear acoustics: A view from the former Soviet Union. Lev Ostrovsky (Appl. Mathematics, Univ. of Colorado, 1111 Eng. Dr., ECOT 225, Boulder, CO 80309, lev.ostrovsky@gmail.com)

This presentation outlines some moments in the development of modern nonlinear acoustics in the 1960s–1980s, in which Professor David Blackstock played a significant role. In those times of limited contacts between the American and Soviet scientific communities, David was the first to notice our works in the area. He initiated and supported our contacts, first by regular mail (no Internet at that time!) and then, when it became possible, by inviting us to the USA. Our areas of common interest included, among others, radiation and diffraction of nonlinear acoustic beams, horn antennas, and parametric arrays. A few examples of parallel developments in these areas will be described.

10:20

4aPA7. David Blackstock and the physics of nonlinear sound beams. Thomas G. Muir (Appl. Res. Labs, Univ. of Texas, P.O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu) and Mark F. Hamilton (Appl. Res. Labs, Univ. of Texas, Austin, TX)

In the 1970s, David Blackstock and his students and colleagues examined nonlinear sound beams both theoretically and experimentally. Two noteworthy articles in JASA resulted from this research, one by Shooter, Muir and Blackstock (1974) and another by Lockwood, Muir and Blackstock (1973). Of particular interest was the generation of highly directive radiation at harmonics of the source frequency that at moderate source levels decreased in beamwidth with harmonic order. Additional sidelobes were also observed in the harmonics. The main lobe in both the fundamental and the harmonics eventually broadened with increasing source level as shock waves

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were formed. Eventually finite-amplitude absorption became significant, acoustic saturation set in, and the medium did not permit further increase in amplitude. David's application of weak-shock theory to model this process was validated by underwater acoustic experiments at the Applied Research Laboratories Lake Travis Test Station involving transducers driven at high intensities, at several hundred kilohertz, with waveforms and beams measured at ranges of several hundred meters. The results provided guidance for many applications, ranging from audio frequency tools in ocean acoustics to ultrasonic biomedical instruments now used worldwide in diagnostic and therapeutic applications. [Work supported by ARL:UT.]

10:40

4aPA8. David Blackstock and the parametric array in air. Joseph Pompei (Holosonics, 400 Pleasant St., Watertown, MA 02472, fjpompei@holosonics.com) and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

In 1971, Peter Westervelt and David Blackstock attended a Navy meeting on underwater acoustics. Following mention of rumored failed attempts to create a parametric array in air, Peter offered reasons why it should not be expected to work in air. David, sitting nearby, said to him "Bulls**t Peter!," to which Peter responded "Alright, you prove it!" After returning to Austin, David proceeded to do just that with master's student Mary Beth Bennett, whose experiments resulted in their 1975 JASA paper "Parametric array in air." In the 1990s, Joseph Pompei, then a master's student and former engineer at Bose, became interested in the airborne parametric array. His vision was to use the technology to make a practical directional loudspeaker and enable new ways to control sound. Facing similar skepticism from the audio industry, Pompei responded with a doctoral dissertation demonstrating the first high performance, low distortion parametric loudspeaker. The main technical challenges he overcame were developing correct signal processing methods and creating an ultrasonic transducer and amplifier with adequate bandwidth, power and fidelity. Now commercially successful as the "Audio Spotlight" by Holosonics, it has proved successful in digital signage, museums, retail displays, libraries, and similar places that require well localized sound.

11:00

4aPA9. Propagation of plane acoustic noise of finite amplitude. Frederick M. Pestorius (Appl. Res. Labs., Univ. of Texas at Austin, 8102 West Court, Austin, TX 7759, fmpestorius@gmail.com)

I was Professor Blackstock's first doctoral student at the University of Texas. I arrived at Texas having been away from serious academic work for six years while serving on submarines. David took me under his guidance and proposed an investigation of the propagation in air of plane finite-amplitude noise waveforms. The work was both theoretical and experimental, and the results obtained were valid both before and after shock formation. Starting with David's formulation of weak shock theory, we included Kirchhoff tube wall attenuation and dispersion because the experimental work was done in a pipe. The 96-foot pipe possessed an anechoic termination to permit the study of traveling waves. The theoretical model is unique in that it combines the effects of nonlinear distortion as well as tube wall attenuation and dispersion, and it is valid for input waveforms of arbitrary shape. The model was implemented in FORTRAN code and uses FFTs, relatively new at the time, to alternate between the time and frequency domains at each step along the propagation path. We initially tried to solve this problem analytically, but that effort failed in a sea of algebra. Outstanding agreement with the measured waveforms was obtained.

11:20

4aPA10. Reflecting on David Blackstock's contributions to understanding nonlinear propagation of jet noise. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu)

In 2002, my Ph.D. advisor, Vic Sparrow, and I began to research a problem in physical acoustics that had been relatively untouched for many years: modeling the nonlinear propagation of noise from high-speed jets. Without a steady progression of recent research to rely on, and flummoxed by seemingly nonphysical approaches to the problem by others, I found myself revisiting papers and reports authored by David Blackstock and his students in the 1970s: Pestorius, Anderson, Webster, and others. Within these reports, I found both critical insights and direction. In this talk, I will review David's foundational contributions to nonlinear jet noise propagation and some meaningful interactions.

11:40–12:00
Open Mic

Session 4aPP

Psychological and Physiological Acoustics: Open Source Audio Processing Challenge Results—Hackathon Challenge Presentations

Odile Clavier, Chair
Creare LLC, 16 Great Hollow Rd, Hanover, NH 03755

Chair's Introduction—8:20

Invited Papers

8:25

4aPP1. Speech envelope enhancement to improve cocktail-party listening. Daniel Cardosi (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, dcardosi@bu.edu), Lucas Baltzell, and Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

A novel speech enhancement scheme is being developed in our lab with the goal of improving speech intelligibility in "cocktail party" listening environments. By manipulating the temporal envelope to increase the salience of acoustic onsets, the algorithm improves access to binaural cues sampled at these onsets. The hope is that this will lead to a more robust spatial perception and improved streaming of competing talkers that could impact a variety of applications including hearing aids and cochlear implants. The algorithm is designed to run in real time, and in order to test its efficacy in real-world environments, it must be implemented on a portable device. For this project, the algorithm is being written in the Teensyduino software and uploaded to a Tympan Rev E DSP unit fitted with a Tympan AIC CODEC daughterboard. External speech is both captured by and played back through a pair of Tympan BTE earpieces, and all processing is performed independently for each ear by the Tympan's Teensy 4.1 microcontroller. Assuming that the algorithm is successfully implemented, speech signals delivered to the listener will have perceptibly higher crest factors (i.e., higher transient energy) than at input. The portable, bilateral, real-time implementation developed here will allow us to evaluate any associated binaural effects and their consequences in complex real-world listening tasks.

8:40

4aPP2. Influence of number of hearing aid compression channels on spatial release from masking. Marc Brennan (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, 4075 East Campus Loop South, Lincoln, NE 68583, Marc.Brennan@unl.edu) and Ava Feller (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, Lincoln, NE)

Listeners with NH take advantage of differences in the spectrum of speech and noise, dips in the noise level, and spatial separation between sound sources to improve their understanding of speech. Listeners with SNHL are less able to take advantage of these differences. Possibly by not considering the influence of access to spectral information under conditions of amplification, hearing-aids only provide marginal improvement in complex listening environments. This experiment examined whether modifying the number of compression channels can improve spatial release from masking (SRM). We obtained sentence recognition for AzBio sentences presented from in front of each participant. Noise that varied temporally and spectrally over time was presented in a collocated condition and in a spatially separated condition. Test conditions included unaided and binaural amplification with an open-source hearing aid. Four and 16 channels of compression were tested. These experimental conditions were based on prior data that documented improved measures of spectral resolution with 16 relative to 4 channels of compression. We hypothesized that SRM would be greater for the condition with 16 channels of compression, relative to the condition with 4 channels of compression. SRM was hypothesized to be poorest for the unaided condition.

8:55

4aPP3. Measuring hearing aid compression algorithm preference with the Tympan. Jennifer Lentz (Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu), Donghyeon Yun (Indiana Univ., Bloomington, IN), and Stuart Smiley (5am Solutions, Bloomington, IN)

In this Tympan challenge, we aim to evaluate listeners' preferences for different Wide Dynamic Range Compression (WDRC) settings in a real-world environment. While WDRC is used to provide comfort for high-level sounds and audibility for low-level sounds, the nonlinear algorithm distorts the stimulus envelope and can alter the input signal-to-noise ratio (SNR). To date, most measurements of speech understanding and preference under different WDRC parameters (e.g., compression speed and compression ratio) have used pre-processed stimuli presented in a sound-attenuating room. In this study, we plan to measure listeners' preference for different algorithm settings by using the Tympan in real-world environments. The Tympan will be programmed based on the listeners' hearing loss, but the user will be able to switch between algorithm settings immediately in their environment using the associated Smartphone app. Using ecological momentary assessment, we will briefly survey the users about their perceptions as they switch between settings. We

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also will use the Tympan to save acoustic samples and to measure the acoustic properties of the environment and the hearing output, in an effort to determine the drivers of the preference scores [i.e., amount of distortion, degree of audibility, SNR changes (if possible), etc.].

9:10

4aPP4. Directionality characteristics of the Tympan open-source hearing aid and earpieces. Joshua M. Alexander (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Directional systems in hearing aids are critical for improving speech understanding in the presence of background noise and competing talkers. Tympan is an open-source research platform for speech processing and hearing improvement. The latest Tympan version (Rev E) includes earpieces with a front and a rear microphone, allowing researchers to explore the effects of different directional settings using a realistic hearing aid configuration. The directionality characteristics of the Tympan earpieces were evaluated using a KEMAR in a semi-reverberant sound field. To demonstrate the functionality of Tympan's directionality features, different software settings were manipulated, including the relative delays and gains between the two microphones. Results will be discussed along with considerations for more advanced directional solutions, such as automatic directional switching, adaptive directivity patterns, binaural signal processing, and beamforming.

9:25

4aPP5. Immersive multitalker remote microphone system. Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 456 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801, corey1@illinois.edu) and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Hearing aids and other listening devices perform poorly in noisy, reverberant environments with many overlapping sound sources. Remote microphones, which are worn by a talker and transmit low-noise, low-reverberation speech directly to a hearing device, can improve intelligibility even in adverse environments. However, most commercial remote microphone accessories can only be used with one talker at a time and remove the interaural cues that listeners use to localize sound sources. Using the Tympan open-source hardware platform, we demonstrate a multitalker remote microphone system that preserves interaural cues and room acoustic effects. A pair of wireless microphones transmits sound from two talkers to the Tympan system. A set of real-time adaptive filters running on the embedded processor match the magnitude and phase of the remote microphone signals to the sound received by the earpiece microphones. The earpiece output has the signal-to-noise ratio of the remote microphones but the spatial cues of the on-ear microphones. Because the remote microphones provide low-noise reference signals, the system can track motion in real time and does not need to perform explicit source localization or separation.

9:40–9:55 Break

9:55

4aPP6. A general-purpose pipeline to interface the Tympan hardware with an external computer. Danielle Benesch (Ctr. for Interdisciplinary Res. in Music Media and Technol., McGill Univ., Danielle Benesch c/o Jérémie Voix, École de Technologie Supérieure, 1100 Notre-Dame St. W, Montreal, QC H3C 1K3, Canada, dbenesch@critias.ca), Kocherla Nithin Raj (NSERC-EERS Industrial Res. Chair in In-Ear Technologies, École de Technologie Supérieure, Bangalore, Karnataka, India), and Jeremie Voix (NSERC-EERS Industrial Res. Chair in In-Ear Technologies, École de technologie supérieure, Montréal, QC, Canada)

Some audio applications require resource-heavy algorithms which cannot be run on real-time digital signal processors such as the Tympan hardware directly due to memory and processing constraints. These algorithms can, however, run on an external computer (PC), and their outcomes can be relayed back to the Tympan where necessary adjustments can be made in the audio processing elements of the underlying hardware. The proposed pipeline includes a 4-channel audio input, the algorithm running on the PC, the Tympan hardware, and a 4-channel output. The Tympan hardware acquires input audio from the Tympan's onboard and/or external microphones with the I2S protocol. The input data are transmitted to the algorithm running on the external PC via USB/Serial communication by exposing the Tympan as a soundcard interface. The output of the PC algorithm is then sent back to the Tympan, through serial communication, so that the underlying audio processing parameters are adjusted. A possible use case is also discussed: a detection algorithm runs on the PC and serial commands are sent back to Tympan to alter the audio outputs, e.g., by playing masking noise. The pipeline is built such that each element can be used independently, abstracting the interconnection of the pipeline elements.

10:10

4aPP7. Abstract withdrawn.

10:25

4aPP8. Open-source baby monitor. Shane W. Lani (Johns Hopkins Univ., 1454 Catherine St., Decatur, GA 30030, shane.w.lani@gmail.com), Jennifer Cooper, Tyler J. Flynn (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD), Jordan Schleif (DoD, Laurel, MD), Adaleena Mookerjee, and O. H. Ott-Pietrak (Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD)

New parents often want to know everything that their new little one is up to, even while they are sleeping. The easiest way to accomplish this task is to install an audio baby monitor, comprised of a baby monitor receiver unit near the child's bed and a parental unit which can be carried about the house. These monitors can alert the parents when their bundles of joy are quite upset, but they can also alert the parents with annoying audio from white noise sound machines, lullaby tunes, as well as transient passing motorcycles and landing airplanes. Too much unwanted noise transmitted to the parental unit could lead to the parent turning down their receiver volume and later missing the true alert of their crying little one. The ideal baby audio monitor alerts the parental unit via real time audio only when the baby needs attention, and does not transmit or amplify the received audio for all other noise sources. This presentation outlines the

efforts undertaken to develop and test a smart baby monitor using an open-source audio platform and includes performance comparisons of the developed system to commercially available baby monitors.

10:40

4aPP9. Open source audio platform ultrasound dosimeter. Jennifer Cooper (Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723, jennifer.cooper@jhuapl.edu), Tyler J. Flynn, O. H. Ott-Pietrak, Adaleena Mookerjee (Johns Hopkins Appl. Phys. Lab., Laurel, MD), Jordan Schleif (DOD, Laurel, MD), and Shane W. Lani (Johns Hopkins Appl. Phys. Lab., Decatur, GA)

It is well known that hearing loss can occur from exposure to high intensities of sounds >80 dBA; however there is less information about hearing loss from ultrasonic frequencies. There are many commercial and industrial devices that produce ultrasonic acoustic energy ranging from humidifiers, pest and pet repellent devices, and crowd control devices. Even though there are fewer regulations with regard to ultrasound, there have been documented cases of effects including temporary or permanent threshold shifts from exposure to ultrasonic energy. However, there is not a readily available open source solution for quantifying exposure. In this work, a TYMPAN open source hearing aid development platform is used to develop and test a prototype ultrasonic dosimeter. This dosimeter works like a SPL meter, with the inclusion of the ultrasonic 1/3 octave bands, with an audible alarm emitted when the user happens to be in the presence of ultrasonic energy. From the full data recording of a test covering various environments, several key features of the exposure are highlighted. The presentation will cover the development of the algorithms, testing of the device, and post test analysis. Code will be made available via the open source libraries for future users to build upon.

10:55

4aPP10. Spatial acoustic processing with a laser distance sensor using a Tympan device. Kanad Sarkar (Univ. of Illinois at Urbana Champaign, B10 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801-2447, kanads2@illinois.edu), Ryan M. Corey, and Andrew C. Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

We will produce a setup for source localization experiments utilizing laser distance sensors with the Tympan device. The Tympan will allow us to perform on-device post processing, improve the portability of our experimental setup, and potentially reduce the electrical noise in our data. We demonstrate the benefits of this device through localizing an acoustic source through lasers placed at different points in space and comparing the Tympan setup to our previous setup.

11:10–11:40
Panel Discussion

Session 4aSA

Structural Acoustics and Vibration and Engineering Acoustics: Historical Development of Fuzzy Structures Concepts

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

David Feit, Cochair
None, 4601 N. Park Ave., Apt. 213, Chevy Chase, MD 20815

Chair's Introduction—8:00

Invited Papers

8:05

4aSA1. Fuzzy structure theory, statistical energy analysis, and resonant-frequency-distribution of internal mass. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

The present talk gives a historical perspective on two early and relatively little- known papers having to do with fuzzy structure theory. The first is a letter to the editor by Lyon titled "Statistical energy analysis and structural fuzzy" (*JASA* 1995). The second paper is a slightly later 1997 Journal of Vibrations and Acoustics (ASME) article titled "Resonant-frequency-distribution of internal mass inferred from mechanical impedance matrices, with application to fuzzy structure theory." The present paper seeks to clarify and qualify Lyon's statement, "It appears hat the fuzzy structure theory of structural interaction is highly compatible with the SEA framework." The second paper clarifies the concept of modal mass and gives a proof that, for a lightly damped closed mechanical system, you can express the mechanical displacement fiield as a sum of natural modes. With each natural mode, one can associate a fixed amount of mass. The mass, moreover. is additive and sums to the total mass of the system. The idea goes back to some earlier work (1963) of Cunniff and O'Hara of the U. S. Naval Laboratory. The theory leads to the concept of modal mass per unit natural frequency bandwidth, and leads to a justification for the SEA assumption that interactions are between motions of natural modes with comparable natural frequencies.

8:30

4aSA2. Asymptotic behavior of the added damping effect of a fuzzy substructure and the importance of modal overlap. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu) and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The theory of fuzzy structures was introduced by Christian Soize in 1986 as a means of predicting the average vibroacoustic behavior of complex structures with imprecisely known internals. The damping and mass loading effects of a fuzzy internal substructure, modeled by random 1-dof oscillators, are analyzed through a boundary impedance. The Pierce-Sparrow-Russell and Strasberg-Feit models of fuzzy structures simplify the mathematics involved and concentrate on the physical concepts behind the added damping and mass loading effects of the internal substructure. Both models contain a problematic step in which a discontinuous impedance sum is replaced with a smooth integral. This paper explores the asymptotic behavior of the impedance sum in the limits of a large number N of attachments and small attachment damping ζ in order to determine the viability of this step. Modal overlap plays a crucial role, since half power points of individual attachment resonances overlap for small ζ when the density of attachments is high enough or for fewer attachments when the damping of individual attachments is high enough. It is found that the impedance sum sufficiently approaches the integral limit when $N\zeta \geqq 2$. [Work originally sponsored by ONR.]

8:55

4aSA3. Damping and dissipation in a fuzzy structure. Adnan Akay (Bilkent Univ., Bilkent University, Ankara 06800, Turkey, akay@cmu.edu) and Antonio Carcaterra (Theor. and Appl. Mech., Sapienza Univ. of Rome, Rome, Italy)

Damping describes processes that lead systems to reach equilibrium during which energy either escapes from the system or is redistributed within it. In both cases, the kinetic energy cascades to shorter spatial and temporal dimensions, eventually reaching a state of thermalization. Radiative damping due to energy escape from a system is generally studied using the system-environment paradigm. This approach also explains how a structural fuzzy that refers to ancillary substructures attached to a primary structure leads to apparent damping. It is known that an infinite number of lossless harmonic oscillators with continuously distributed frequencies yield a perfect loss mechanism, since energy has infinitely many paths to travel within the complex structure. For a more realistic structure with a finite

number of oscillators, however, energy may return to the primary, unless its frequencies follow a certain distribution (Carcaterra and Akay, JASA). This presentation will focus on the loss mechanism of a fuzzy structure and compare it with dissipation mechanisms at the molecular level.

9:20

4aSA4. Energy flow, connectivity and entropy in the light of fuzzy structures. Antonio Carcaterra (Mech. and Aerosp. Eng., Sapienza Univ., Rome 00184, Italy, antonio.carcaterra@uniroma1.it) and Adnan Akay (Bilkent Univ., Ankara, Turkey)

Fuzzy structures, as presented in the literature, refer to a dynamic coupling between a master and a population of slave resonators of uncertain characteristics. However, the investigation of such systems opened new perspectives looking at the way energy flows between the two components, the masters and the slaves. New elements, not originally included into the concept of fuzzy, have been subject of investigations (Carcaterra-Akay, JASA, JSV) disclosing relevant questions: how the topology of connectivity between master and slaves affects the energy transfer? Can be a thermodynamic analogy established between fuzzy structures and thermal systems? Can entropy gain a role in structural dynamics, as it plays a key role in thermodynamics? This paper makes a historical reconstruction of these puzzling questions related to fuzzy structures.

9:45–10:00 Break

10:00

4aSA5. Meta-mass systems: A harbinger for a vibration energy harvesting technology. John J. McCoy (The Catholic Univ. of America, 1922 New Hampshire Ave., Washington, DC 20009, mccoy@cua.edu)

For a limited class of fuzzy structures, the "master" structure is a grounded Newtonian mass wherein are distributed the "fuzzies," modeled as nearly equal magnitude sprung masses with resonances densely filling a frequency band. We term said system a grounded "meta-mass," showing a properly designed "meta-mass vibration energy harvester" addresses a fundamental technical obstacle to the harvesting of waste mechanical energy, the so-called "aperture" problem; thereby making meta-mass systems a harbinger for a vibration energy harvesting technology. Shown is the oscillators' resonances provide"bounds'' on the system resonances, the relative "locations" of the system resonances within its bound-pair being determinative of system behavior. Then shown is the value of a certain combination of non-dimensional system measures—twice the fractional bandwidth squared divided by the product of the mass ratio and the natural log of N—is determinative of said relative locations. Finally shown is virtually all the energy inputted impulsively to the Newtonian mass is transferred to the encapsulated oscillators in a time of the order of the inverse of the band of oscillators, for a narrow system subclass wherefore each system resonance is near the center of its bound-pair.

Contributed Paper

10:25

4aSA6. Experimental validation of a switchable oscillator array. Seth Hubbard (NSWC Carderock, 9500 MacArthur Blvd, Bldg 15 Rm. 124A, W. Bethesda, MD 20817-5700, seth.hubbard@navy.mil), Joseph Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), John Sterling (NSWC Carderock, Bethesda, MD), and Sarthak Regmi (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This presentation will discuss an experimental validation of a switchable oscillator array that can be used to absorb energy from a resonant system. The individual elements of the oscillator array are composed of thin piezocomponents that vibrate as cantilevers. Each piezocomponent is designed to have a fundamental frequency, independent of the other elements, forming a band around the primary resonance of the host structure. The system is switched by electrically shunting the individual piezocomponents, lowering the stiffness of the cantilever, which has a significant effect on the energy transfer into the oscillator array and, hence, its ability to damp the motion of the host structure. Modeling in MATLAB and COMSOL is used in the design to predict the performance of the specified system used in the experiment, as well as larger systems with more than one resonance.

Invited Papers

Session 4aSC

Speech Communication: Speech Communication in Challenging Situations (Poster Session)

Matthew Masapollo, Chair
University of Florida, 1225 Center Dr., Gainesville, FL 32610

All posters will be on display from 9:00 a.m. to 12:00 noonn. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

4aSC1. Impaired phonetic learning in Parkinson's Disease. Christopher C. Heffner (Communicative Disord. and Sci., Univ. at Buffalo, 122 Cary Hall, South Campus, Buffalo, NY 14214, ccheffne@buffalo.edu) and Emily B. Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs, CT)

Parkinson's Disease (PD) is a neurodegenerative disorder primarily associated with its motor consequences. However, it also has important implications for cognitive functions, including speech production (Altmann and Troche, 2011), the perception of speech intensity (Richardson and Sussman, 2019), and visual category learning (Shohamy *et al.*, 2008), due to its effects on subcortical brain systems that rely on dopamine. In the present study, we examine the effects of PD on a variety of speech perception tasks: one of non-native phonetic learning, one of learning to comprehend rate-compressed speech, one of learning to understand accented speech, and one probing the use of speech rate to guide word segmentation. People with PD showed significant deficits when compared to young and age-matched controls in their ability to adapt to rate-compressed speech. Within the population of those with PD, participants using levodopa-based medication performed significantly worse on the non-native learning task than a small subset of participants without such medication. This suggests that changes in dopamine regulation and use associated with PD have significant effects on higher-level aspects of speech perception such as speech learning.

4aSC2. Tongue position variability in sustained vowels produced by healthy children and young adults with and without a semi-occluded vocal tract: A pilot study. Steven M. Lulich (Speech, Lang. & Hearing Sci., Indiana Univ., Bloomington, IN, slulich@indiana.edu), Megan Diekhoff, and Rita R. Patel (Speech, Lang. & Hearing Sci., Indiana Univ., Bloomington, IN)

Speech development is a process by which talkers learn, among other things, to coordinate neuromuscular activations of motor units spread across several muscles that effect articulator positioning and movement. This paper presents results from an analysis of tongue position variability during sustained vowel production. Data from two children (7 years 11 months, and 9 years 7 months) and two young adults (both 20 years 7 months) are presented. Articulatory data were recorded with a 3D/4D ultrasound system, in two conditions: without a semi-occluded vocal tract ("without straw") and with a semi-occluded vocal tract ("with straw"). Statistical analyses of tongue position variability revealed a main effect of age, with children showing greater variability than adults. There was no statistically significant main effect of condition, but the main effect of age was driven primarily by the "with straw" condition. We speculate that tongue position variability during sustained vowels may be an index of speech motor control maturity, and that the interaction between age and condition suggests children are less able than adults to rely on learned feed-forward motor programs under conditions of somatosensory perturbation, such as the presence of a straw between the lips.

4aSC3. An acoustic study of vowels produced by Cantonese alaryngeal speakers using clear speech. Steven R. Cox (Commun. Sci. and Disord., Adelphi Univ., Hy Weinberg Ctr. Rm. 136, Garden City, NY 11530, scox@adelphi.edu), Ting Huang (Graduate Inst. of Linguist., National Tsing Hua Univ., Hsinchu, Taiwan), Wei-rong Chen (Haskins Labs., New Haven, CT), and Manwa L. Ng (Speech Sci. Lab., Univ. of Hong Kong, Pokfulam, Hong Kong)

Alaryngeal speech is an alternative method of verbal communication following the removal of the larynx. Common alaryngeal speaking methods include esophageal (ES) speech, tracheoesophageal (TE) speech, and speech produced using an electrolarynx (EL). A recent study [Hui *et al.*, in press, Folia Phoniatr Logop] demonstrated better intelligibility in Cantonese alaryngeal speakers while using clear speech (CS) compared to their everyday 'habitual speech' (HS). Yet the correlated change in the acoustics associated with CS is still not clear. The purpose of this study was to assess the acoustic characteristics of vowels produced by Cantonese alaryngeal speakers during HS and CS conditions. Thirty-one alaryngeal speakers (9 EL users, 10 ES speakers, and 12 TE speakers) produced sentences from the Cantonese Sentence Intelligibility Test in HS and CS. Acoustic analyses considered formants (measured by reassigned spectrogram), vowel space area (VSA) and vowel duration across alaryngeal speaking methods and styles. We also examined these acoustic differences in relation to duration of alaryngeal speech use and intelligibility. Results suggest that EL users had larger VSAs and larger increases in vowel duration while using CS compared to ES and TE speakers. Future research will examine the effects of refined CS protocols on improving acoustic and perceptual characteristics of Cantonese alaryngeal speech.

4aSC4. Effects of background noise on the speech acoustics of people with aphasia. Kirsten Dixon (Commun. Disord., Brigham Young Univ., Provo, UT), Christopher Dromey (Commun. Disord., Brigham Young Univ., 133 TLRB, BYU, Provo, UT 84602, dromey@byu.edu), Tyson Harmon (Commun. Disord., Brigham Young Univ., Provo, UT), and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

We investigated the effect of six background noise conditions (silent baseline, pink noise, monologue, lively conversation, one-sided phone call, and cocktail noise) on acoustic measures of speech during story retells in people with aphasia. Eleven individuals with aphasia and 11 age- and gender-matched control participants heard the background noise through headphones while they retold short stories. We measured mean and standard deviation of intensity and fundamental frequency (F_0), and speech rate. A Matlab application computed a speaking time ratio measure (i.e., time

speaking versus time pausing). With the exception of the monologue and one-sided phone call condition, both people with aphasia and control participants significantly increased their intensity and F_0 in the presence of background noise, regardless of noise type. Additionally, participants with aphasia had significantly lower speaking time ratios and speaking rates than control participants. All speakers made acoustic changes while hearing background noise, likely because speech intensity rises in an effort to increase the signal-to-noise ratio, while mean F_0 increases due to a presumed rise in subglottal pressure. Further research is suggested to investigate other acoustic differences, possibly at the segmental level, between speech produced in informational and energetic background noise conditions.

4aSC5. Speech characteristics of children with ASD to a humanoid robot and peers. Kaelin Kinney (Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40242, kmkinn03@louisville.edu), Maria V. Kondaurova (Psychol. & Brain Sci., Univ. of Louisville, Louisville, KY), Karla C. Welch (Elec. & Comput. Eng., Univ. of Louisville, Louisville, KY), Grace M. Kuravackel (Norton Children's Autism Ctr., Louisville, KY), Robert Pennington (Special Education and Child Development, Univ. of North Carolina Charlotte, Charlotte, NC), Gregory Barnes (Norton Children's Autism Ctr., Louisville, KY), and Dan Popa (Elec. & Comput. Eng., Univ. of Louisville, Louisville, KY)

Children with Autism Spectrum Disorder (ASD) display difficulty engaging in social interactions. How does the use of a NAO robot during a social skills intervention affect the speech of children with ASD? This study examined whether the use of a NAO robot during a social skills intervention affects the quantity and quality of speech in six children (mean age = 11.6 y, range = 10.5–12.6) with ASD who attended four weekly social skills intervention sessions. The speech quantity was defined as an overall speech rate (utterances per minute), initiation rate (utterances per minute addressing the robot/other child), and response rate (utterances per minute responding to the robot/other child). Mean length of utterance (MLU) in overall speech, initiations, and responses was also measured. A marginally higher initiation rate and overall MLU in speech to the robot were identified in later sessions. Children produced a significantly higher MLU in responses to the robot in later sessions. MLU for initiations were marginally higher than MLU for responses to the robot. No changes in speech rate or MLU in speech directed to other children were identified. This suggests children may require repeated exposure to the NAO robot before changes in speech characteristics are revealed.

4aSC6. Multitalker speech perception in neurodiverse populations. Katherine A. Emmons (University of Washington Autism Ctr., Univ. of Washington, 1701 NE Columbia Rd., Seattle, WA 98195, kemmons2@uw.edu), Annette Estes (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Stephen Dager (Dept. of Radiology, Univ. of Washington, Seattle, WA), Ross K. Maddox (Univ. of Rochester, Rochester, NY), Susan Astley Hemingway (Ctr. on Human Development and Disability, Univ. of Washington, Seattle, WA), John C. Thorne, Adrian KC Lee (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and Bonnie K. Lau (Dept. of Otolaryngol.—Head and Neck Surgery, Univ. of Washington, Seattle, WA)

The ability to selectively attend to one talker in the presence of competing talkers is a crucial skill employed in everyday life. In this study, multitalker speech perception thresholds were measured in three groups; Autism Spectrum Disorder (ASD), Fetal Alcohol Syndrome Disorder (FASD), and an age- and sex-matched typically functioning (TF) group. Participants listened to three simultaneous sentences from the Coordinate Response Measure corpus: the target stream to be attended (0° azimuth) and two spatially separated ($\pm 45^\circ$ azimuth) masker streams. Participants were asked to identify the color and number associated with the callsign "Charlie." Target-to-masker ratios (TMRs) were estimated based on the average of four runs in which the target was fixed at 40 dB SPL and maskers were adaptively varied using a one-up-one-down procedure to estimate 50% correct. The target speaker was always male; the two maskers were either male/male or female/female. Overall, TMR thresholds were higher in both ASD and FASD groups than the TF group. Additionally, a negative correlation between intellectual ability and TMR thresholds was observed. These preliminary results suggest intellectual ability may impact how well listener's perceive speech in multitalker situations, especially in neurodiverse populations.

4aSC7. Lexical effects on the perceived clarity of noise-vocoded speech in younger and older listeners. Terrin N. Tamati (Otolaryngol., The Ohio State Univ., 1608 Aschinger Blvd., Columbus, OH 43212, terrintamati@gmail.com) and Aaron C. Moberly (Otolaryngol., The Ohio State Univ., Columbus, OH)

When listening to degraded speech, such as speech delivered by a cochlear implant (CI), listeners can make use of linguistic knowledge to facilitate speech recognition. Yet, the extent to which linguistic knowledge can be effectively used to compensate for degraded input may depend of the level of degradation and the age of the listener. The current study investigated lexical effects in the compensation for degraded speech via noise-vocoder simulations of CI hearing in younger and older normal-hearing listeners. In two online experiments, listeners rated the clarity of noise-vocoded sentences, with varying filter slopes. Lexical information was provided in the form of text priming and/or the lexical content (i.e., lexical frequency and neighborhood density) of key words in the sentences. Results showed that lexical information from both text priming and the lexical content enhanced the perceived clarity of degraded speech, but the strength of the lexical effects depended on the level of degradation. Additionally, lexical information appeared to equally benefit younger and older listeners. Findings demonstrate that lexical knowledge may be used in cognitive compensation during the recognition of noise-vocoded speech in both younger and older listeners. However, these mechanisms may not be as effective when the signal is highly degraded.

4aSC8. Talker variability in Mandarin spoken word recognition: Comparing normal and hearing-impaired listeners. Yu Zhang (Commun. Sci. and Disord., Oklahoma State Univ., 019 Murray Hall, Stillwater, OK 74078, yu.zhang10@okstate.edu), Suju Wang, and Yingying Shang (Peking Union Medical College Hospital (PUMCH), Beijing, China)

During natural speech processing, talker variability is inherently embedded in the acoustic signal and may affect the speed and accuracy of processing. Prior studies have demonstrated the effect of talker variability in normal listeners. What is the effect of inter-talker variability in people with hearing impairment? This study aims to explore the question in native speakers of Mandarin Chinese, trying to compare listeners with and without hearing impairment. Participants are instructed to listen to pre-recorded pairs of Mandarin Chinese words (e.g., in English translation, *happy – sad*) and make responses only to the second item of each pair (e.g., *sad*) in a short-term priming paradigm. The speed and accuracy of the responses are measured. The fact that the pairs were spoken by either the same or different native speakers of Mandarin Chinese allows us to examine the effect of speaker voice change on the reaction time and accuracy. The hearing-impaired listeners appear to be facilitated more by the same speaker voice condition in terms of response accuracy, compared with the listeners without hearing impairment. There also seems to be more semantic facilitation to the hearing-impaired in terms of reaction time, although this effect does not seem to be modulated by talker variability.

4aSC9. Electromagnetic articulography is feasible for assessment of speech motor skills in cochlear implant users. Matthew Masapollo (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Gainesville, FL 32610, mmasapollo@phhp.ufl.edu), Yonghee Oh, Jessica Goel, Joanna Lowenstein, and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Auditory input is essential to the acquisition of fluent speech production, and yet speech production has been scarcely studied in congenitally deaf individuals since cochlear implants (CIs) became a standard treatment. In particular, methods of directly observing speech movements have not been applied, such as electromagnetic articulography (EMA). Several EMA studies have been done with adults with CIs, but those studies used older CI and EMA technology and included subjects with adult-onset hearing-loss, who would have developed speech motor control and phonological

representations before losing their hearing. We here investigated whether current EMA technology can be reliably used with CIs by testing (1) whether CIs affect the recordings obtained from EMA, and (2) whether EMA interferes with the signal processing of CIs. In an initial experiment, we calibrated EMA sensors with and without a CI present in the EMA field, and measured impedances of all CI electrodes when in and out of the EMA field. In a subsequent experiment, speech movements were recorded for a normal-hearing talker with and without a CI present in the EMA field. Results indicate that there is no cross-interference between current EMA and CIs, and that EMA data can reliably be obtained from CI users.

4aSC10. Telepractice effects: Vowel space characteristics in deaf and hard-of-hearing pediatric patient, provider and caregiver speech. Maria V. Kondaurova (Psychol. & Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40242, maria.kondaurova@louisville.edu), Cheryl W. Donaldson (Heuser Hearing Inst. & Lang. Acad., Louisville, KY), Qi Zheng (Biostatistics, Univ. of Louisville, Louisville, KY), and Elizabeth Pike (Psychol. & Brain Sci., Univ. of Louisville, Louisville, KY)

When talkers are aware of speech perception difficulty on the part of the listener due to adverse conditions, they will adopt a "clear speech" style that provides the listener with more salient acoustic cues to access and comprehend the message. Does the use of telepractice affect the vowel articulation in deaf and hard-of-hearing (DHH) pediatric patients, providers and caregivers? This study examined the phonetic modification of the vowel space area in speech of DHH children with cochlear implants (CI) ($n = 7$, mean age 4 years 11 months), their caregivers and a speech-language pathologist (SLP) during one 30-min in-person and one sequential tele-session, order counter-balanced. The first and second formants of vowels /i/, /a/ and /u/ in child, provider and caregiver productions were measured and vowel space area was calculated. The preliminary analysis including four child-mother dyads demonstrated a significantly larger vowel space area in speech of DHH children and the SLP in the tele- compared to the in-person session. No significant difference in the caregiver vowel space area was identified. The results suggest that DHH children and the clinical provider adjust their vowel articulation in order to accommodate the listener during tele- compared to in-person speech-language intervention.

4aSC11. A robust tongue shape model from ultrasound recordings of normal and impaired speech. Sahba Changizi (Elec. Eng., Ecole de Technologie Superieure, 2 Chemin Radcliffe, Montreal, QC H4X1B9, Canada, sahba.ch@gmail.com), Catherine Laporte (Elec. Eng., Ecole de Technologie Superieure, Montreal, QC, Canada), and Lucie Ménard (Linguist, UQAM, Montréal, QC, Canada)

Ultrasound imaging is a helpful tool to observe tongue movements without interfering with natural speech. There exist a variety of models to quantify tongue shape based on contours extracted from ultrasound images. However, these can be affected by poor image quality, e.g., when parts of the tongue are missing from the images due to imaging artifacts. In this study, we investigate the effects of various contour extraction errors on the accuracy and consistency of different shape measures. We developed exponential and polynomial contour perturbation models, then simulated missing tongue tip and root, and investigated the impact of these perturbations on shape measures based on the discrete Fourier transform (DFT), modified curvature index (MCI), and triangular fitting. This was applied to a set of CV utterances collected from healthy and impaired speakers. Results demonstrate the effectiveness of DFT and triangular fitting in clustering different phonemes despite the added noise. A high degree of correlation was found between the DFT coefficients of the perturbed and original tongue contours. There is also a trade-off between the robustness of the model and sensitivity to minor actual differences in tongue shape. Sometimes, these slight differences help group tongue shapes that differ, e.g., due to coarticulation effects. Therefore, we have attempted to improve the precision of the DFT model by adding palatal contact information.

4aSC12. Role of training task on learning and maintenance of noise-vocoded speech. Julia R. Drouin (California State Univ. Fullerton, 2600 E Nutwood Ave., Ste. 285, Fullerton, CA 92831, judrouin@fullerton.edu) and Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Listeners who utilize cochlear implants demonstrate variability in speech recognition outcomes. Experience with the auditory signal via structured auditory training has been shown to improve speech perception. A primary goal of aural rehabilitation is optimizing training paradigms to maximize speech understanding gains observed following training. The current study examined the role of the training task on adaptation to noise-vocoded speech signals as a foundational step towards clinical recommendations for cochlear implant users. Three groups of listeners were assigned to one of three training tasks: transcription with feedback, transcription without feedback, or talker identification. All listeners completed transcription test phases before training, immediately after training, and one-week following training. At each test, keyword accuracy was measured at the sentence-initial and sentence-final position for high and low-predictability sentences. Immediately following training, listeners showed improved transcription for both sentence positions, which was mediated by sentence predictability, and learning was comparable across training tasks, in line with previous findings. Critically, gains were also maintained equivalently among training groups one week later. Our results converge with growing findings pointing towards utilizing non-traditional training tasks to maximize perceptual learning of noise-vocoded speech and point towards rehabilitation considerations for cochlear implant users.

4aSC13. Variability in the production of /s/ by adults who do and do not stutter. Xiaofan Lei (Speech Lang. and Hearing Sci., Univ. of Minnesota—Twin Cities, 115 Shevlin, Minneapolis, MN 55108, leixx147@umn.edu) and Benjamin Munson (Speech Lang. and Hearing Sci., Univ. of Minnesota—Twin Cities, Minneapolis, MN)

Speech produced by typically developing adults at their habitual speech rate is characterized by low utterance-to-utterance variability (Smith *et al.*, 1995). Wiltshire *et al.* (2021) posited that higher production variability underlies or causes stutter-like disfluencies among individuals who stutter. Given the hypothesis, higher variability should be observed among individuals who stutter at all linguistic units. Adults who stutter (AWS) have been found to have higher articulatory variability when producing sentences and nonwords relative to controls (Kleinow and Smith, 2000; Smith *et al.*, 2010). However, no studies have investigated the group difference at the level of individual phones, such as when producing the fricative /s/ (Munson, 2001; 2004). The current study attempts to fill this gap of knowledge by comparing the acoustic variability of /s/ between matched pairs of AWS and fluent controls ($n = 18$). We found that /s/ was produced with greater variability in nonwords relative to words ($\beta = 0.48$, $F[1,617] = 23$, $p < 0.001$, 95% CI [0.29, 0.68]). However, AWS did not show higher acoustic variability relative to their fluent peers ($\beta = 0.16$, $F[1,16] = 0.09$, $p > 0.1$, 95% CI [-0.94, 1.26]). Results are not consistent the hypothesis that the speech of AWS is always more variable than fluent controls. Implications will be further discussed.

4aSC14. Individual differences in speech and language within 72 h after a concussion. Caryn Grabowski, Jisook Ahn (Dept. of Speech-Lang. Pathol., Seton Hall Univ., Nutley, NJ), Anthony J. Testa (Sports Medicine, Seton Hall Univ., South Orange, NJ), Sarah Ackermann (none, Zurich, Switzerland), Emily Hernandez, Austin Terlecky, Matthew Grunstein, Ava Silverman (Dept. of Speech-Lang. Pathol., Seton Hall Univ., Nutley, NJ), Gabe Scher, Deja Craig, Kaitlin Kelly, Mercedes Cunningham (Sports Medicine, Seton Hall Univ., South Orange, NJ), and Sona Patel (Dept. of Speech-Lang. Pathol., Seton Hall Univ., 123 Metro Blvd Interprofessional Health Sci. Campus, Nutley, NJ 07110, sona.patel@shu.edu)

Traumatic brain injuries (TBIs) cause immediate and temporary alterations in brain function that result in a variety of symptoms including impacts on speech. Symptoms of dysarthria and aphasia occur but are primarily

reported in moderate-severe injuries. Less is known about the specific changes in acoustic parameters and language that occur in mild concussions. A total of 250 athletes at Seton Hall University participated in baseline speech testing. Speech testing was performed again in a subset of 15 athletes who sustained a mild concussion (within 72 h). Speech and language analysis was performed on samples derived from a picture description task. Statistical comparisons showed minimal changes in acoustic parameters on average, but individual differences were present. Specifically, some individuals presented with a flatter prosody and/or lower mean pitch than others post-injury compared to baseline. Although no clear pattern of acoustic changes emerged, language analysis (semantics, syntax, morphology) showed an overall reduction of complexity of productions. Language outcomes indicate that changes in speech symptoms occur even in mild cases. Further investigations should examine the speech and language changes after a head injury against the individual's complete symptom profile and neurological regions of damage in a larger sample size.

4aSC15. Effects of dynamic pitch manipulation methods on speech perception in noise by older listeners with hearing loss. Jing Shen (Commun. Sci. and Disord., Temple Univ., 1701 N. 13th St., Philadelphia, PA 19122, jing.shen@temple.edu) and Jingwei Wu (Epidemiology and Biostatistics, Temple Univ., Philadelphia, PA)

Dynamic pitch, as defined by the variation in fundamental frequency (f0) of speech, is a major acoustic cue in speech perception. While dynamic pitch manipulation negatively affects speech perception in noise on a group level, there is substantial variability across older listeners with hearing loss. The manipulation method used in previous studies, however, may distort pitch contours in natural speech. In this study, a new manipulation method is employed by exaggerating pitch contours above mean pitch. The primary purpose of the study is to compare the effects of these two manipulation methods on speech recognition in noise by older listeners with hearing loss. The secondary purpose is to examine the connection between individuals' responses to dynamic pitch manipulation and intonation perception. Speech reception threshold in noise was measured from 8 older listeners (mean age 74.5 years) with mild-moderate sensorineural hearing loss. Intonation perception was measured by a 2-alternative discrimination paradigm. Data up to date suggest a stronger effect from the new method, with the weakened dynamic pitch being more beneficial than the original pitch contour. Further, individuals' intonation perception is associated with their response to pitch manipulation. The theoretical and clinical implications will be discussed. [Word supported by NIH.]

4aSC16. Early evidence of cognitive involvement during sensorimotor integration for speech. Karen Hebert (Dept. of Occupational Therapy, Univ. of South Dakota, Vermillion, SD), Vikram Dayalu, Caryn Grabowski (Dept. of Speech-Lang. Pathol., Seton Hall Univ., Nutley, NJ), and Sona Patel (Dept. of Speech-Lang. Pathol., Seton Hall Univ., 123 Metro Blvd Interprofessional Health Sci. Campus, Nutley, NJ 07110, sona.patel@shu.edu)

Executive functioning is fundamental to motor processes that involve decision-making, and yet, the role of cognitive control during speech production is not well understood. The purpose of this study was to examine the relationship between auditory attentional decision-making ability and changes in acoustic parameters and speech error rates during sensorimotor integration for speech production. Seventeen healthy individuals first completed an auditory analog of the Attentional Network Test to assess attentional decision-making and then read passages under delayed auditory feedback (DAF). None of the acoustic parameters (speech rate, mean and standard deviation of fundamental frequency and intensity) were related to metrics of cognitive processing. However, auditory processing speed, measured as reaction time to pitch identifications ($r = -0.58$, $p < 0.05$) and auditory location identifications ($r = -0.66$, $p < 0.05$), was significantly related to the total number of dysfluencies (across articulation errors, stuttering-like disfluencies, and other disfluencies) under DAF. In other words, individuals who made quick auditory discriminations produced more dysfluencies under DAF. While preliminary, these findings suggest that cognitive functions including executive control are engaged during speech production. These findings can be understood in the context of diffusion models of decision-making and the impact of cognition on speech perception.

4aSC17. Learnability of ultrasound tongue imaging devices in speech-language pathology. Isabelle Marcoux (Linguist., UQAM - Univ. du QC a Montreal, Pavillon J.-A. De Sève, 320 Rue Sainte-Catherine Est, Montreal, QC H2X 1L7, Canada, marcoux.isabelle.8@courrier.uqam.ca), Lucie Ménard (Linguist., UQAM - Univ. du QC a Montreal, Montréal, QC, Canada), and Catherine Laporte (Dept. of Elec. Eng., École de technologie supérieure, Montreal, QC, Canada)

Ultrasound tongue imaging has shown potential for speech-language pathologists (SLPs) to evaluate and treat persistent articulatory disorders. However, SLPs typically begin with low to no familiarity with ultrasound. Thus, this study investigated an important aspect of ultrasound device usability: learnability for SLPs. The project was funded by an NSERC Engage grant in partnership with Clarius Mobile Health. 12 SLPs learned to use two ultrasound devices: a wireless device, provided by our partner Clarius, and a traditional device, to record clips of their or the experimenter's tongue. They then completed a questionnaire [French translation of the *System Usability Scale* (Brooke, 1996)]. Two expert judges evaluated the clips recorded by the SLPs for the choice of settings and the positioning of the probe. Results of the *SUS* show a better usability for the wireless device than the traditional device. SLPs appreciated the user-friendly tablet interface, possibly because they are already used to interacting with tablets. Clips analyses show a better choice of settings by the SLPs with the wireless device. The positioning of the probe, however, was better with the traditional device, possibly due to its smaller probe. In conclusion, US seems to have a good potential of usability in speech-language pathology, provided that SLPs receive training for US image interpretation. A traditional US device may require a longer learning period than a wireless model with tablet interface.

4aSC18. The tongue dorsum activity in children with cleft palate and typically developing children. Hedieh Hashemi Hosseinabad (Commun. Sci. and Disord., Eastern Washington Univ., 310 N. Riverpoint Blvd, Spokane, WA 99202, hhosseinabad@ewu.edu) and Joanne Cleland (Speech Lang. Therapy, Univ. of Strathclyde, Glasgow, Scotland, United Kingdom)

Objective: Cleft palate speech is characterized by abnormal lingual articulation. To date, most articulatory information has been collected using electropalatography, which records the tongue-palate contact but not the tongue shape. The present study uses ultrasound tongue imaging for quantitative analysis of tongue function in speakers with cleft palate. Two measures aim to quantify overuse of tongue dorsum in cleft palate articulations were compared with typical speakers. The hypothesis is higher degrees of tongue dorsum activity in cleft palate speech. Measures: High-speed ultrasound tongue contour data synchronized with the acoustic signal were recorded. Data from three speakers of American English (11–13 years old) with speech disorders resulting from cleft palate and three typically developing speakers were analyzed. Annotation and tongue tracing from ultrasound images were carried out in Articulate Assistant Advanced software. One of the measures quantified the extent of tongue dorsum excursion (DEI), and the other measure represented the place of maximal excursion (TCPI). The stimuli included a range of consonants in consonant-vowel sequences, with the vowels /a/ and /i/. In total, 360 values were obtained for each index (six speakers $\times$ six consonants $\times$ two vowels $\times$ five repetitions). Further results will be discussed in the meeting.

4aSC19. How hospital noise impacts intelligibility of medically related sentences. Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, tbent@iu.edu), Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR), Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska-Lincoln, Omaha, NE), and Sydney Perry (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Hospital noise often exceeds recommended sound levels set by health organizations, potentially leading to communication breakdowns between healthcare providers and patients. Despite this concern, there is a paucity of literature on speech perception in healthcare settings. Studies investigating how hospital noise impacts intelligibility have been limited by not using medical terminology or not including objective intelligibility measures with

listeners. To address these gaps, a new corpus of medically related sentences, with quantified word frequency, word familiarity, and sentence predictability, was employed in an intelligibility task. Recordings of the medically related sentences with different familiarity/frequency types (low/low, high/low, high/high) plus standard speech perception sentences were presented to listeners in quiet, speech-shaped noise, or hospital noise for transcription. Results showed main effects of sentence type and listening condition and an interaction between the two. Although listeners were less accurate on all sentence types in noise, sentences with low frequency and/or familiarity words showed the greatest intelligibility decreases in noise. Thus, miscommunications between healthcare providers and patients could occur when medical information is provided in noisy settings like hospitals, situations that could lead to lack of comprehension of discharge instructions or other medical communication errors. [Work supported by the IU Institute for Advanced Study.]

4aSC20. Amplitude modulation of background noise varies listeners' spectral weights for sentence recognition. Yi Shen (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246, shenyi@uw.edu) and Lauren Langley (Univ. of Washington, Seattle, WA)

The band importance function that captures how the spectral weight varies across frequencies was estimated for sentence recognition in noises with steady-state or fluctuating temporal envelopes from ten young, normal-hearing adult listeners. The test sentences were from either the IEEE or AzBio corpus. The background noise was a 12-talker babble, either unmodulated or amplitude-modulated using an 8-Hz sinusoidal modulator. In the co-located condition, the noise was presented from the same loudspeaker as the target sentence in front of the listener (0°). In the spatially separated condition, the noise was presented simultaneously from two loudspeakers on either side of the target speaker (±90°). The co-located condition was replicated in a separated test session, at least one week from the first session, which demonstrated the reliability of the estimated band importance functions. With the introduction of amplitude modulation, an increase of spectral weight in the 2-kHz frequency band and a decrease of weight in the 250-Hz band was observed, which was consistent for the two test materials and for the two spatial placements of the noise. Compared to amplitude modulation, spatial separation between the target and noise had a less influence on the band importance function.

4aSC21. Communication channel, emotion category and attentional requirement modulate emotional speech perception: Electrophysiological and behavioral evidence. Yi Lin (Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, carol.y.lin@foxmail.com), Xinran Fan (Shanghai Jiao Tong Univ., Shanghai, China), Hao Zhang (Shangdong Univ., Jinan, China), Fei Chen (Hunan Univ., Changsha, China), Hui Zhang, Hongwei Ding (Shanghai Jiao Tong Univ., Shanghai, China), and Yang Zhang (Univ. of Minnesota, Minneapolis, MN)

The present study employed electrophysiological and behavioral measures to examine the modulatory effects of communication channels, emotion categories and attentional requirements on the timing, neural oscillatory activities and behavioral performances of emotional speech processing. Thirty participants (15 women and 15 men) were presented with spoken words denoting happiness, neutrality and sadness in prosody or semantics. They were asked to attend to the emotional content (explicit tasks) and speakers' gender (implicit tasks) of the stimuli. Auditory event-related potential (AERP), inter-trial phase coherence (ITPC), accuracy, and reaction time data were recorded and analyzed. Results indicated that prosody (relative to semantics) and happiness (relative to neutrality and sadness) enjoyed greater processing advantages during the sensory processing of acoustic signals, initial derivation of emotional significance, and cognitive evaluation of emotional speech. However, these channel and category-specific effects were attenuated in implicit tasks and sadness processing. Explicit tasks also displayed greater perceptual sensitivities in early auditory processing, but exhibited greater processing difficulty during decision making. In addition, ITPC values served as significant predictors of AERP amplitudes across the three contextual factors. These findings reveal the neural and behavioral underpinnings of emotional speech processing, which is mediated by the dynamic interplay among channels, emotions and attention.

4aSC22. Prosodic characteristics of English speakers with Alzheimer's disease. Nicole Ebbutt (Linguist, Univ. of BC, 6368 Stores Rd., Vancouver, BC V6T 2B4, Canada, nicoleebbutt@gmail.com), Arian Shamei, Charissa Purnomo, and Bryan Gick (Linguist, Univ. of BC, Vancouver, BC, Canada)

Alzheimer's disease (AD) is a progressive neurodegenerative disorder. A large amount of work has focused on the automatic detection of AD, but few have examined the underlying acoustic correlates. Through the analysis of 42 speakers (21 AD and 21 control), research by Martinez-Sanchez *et al.* have reported flattened pitch trajectories which distinguish Spanish-speaking AD patients from those of controls [Martinez-Sanchez *et al.*, *Psicothema* **24**(1), 16–21 (2012)]. The present study aims to determine whether a similar effect is found in a large-scale study of English speakers with AD. Prosogram [Mertens, in Proceedings of Speech Prosody (2004)] was used to assess pitch trajectories of 203 English-speaking participants (128 AD and 75 control) from Dementiabank's Pitt Corpus [Becker *et al.*, *Arch. Neurol.* **51**(6), 585–594 (1994)]. AD patients exhibited greater pitch range and trajectories across phonation than controls, both within and across syllables. These preliminary findings conflict with previous observations of a flattened prosodic profile for AD patients. A possible reason for this discrepancy is a difference in speech elicitation tasks. Previous studies on Spanish-speakers used reading and delayed pronunciation tasks, whereas the present study utilizes a picture description task, a more natural elicitation method.

4aSC23. A Gaussian-mixture-model-based approach to classifying vowel place in speech signals. Ritik Patnaik (MIT, 362 Memorial Dr., Cambridge, MA 02139, rik01@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

In recent years, speech recognition systems have dramatically improved in performance through the development of general machine learning techniques. However, it is not always straightforward to interpret the mapping from the signal to the detected category. In the present work, we focus on the goal of transparency, specifying the processing steps that lead to robust modeling of vowel place using simple, descriptive Gaussian mixture models. We present a pre-processing and detection framework for vowel place, involving formant measurements, smoothing, and the GMM. This research aims to classify vowel place as a part of a larger speech recognition system. Vowel place was divided into 8 groups based on tongue advancement [Front/Back], height [High/Mid/Low], and root [Atr/Ctr]. Studies were performed using ~700 vowel-consonant-vowel utterances from the LAFF VCV database.

4aSC24. Acoustic analysis of vocal effort and speaking style. May Pik Yu Chan (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing University of Pennsylvania, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu) and Mark Liberman (Dept. of Linguist, Univ. of Pennsylvania, Philadelphia, PA)

"The singer's formant" is used by singers during voice projection. While a similar "actor's formant" has been found (Leino, 1993), other work finds greater source effects in projected voices (e.g., Master *et al.*, 2012). It remains unclear whether differences in source and/or filter remain among non-actors/singers in varying speaking styles. Recordings from the SRI-FRTIV corpus (Shriberg *et al.*, 2008) were used to examine the effects of vocal effort and speech style on vocal characteristics. 34 speakers of American English spoke in up to three levels of vocal effort in (1) Interview, (2) Conversation, (3) Reading, and (4) Oration tasks. The following measures were measured by LTAS based on the mid-50% of the recording's duration: (a) CPPS; (b) L1-L0; (c) alpha ratio; (d) mean frequency of the lowest peak; (e) the level difference between strongest peaks at 3–4 kHz and 0–1 kHz ("the actor's formant"); the level difference between the first formant region (300–800 Hz) and the (f) 1–2 kHz, (g) 2–3 kHz, (h) 3–4 kHz, and (i) 4–5 kHz range. Preliminary results using PCA find that general measures of spectral slope (PC1: h, i, c, g, b) better capture differences in vocal effort and speech style than more fine-grained source/filter measures (PC2: e, f, a, b, d).

4aSC25. Using ideal binary masking based on signal-to-noise ratio of temporal amplitude envelope to improve the intelligibility of speech in noise. Rahim Soleymanpour (Biomedical Eng., Univ. of Connecticut, 263 Farmington Ave., Farmington, CT 06030, rahim.soleymanpour@uconn.edu), Kia Golzari, Insoo Kim (Biomedical Eng., Univ. of Connecticut, Farmington, CT), Erin Heiney, Hillary Marquis (Dept. of Surgery, Div. of Otolaryngol., Head and Neck Surgery, Univ. of Connecticut, Farmington, CT), and Anthony J. Brammer (Biomedical Eng., Univ. of Connecticut, Farmington, CT)

Ideal Binary Masking (IBM), using prior information, improves speech intelligibility by attenuating noisy components with a scaling factor applied to the noise. The main challenge is to construct an appropriate decision-making model to identify noise- or speech- dominant components. In this study, we utilized the signal-to-noise ratio (SNR) of the temporal amplitude envelope in the frequency-time domain. We firstly divided the noisy speech from 200 Hz to 6 kHz, processed by MATLAB, into 16 contiguous subbands each with bandwidth approximately 1.5 times an equivalent rectangular bandwidth. The subband envelopes were produced by means of the absolute value of the signal. SNRs of the temporal envelope were calculated for 40 ms windows. The mask was unity when the SNR was greater than -5 dB; otherwise, it was 0.5. We evaluated the performance of the proposed IBM on word scores obtained with different speech in speech-spectrum shaped noise SNR values of -2, -4, -6, and -8 dB. Sixteen native speakers (age 28 ± 3 years) with normal hearing were recruited for the study and underwent the Modified Rhyme Test to assess intelligibility. Statistically significant increases of up to 20% in mean word scores were obtained by this IBM. [Work supported by NIOSH.]

4aSC26. Effects of transparent and opaque facemasks on audiovisual speech perception. Laura Koenig (Haskins Labs., 300 George St., New Haven, NY 06511, koenig@haskins.yale.edu), Melissa Randazzo (Commun. Sci. & Disord., Adelphi Univ., Garden City, NY), Paul J. Smith (Biobehavioral Sci., Teachers College, New York, NY), and Ryan Priefer (Commun. Sci. & Disord., Adelphi Univ., Garden City, NY)

Face masks reduce the spread of COVID-19 but hinder speech perception. Masks reduce visual feedback, filter the acoustic signal, and affect speech movements. Older adults, particularly those with aging related hearing loss, are impacted by the reduction of auditory and visual cues when communicating with masks. Face masks also pose a challenge to medical professionals when relaying important information to patients. Transparent masks with a clear window across the mouth may facilitate masked communication; however it is unknown if visual feedback through the transparent mask facilitates speech perception given the reduced acoustic signal. We examined speech intelligibility through masks using unpredictable sentences, recorded by two adults, that limit guessing words based on context. We presented sentences mixed with multitalker babble in four conditions (no mask, disposable, transparent, and N95 mask). Listeners, recruited online, typed what they heard. Percent words correct was scored by a text-matching algorithm, allowing for morphological errors and homonyms. We will determine the visual benefit of the transparent masks over the other masked conditions. Results of this study may increase awareness among healthcare providers about how different mask types can affect communication and empower individuals to decide which masks to wear in different communicative contexts.

4aSC27. Face masks, speech intelligibility, and listening effort. Sita Carraturo (Psychol. & Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, sita@wustl.edu), Violet A. Brown, Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO), and Jonathan Peelle (Otolaryngol., Washington Univ. in Saint Louis, St. Louis, MO)

The use of face masks recently has raised awareness about their effects on speech comprehension. In a series of two studies, we assess how different masks speech intelligibility and listening effort. In the first study, we compared four mask types in three noise levels for young and older adult listeners. Stimuli were presented audiovisually. Results showed that, in quiet, speech intelligibility was not greatly affected by any of the masks relative to no-mask conditions. However, in background noise, all masks reduced intelligibility. Masks with transparent windows and cloth masks with filters resulted in the lowest intelligibility scores, while surgical face masks had the least effect. Participants' subjective effort ratings also reflect that comprehending speech through face masks requires greater effort. Overall, older adults' listening effort ratings were higher and their intelligibility scores were lower. In the second study, we assess listening effort with pupillometry. We present the same stimuli to a group of young adults in six audio-only conditions: no-mask in quiet, no-mask in moderate noise, and four masks in moderate noise. In addition to intelligibility accuracy and pupil dilation, we also collect subjective effort and performance ratings, pure tone averages, and working memory performance scores.

4aSC28. Recommendations for improving communication while performing tasks under noisy, reverberant and delayed communication transmission conditions. Vishakha Rawool (Commun. Sci. & Disord., Univ. of MS, 1006 Briarwood Dr., Oxford, MS 38655, vishakharawool8@gmail.com)

This study, which is part of a larger study, was designed to generate recommendations for improving remote collaborative communication. Ninety (45 men and 45 women) participants were divided into 30 teams with 3 members in each team. Each team built different LEGO models under three communication transmission delay conditions in the following order: No delay, 750 ms delay and 5000 ms delay. One individual served as the LEGO builder, who did not have access to the instructions. The remaining two team members provided instructions for building the models. Participants were in separate rooms and communicated with each other via earphones. Background white noise (50 dB SPL) was introduced in the ear canals and reverberation was applied to the transmitted communication. After the three delay conditions, participants responded to questions. Their responses were coded into different categories for conceptual analyses. Several recommendations emerged including providing accurate, concise and clear instructions, ensuring ongoing alignment of the spatial orientation of the LEGO model across team members, standardizing/aligning phrases or terms across the team members, and increasing team and task familiarity. Additional recommendations will be discussed during the presentation with examples. [Acknowledgement: Graduate students Sydney Osbarn and Melissa Reyes assisted in data collection.]

4aSC29. Perceived effects of communication transmission delays during remote collaboration under noisy and reverberant settings on communication, team and task performance, and well-being. Vishakha Rawool (Commun. Sci. & Disord., Univ. of MS, 1006 Briarwood Dr., Oxford, MS 38655, vishakharawool8@gmail.com)

This study, which is part of a larger study, was designed to investigate the perceived effects of communication delays on well-being, communication, and task and team performance. Ninety (45 men and 45 women) participants were divided into 30 teams with 3 members in each team. Each team built different LEGO models under three communication transmission delay conditions in the following order: No delay, 750 ms delay and 5000 ms delay. One individual served as the LEGO builder, who did not have access to the instruction manual. The remaining two team members provided instructions for building the models. Participants were in separate rooms and communicated via earphones. Background white noise (50 dB SPL) was introduced in their ear canals and reverberation was applied to the transmitted communication. After each of the delay conditions, participants completed a visual analogue scale with 39 items related to well-being, communication and team and task performance. ANOVA revealed significant perceived effects of 5000 ms delay on communication, team dynamics, task performance and well-being. In the well-being categories, women yielded poorer scores than men. Detailed results will be included in the presentation along with recommendations. [Acknowledgement: Graduate students Sydney Osbarn and Melissa Reyes assisted in data collection.]

4aSC30. Effects of word frequency and phonological neighborhood density on speech recognition in background noise and competing speech.

Nicole Whittle (Res. Service, Veteran's Affairs Loma Linda Healthcare System, 11201 Benton St., Loma Linda, CA 92357, nicole.whittle@va.gov), Christian Herrera Ortiz, Marjorie R. Leek, Jerome Heidrich (Res. Service, Veteran's Affairs Loma Linda Healthcare System, Loma Linda, CA), Mark Jenkins (Veteran's Affairs Northern California Healthcare System, Martinez, CA), and Jonathan H. Venezia (Res. Service, Veteran's Affairs Loma Linda Healthcare System, Loma Linda, CA)

Clinical speech-in-noise tests typically use materials without contextual constraint or balanced for linguistic properties like word/phoneme frequency. However, real-world linguistic context effects can be substantial and vary by listener and scenario. Here, 38 participants completed the Theo-Victor-Michael (TVM) speech test in four types of background: speech shaped noise (SSN), speech-envelope modulated noise (envSSN), one competing talker (1T), and two competing talkers (2T) (Helfer and Freyman, 2009). The TVM is a matrix test using keywords from a corpus of one- and two- syllable nouns that vary considerably in word frequency (FREQ) and phonological neighborhood density (DENS). Bayesian logistic regression was used to estimate the effects of FREQ/DENS on TVM performance. A multinomial model was used for 1T/2T to assess reporting of target and distractor keywords. Overall, percent-correct recognition increased with increasing keyword FREQ and decreased with increasing keyword DENS. Effects were larger in SSN/envSSN than 1T/2T. Statistically significant but small effects of FREQ/DENS were observed on distractor responses in 1T/2T. Adjusting performance for FREQ/DENS substantially shifted the distribution of scores but only for SSN/envSSN. Performance in 1T/2T may be dominated by non-linguistic factors, and/or less sensitive to FREQ/DENS due to higher difficulty or linguistic competition from the background talkers. [Work supported by VA RR&D Service.]

Session 4aSP

Signal Processing in Acoustics, Physical Acoustics, Animal Bioacoustics, Noise, and Biomedical Acoustics: Signal Processing Methods for Source Classification and Localization in Real Acoustic Environments I

Ning Xiang, Cochair
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Kay L. Gemba, Cochair
Code 7162, U. S. Naval Research Laboratory, 4555 Overlook Ave., SW Washington, DC 20375

Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. Oceanographic property estimation with ambient noise on the New England shelf. John Gebbie (Metron, Inc., 2020 SW 4th Ave., Ste. 170, Portland, OR 97201, gebbie@metsci.com) and Martin Siderius (Portland State Univ., Portland, OR)

Acoustic propagation through the ocean waveguide is strongly affected by the vertical water sound velocity profile and the geoacoustic properties of the seabed. Direct measurement is often prohibitive, however, the ambient noise field itself contains information about these properties, and therefore can be estimated via passive acoustics. The ambient noise field is driven by wind and wave action at the surface. This energy propagates through the waveguide ultimately affecting the vertical distribution of the noise energy, but also picking up environmental information along the way. A vertical line array is a useful tool for observing the noise field and estimating properties of the environment. The observed element data consists of correlated stochastic processes. The fundamental limit on the accuracy with which environmental properties can be estimated is given by the Cramér-Rao Lower Bounds (CRLB). The CRLB thus provides useful insights into which environmental properties can be independently or jointly estimated. A maximum likelihood (ML) estimator will asymptotically approach the CRLB as the number of snapshots increases. The ability to estimate different environmental properties and ML performance is demonstrated with simulations and real data collected in May 2021 off the New England shelf. [Work sponsored by the Office of Naval Research.]

8:25

4aSP2. Localizing a quiet moving source with range-coherent matched field processing. Franklin H. Akins (Scripps, UCSD, 9500 Gilman Dr., La Jolla, CA 92093, fakins@ucsd.edu) and William Kuperman (Scripps, UCSD, La Jolla, CA)

We introduce range-coherent matched field processing (MFP) to lower the signal-to-noise ratio (SNR) threshold required to successfully localize a narrowband source in a well-characterized shallow water environment. To accomplish this, range-coherent MFP coherently processes snapshots formed from a moving source. This approach differs from existing MFP approaches that treat each snapshot as having a random phase. Range-coherent MFP requires determination of the source phase acquired between snapshots. With that information, MFP can be applied to the cross-spectrum of the ''supervector'' of concatenated snapshots acquired at different times. Viewed another way, range-coherent MFP is simply MFP applied to a passive synthetic aperture formed from a moving source. The synthetic aperture geometry depends on source velocity, which is included in the MFP search space. Range-coherent MFP produces robust velocity estimates at low SNR, which permits the use of a longer FFT in pre-processing. The synthetic aperture array gain plus the increased input SNR afforded by the enhanced pre-processing significantly lower the signal level required for successful localization. We present the application of range-coherent MFP to data from the SWellEx-96 experiment to localize a source that is too quiet for conventional methods to localize.

8:45

4aSP3. Estimation of sound speed profiles in deep water using perturbative inversion. Geoffrey F. Edelmann (U. S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil) and Joeseph F. Lingevitch (U. S. Naval Res. Lab., Washington, DC)

A method to invert for the local sound speed profile (SSP) using only acoustic wavenumbers is proposed. Ocean sounds can be treated as acoustic energy trapped as discrete modes within the water column. Using estimated Eigenvalues (wavenumbers) from one or more modes, a method to numerically invert for the SSP can be posed as a linear Fredholm integral equation of the first kind. Therefore, the vertical SSP can be estimated with a well-placed, sufficiently long, array at one or more frequencies. Though this inversion can be unstable and non-unique, recent improvements in sparse inversions can lead to robust estimates even without an accurate starting SSP. Using the perturbative difference between the estimated wavenumbers and those calculated from the acoustic kernel, both the SSP and

kernel are updated each iteration. Careful consideration must be made of the acoustic frequency, number of modes, a priori environmental information (e.g., water depth), and array length. The method will be first demonstrated on simulated recordings based on the LIDDEX data set. [Work supported by ONR.]

9:05

4aSP4. Real-time, model-aided ranging for small scale operations in the Beaufort Sea, an ice-covered, double ducted environment. EeShan Bhatt (Woods Hole Oceanographic Inst., 77 Massachusetts Ave., 5-223, Cambridge, MA 02139, eesh@mit.edu), Oscar A. Viquez (Massachusetts Inst. of Technol., Cambridge, MA), Bradli Howard (MIT-WHOI, Cambridge, MA), and Henrik Schmidt (Massachusetts Inst. of Technol., Cambridge, MA)

This talk will cover operational methods and results for vehicle navigation from the Ice Exercise in March 2020 (ICEX20), in the Beaufort Sea. For short range acoustic transmissions, the total ice cover and the double ducted environment co-create complex multipath uncertainty. We assumed that the horizontal group velocity between source and receiver was smoothly varying, and embedded a "Virtual Ocean" with a real-time ray tracing engine to predict the horizontal group velocity for range estimation. This model-aided psuedorange approach, which enabled a successful vehicle recovery, favored the least multipath possible such that psuedoranges tended to overestimate the GPS-derived range by roughly 10 m. In post-processing, we modify the group velocity estimation method to consider varying degrees of multipath and achieve a mean absolute error of roughly 4 plus or minus 4 meters, rivaling GPS performance. To our knowledge, these results are the first field experiment to demonstrate a real-time, model-based data processing to constrain ranging error for underwater navigation. [Work supported by ONR & the NDSEG fellowship.]

9:25

4aSP5. Moving source tomography with controlled source-tow observations. Kay L. Gemba (Acoust. Div., U. S. Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375, kay.gemba@nrl.navy.mil), Heriberto Vazquez, Jit Sarkar (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), Jeffrey D. Tippman (Intelligence and Space Res. Div., Los Alamos National Lab., Los Alamos, NM), Bruce Cornuelle (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California at San Diego, San Diego, CA), and William Kuperman (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA)

A moving source tomography experiment was conducted in 2016 off the coast of Southern California. From the travel time data extracted from the experiment, ocean sound speed and its uncertainty were estimated. The moving source poses extra complexity and adds components that need to be accounted for in a successful inversion. The sound speed estimation with uncertainties was validated against *in situ* observations to quantify the efficacy and accuracy of the method. Experimental results and additional simulations suggest that the ray diversity available in the moving source case reduces the posterior sound speed uncertainty compared to the fixed source case. [Work sponsored by the Office of Naval Research.]

9:45–10:05 Break

Contributed Papers

10:05

4aSP6. Target localization in range and depth from a passive linear towed array. Thibault Roche (THALES DMS, Paris, France), Iannis Bennaceur (Thales DMS, 525 Rte. des Dolines, Valbonne 06560, France, iannis.bennaceur@fr.thalesgroup.com), Xavier Cristol, and Gilles Gaonach (Thales DMS, Valbonne, France)

Stealthily estimating the range and depth of a sound source from a submarine passive sonar is of capital tactical importance: it allows to adopt the appropriate escape or attack strategy towards the threat. The main difficulty in passive localization is that the transmitted waveform is unknown. Nevertheless, array processing can reveal some features of the acoustic field allowing localization, such as the arrival angles of the multiple paths, and the time delays between them. In this paper, we have compared two methods. The first one is a direct method, which simulates the field features from a set of possible source locations, and compares them to the measured features. The second one, an indirect method, uses the measured features to backpropagate the acoustic field. The former method has already been tested on simulated and real data in (Bennaceur *et al*., in OCEANS'18 MTS/IEEE, Charleston) and (Bennaceur *et al*., in OCEANS 2021, Singapore-U.S. Gulf Coast). Here, both methods are applied on real signals from a linear towed array, using bearings only and both bearings and time delays. The main advantage of these methods is that they are automatic in the sense that no sonar operator is required to estimate the source location.

10:20

4aSP7. A non-Euclidean approach to source localization in an ocean waveguide. Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

In recent years, there has been significant interest in the natural extension of signal processing techniques based on Euclidean distances to non-Euclidean metrics. Covariance or cross-spectral density matrices (CSDMs), often the basis for localization and detection processors, can be interpreted as "points" in a Riemannian manifold on which a distance metric can be defined. This mathematical structure has proven quite useful in a number of processing schemes involving detection, source localization and classification of data. In this presentation, I consider a fully geometric, non-Euclidean approach to source localization in a simulated shallow water ocean waveguide where the sound speed field is stochastic due to the presence of internal gravity waves. A number of different CSDM estimators are constructed by formulas describing the mean or median of a set of sample dyads obtained from Green function realizations of the propagating field from a source point to a vertical array of acoustic sensors. These CSDM estimates are not derived by statistics, but through the geometric structure of the CSDM. The resulting estimates are incorporated into matched-field localization processors whose measure of similarity or distance between CSDMs is determined on the manifold. Work supported by funds from the Office of Naval Research.

10:35

4aSP8. Automated matching of long range underwater acoustic arrivals to predictions using a minimum variance Approach. Cristian E. Graupe (Ocean Eng., Univ. of Rhode Island, 30 Fish Rd., URI Narragansett Bay Campus Rm 13-14, Narragansett, RI 02882, graupec@uri.edu), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Peter F. Worcester, Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), and Bruce M. Howe (Dept. of Ocean and Resources Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

An automated method was developed to align underwater acoustic measurements taken at various depths and ranges to a reference model of acoustic arrival structure. Data used to demonstrate the method were collected by four autonomous underwater vehicles deployed in the Philippine Sea as part of an ocean acoustic tomography experiment. The arrivals were measured in the upper 1000 m of the ocean at ranges spanning several hundred kilometers from 5 moored acoustic tomography sources. The primary objective is to accomplish automatic source-receiver ranging by aligning measurements of long range acoustic arrivals to a single reference model. Acoustic arrival time structure for pulse compressed signals at long ranges is relatively stable, yet real ocean variability presents challenges in acoustic arrival matching. The presence of internal waves scatters the acoustic arrival structure and can introduce spurious arrivals in the measured data. This method takes advantage of simple projections of the measured structure onto the model space with constraints informed by scattering statistics consistent with the Garrett Munk internal wave energy spectrum. Compared to manual matching of the measured arrivals to eigenray models, more than 90% of the automatically obtained range estimates were within 150 m of the manually obtained range estimates.

10:50

4aSP9. Clustering of sperm whales recorded by a single hydrophone. George Drouant (Oregon Inst. of Technol., 3201 Campus Dr., Klamath Falls, OR 97601, george.drouant@oit.edu) and Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

Sperm whales (*Physter macrocephalus*) produce echolocation clicks composed of several pulses while searching for food when diving. The clicks are composed of segments three of which are: the p_0 pulse (3 kHz to 15 kHz), the p_1 pulse (3 kHz to 15 kHz), and the Low Frequency (LF) signal (<3 kHz) (Zimmer, 2005). The LF signal is omnidirectional in nature unlike the p1 pulse which is very directional. The power in the LF signal recorded by a single hydrophone is a function of the distance of the whale from the hydrophone and the source level of the whale. The power contained in the LF portion of the clicks can be used to assign clicking whales to groups with each group consisting of a least one individual. The inter-pulse interval (IPI) can then be used to determine the lengths of the whales in each cluster adding confidence to the estimated number of whales producing the clicks. Continuous Wavelet Transforms are used with underwater acoustic data recorded in the Gulf of Mexico in 2015 to determine the power of the LF signal, the length of the clicking whale and the alignment of the acoustical axis of the whale with the hydrophone.

11:05

4aSP10. Passive acoustic ship detection performance near the Port of Sept-Iles, Quebec. Marina Antipina (Oceanogr., Dalhousie Univ., 5799 Cunard St., Halifax, NS B3K 1C9, Canada, antipina.marina96@gmail.com), David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), S. B. Martin, and Julien Delarue (Jasco Appl. Sci. (Canada) Ltd., Halifax, NS, Canada)

An acoustic recorder was deployed near the Port of Sept-Iles, Quebec in the fall of 2020 and collected six months of data on a four-channel orthogonal array. The system, a JASCO C-lander, operated on a duty cycle consisting of 340s of data recorded at 32 kHz sampling rate, 1 min of data recorded at 256 kHz sampling rate followed by 500s of sleep. Data were stored on SD memory cards for post-retrieval analysis. Vessels were detected using narrowband tonals produced by their propulsion system and other rotating machinery and the sound pressure level (SPL) for each minute of data in the 40–315 Hz shipping frequency band was then computed. A 10 min shoulder period before and after the detection was then searched for the highest 1 min SPL which was identified as closest point of approach (CPA) time for each acoustic contact. Vessel track data from the automatic identification system (AIS) were used to compute CPAs for vessels carrying an AIS transponder during the deployment period. A comparison of the two results was used to identify missed and false detections, and to assess the algorithm's performance. Recommendations for implementing an improved detection approach will be discussed.

11:20

4aSP11. Gaussian process model selection for parameter estimation in ocean acoustics. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wjenkins@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

The dynamic nature of the ocean environment presents numerous challenges to parameter estimation in ocean acoustics. We present a variational Bayesian method which selects an optimal model for parameter estimation in ocean acoustics using Gaussian process (GP) regression. GP model selection is based on Bayesian model selection, which treats the parameters and kernel selection of the model as hyperparameters and uses Bayesian inference to compute a model's likelihood with respect to its hyperparameters. However, in Bayesian model selection, the integrals over parameter space are often intractable and require approximation or sampling methods such as Markov chain Monte Carlo. GP renders these integrals tractable by treating hyperparameters as multivariate Gaussian distributions, which greatly speeds up the optimization. Furthermore, model selection using GPs allows for straightforward evaluation of uncertainty in the selected model's likelihood. In this study, we compare results of acoustic parameter estimation from Bayesian and GP model selection using an ocean acoustic propagation model.

Session 4aUW

Underwater Acoustics, Signal Processing in Acoustics, Computational Acoustics, Acoustical Oceanography, and Physical Acoustics: Waveguide Invariant Theory and its Applications I

Heechun Song, Cochair
SIO, UCSD, 8820 Shellback Way, Spiess Hall, Rm. 448, La Jolla, CA 92093-0238

Julien Bonnel, Cochair
Woods Hole Oceanographic Institution, 266 Woods Hole Rd, MS# 11, Woods Hole, MA 02543-1050

Altan Turgut, Cochair
Naval Research Laboratory, Acoustics Division, Washington, D.C. 20375

Chair's Introduction—8:20

Invited Papers

8:30

4aUW1. On the invariance of the waveguide invariant. Daniel Rouseff (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, rouseff@apl.washington.edu) and Lisa Zurk (Univ. of Washington, Seattle, WA)

Constructive and destructive interference between different acoustic ray paths or modes is a common feature of propagation in the ocean waveguide. Russian investigators first distilled these interference effects into a single quantity, the so-called waveguide invariant. Over the last two decades, investigators have incorporated the waveguide invariant into signal processing algorithms for applications including localization, tracking, and environmental inversion. An important caveat is that a specific numerical value for the waveguide invariant sometimes applies for only a subset of all propagating modes; different numerical values can apply for different subsets. Under appropriate conditions, the waveguide invariant can appear to bifurcate with acoustic data simultaneously showing traits from both values. In the present work, a derivation is sketched that emphasizes what is actually invariant about the waveguide invariant. The proposed approach permits a more ready integration of the waveguide invariant into signal processing algorithms. Examples are presented for both deep- and shallow-water scenarios.

8:50

4aUW2. The waveguide invariant: Wavefield structure, ray stability, time spreads and modal dispersion. Michael G. Brown (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, mbrown@rsmas.miami.edu)

Prior to the seminal work of Chuprov and Grachev, who introduced the underwater acoustic waveguide invariant, the asymptotic equivalent of the same quantity was recognized as playing an important role in Hamiltonian dynamics. A consequence is that the waveguide invariant controls, to a large extent, underwater acoustic ray dynamics. In this presentation some of the less well known properties of underwater acoustic wavefields, based on both ray- and mode-based descriptors, that are controlled by the waveguide invariant are reviewed. These properties include overall wavefield structure, ray stability, time spreads and modal dispersion. [Work supported by ONR.]

9:10

4aUW3. Nature of waveguide invariant for multilayer sediment on New England shelf in 20–4000 Hz band. David P. Knobles (Phys., Knobles Sci. and Anal., 5416 Tortuga Trail, Austin, TX 78731, dpknobles@kphysics.org), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, RI), William Hodgkiss (MPL Univ Calif. at San Diego, San Diego, CA), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The waveguide invariant β_{mn} (f), $m,n = 1,2,\ldots,N$ where N is the number of *observed* modes is inferred from sound emitted by merchant ships and from a towed mid-frequency source emitting CHIRP signals. The acoustic data were recorded on VLAs deployed in the New England *Mudpatch* during the Seabed Characterization Experiment 2017 in about 75 m of water. The received spectrograms show clear striation patterns out to 4 kHz. The seabed consists of about 12 m of mud over multiple sand layers extending down to about 300 m. Starting at $\beta_{12}(F) = \infty$ at F about 25 Hz, $\beta_{mn}(f)$ is extended to 4 kHz. Below 500 Hz β_{mn} exhibits a *non-Pekeris behavior* which results from the depth dependence of the multilayer seabed. Above 500 Hz the behavior of β_{mn} begins to exhibit more of a *Pekeris behavior*, although there exist features that result from the finite thickness of the mud layer. A conclusion is that β_{mn} (f) is sensitive to

the geoacoustic structure over a large bandwidth, and thus has the potential to be utilized for remote sensing of the seabed with wideband sources.

9:30

4aUW4. Waveguide Invariant-based mitigation of reverberation and interference in active and passive sonar. Jeffrey Krolik (Dept. of Elec. and Comp. Eng., Duke Univ., 100 Sci. Dr., Rm. 130, Durham, NC 27708, jlk@duke.edu)

Multipath propagation poses many challenges for broadband sonar signal processing in underwater acoustic channels. The waveguide invariant (WI) provides a useful relation between the temporal and frequency fading characteristics in shallow-water channels that can been exploited to improve broadband target detection in both active and passive sonars. In active sonar, the WI motivates an alternative to conventional constant-false-alarm rate (CFAR) normalization of the background by averaging over observed time-frequency striations to estimate the reverberation level. A WI-based generalized likelihood ratio test is reviewed and shown to outperform conventional cell-averaged and frequency-invariant CFAR detectors in both simulation and with real Mediterranean data. In passive towed-array sonar, targets near endfire are often masked by angularly spread broadband multipath interference whose statistics must be estimated with limited training data for effective suppression. A WI relationship characterizing the frequency-dependence of wavenumber differences permits averaging of narrowband array snapshots to form a low-rank broadband "focused" covariance matrix. Simulation experiments in a realistic ocean environment are reviewed showing that adaptive beamforming using WI focused covariance matrices can provide a significant array gain improvement over conventional adaptive methods with limited observation time. Finally, the potential for waveguide invariant-based blind source separation is discussed. [Work supported by ONR.]

9:50

4aUW5. Adaptive array invariant for source-range estimation in shallow water. Gihoon Byun (MPL, Scripps Inst. of Oceanogr., 8820 Shellback Way, Spiess Hall, Rm. 444, La Jolla, CA 92093-0238, gbyun@ucsd.edu) and Heechun Song (SIO, UCSD, La Jolla, CA)

The array invariant (χ) developed for source-range estimation in shallow water is based on the broadband dispersion characteristics in ideal waveguides, summarized by the waveguide invariant, $\beta = \cos^2\theta$. This approach relies on the waveguide invariant being constant (e.g., $\beta = 1$), valid for small propagation angles, but introduces a significant range error as the propagation angle increases. In this talk, the standard array invariant is extended to fully incorporate the angle dependence of the waveguide invariant, referred to as the *adaptive array invariant*, denoted by $\chi_\beta = \chi/\beta$. The remarkable performance enhancement is demonstrated (1) using a 56.25 m vertical array for a surface ship (200–900 Hz) in approximately 100-m shallow water and (2) using a 2.8 m vertical array for a broadband source (0.5–3.5 kHz) towed on a continental slope where the water depth varies from 87.5 to 55 m over a 5-km range.

10:10–10:30 Break

Contributed Papers

10:30

4aUW6. Environmental probing using waveguide invariant. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375, altan.turgut@nrl.navy.mil)

In continental shelves, barotropic tides, internal tides, and nonlinear internal waves could potentially be monitored using acoustic waveguide invariant theory. When temporal variability of broadband acoustic signals is measured along a fixed track, frequency shifts of constant acoustic-intensity level curves can be used to capture temporal behavior of these oceanographic processes. In addition, acoustic intensity striations generated by a towed source or a ship of opportunity could be used to estimate seabed properties. The feasibility of environmental probing for the oceanographic processes and sediment types is discussed in terms of the available group of acoustic modes, frequency bands, and experimental geometries. Several experimental measurements and broadband numerical modeling results are provided to assess the limits of each probing method under winter and summer conditions. [Work supported by the ONR.]

10:45

4aUW7. Focusing Green's function using the waveguide invariant theory. Daehwan Kim (Korea Maritime & Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, Busan 49112, Republic of Korea, oceankim823@gmail.com), Donghyeon Kim, and J. S. Kim (Korea Maritime & Ocean Univ., Busan, Republic of Korea)

Sound propagated in an ocean waveguide is greatly influenced by the characteristics of the underwater acoustic channel. Thus, knowledge of the propagation characteristics (i.e., impulse response or Green's function) is key to various ocean remote sensing applications. Recently a relationship was verified that the waveguide invariant property allows a time-domain Green's function observed at one location to be extrapolated to adjacent ranges [Song *et al*., *J. Acoust. Soc. Am.* **149**, 2173–2178 (2021)]. Here, we exploit this relationship to coherently combine the time-domain Green's function from adjacent ranges to improve the SNR of the multipath impulse response. Numerical simulation results are presented and discussed.

11:00

4aUW8. Extraction of seabed properties in mid-frequency range using waveguide invariant applied to broadband signals. Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Rm. 140, Evans Hall, Newark, DE 19716, badiey@udel.edu), Lin Wan, and Christian D. Escobar-Amado (Elec. and Comput. Eng. Dept., Univ. of Delaware, Newark, DE)

One of the fundamental problems in shallow water environments is to assess the seabed properties. The waveguide invariant can be used in both passive and active sonar processing, source ranging, time reversal focusing, and in geoacoustic inversion processes. In Pekeris waveguide, the waveguide invariant value is approximately 1 and the slope of the interfering acoustic intensity fringes relates to the bottom sound speed. Using the slope of the striations as a variable, the spacing of the interference in range (or frequency) can be related to the penetration of energy into the bottom. The depth given by spacing of real arrivals has an effective depth of penetration which depends on the geoacoustic properties of the bottom sediments. Here, we study the frequency dependence of the waveguide invariant due to sediment layer structure and physical properties and provide a detailed data/model comparison for a shallow water range independent waveguide and extract the seabed properties for the frequency band 1.5 to 4.0 kHz using recent data obtained in shallow water environments. [Work supported by ONR 321OA.]

11:15

4aUW9. Range estimation of a distant ship using a vertical array and a guide ship. Gihoon Byun (MPL, Scripps Inst. of Oceanogr., 8820 Shellback Way, Spiess Hall, Rm. 444, La Jolla, CA 92093-0238, gbyun@ucsd.edu) and Heechun Song (SIO, UCSD, La Jolla, CA)

In this talk, we present a method for estimating the range of a distant ship in shallow water using a vertical array and a guide ship at a known range close to the array. The method involves a combination of four well-known approaches: (1) blind deconvolution; (2) waveguide invariant; (3) virtual receiver; and (4) array invariant. The main idea is to transform the original task of estimating the range of a distant ship using a vertical array into estimating the range with a synthesized horizontal array around the guide source exploiting the waveguide invariant theory. The proposed method is demonstrated for a surface ship radiating broadband noise (100–500 Hz) at 10 km range, using a 56.25-m long vertical array in approximately 100-m water and the same ship at 2 km range as the guide ship.

Invited Paper

11:30

4aUW10. The waveguide invariant for a Pekeris waveguide. Heechun Song (SIO, UCSD, 8820 Shellback Way, Spiess Hall, Rm. 448, La Jolla, CA 92093-0238, hcsong@ucsd.edu) and Gihoon Byun (SIO, UCSD, La Jolla, CA)

A closed-form waveguide invariant β for a Pekeris waveguide is derived. It is based on the modal Wentzel–Kramers–Brillouin (WKB) dispersion equation and implicit differentiation, in conjunction with the concept of the angle-dependent "effective boundary depth,'' $\Delta H(\theta)$. First, an explicit expression for $\beta(m,n)$ between mode pairs is derived assuming an ideal waveguide of the effective waveguide depth $H+\Delta H(\theta)$, which provides an excellent agreement with the exact value calculated for the Pekeris waveguide of depth H using KRAKEN. Then, a closed-form expression for a group of adjacent modes is derived: $\beta = (H + \Delta H)/(H/\cos^2\theta - \Delta H)$, which reduces to $\beta = \cos^2\theta$ as $\Delta H/H \ll 1$, the analytical expression for an ideal waveguide. For applications to source-range estimation, the waveguide invariant β is used to derive the theoretical beam-time migration (i.e., dispersion curve) in a Pekeris waveguide.

Session 4pAB

Animal Bioacoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Underwater Acoustics: Applications of Bioacoustics in Killer Whale Conservation II

Tina M. Yack, Cochair
EcoSound Bioacoustics, 9423 Saint Andrews Dr., Santee, CA 92071

Marla M. Holt, Cochair
Conservation Biology Division, NOAA/NMFS Northwest Fisheries Science Center, 2725 Montlake Blvd East, Seattle, WA 98112

Chair's Introduction—1:30

Invited Papers

1:35

4pAB1. A comparison of EAR and SoundTrap performance for acoustic monitoring of resident killer whales. Candice Emmons (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd East, Seattle, WA 98112, candice.emmons@noaa.gov), M. Bradley Hanson, Marla M. Holt (Conservation Biology Div., NOAA/NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), and Arial Brewer (Marine Mammal Lab., Alaska Fisheries Sci. Ctr., NOAA, Seattle, WA)

Since 2006, a network of hydrophones has been deployed along the outer coasts of Washington, Oregon, and California to improve our understanding of endangered Southern Resident killer whale (SRKW) movements outside the inland waters of the Salish Sea and refine Critical Habitat designation. During this time period, improvements in methods and equipment have resulted in longer deployments, improved ability to detect resident killer whales (RKW) and monitor their acoustic behavior, and added the capability to characterize ambient and ship noise throughout SRKW range. It is important to understand how these changes might impact the results of long term studies on RKW movements and distribution. In this study, we co-located Ecological Acoustic Recorders (EARs) and SoundTraps at five locations along the Washington coast, analyzed each data set separately, and compared the results. Preliminary analysis of these data resulted in a 1.6 to 3.4 times increase in the number of days with RKW detections in the SoundTrap data compared to the EAR data depending on location. We will present the results of this comparison, investigate potential causes of the differences between recorder type, and highlight the implications of these differences to previous knowledge of RKW movements along the Washington coast.

1:55

4pAB2. A Monte Carlo approach to modelling detection ranges for killer whales in inshore waters of British Columbia, Canada. Melanie Austin (JASCO Appl. Sci., 2305-4464 Markahm St., Victoria, BC V8Z 7X8, Canada, melanie.austin@jasco.com), Xavier Mouy (JASCO Appl. Sci., Victoria, BC, Canada), Harald Yurk (Aquatic Ecosystems Marine Mammal Sci., Fisheries and Oceans Canada, Pacific Sci. Ctr., Vancouver, BC, Canada), and Jennifer Wladichuk (JASCO Appl. Sci., Victoria, BC, Canada)

Passive underwater listening stations are often used to monitor the presence, distribution and movements of marine mammals. This requires an understanding of the distances at which marine mammal sounds can be detected at a location and at different times given varying ambient noise conditions. Here, we describe a Monte Carlo approach for determining call detection probabilities as a function of distance for networked underwater listening stations deployed by Fisheries and Oceans Canada in the Salish Sea to track endangered Southern Resident Killer Whales. We used ambient sound levels measured *in situ*, modelled propagation losses determined by two acoustic models, and applied Monte Carlo simulations to capture the variability in call source level and animal depth. Given that only some parts of the call frequency spectrum may be responsible for the maximum detection range, the analysis was carried out independently for consecutive 300 Hz frequency bands. Median detection range estimates ranged from 700 m at the noisiest time and location to 8 km at the quietest. A sensitivity analysis revealed that the frequency distribution of source levels used for the analysis was a major factor affecting detection range results, while differences in propagation losses between summer and winter were less important.

2:15

4pAB3. Integrating visual and acoustic observations to build an "intelligent" killer whale movement forecast system. Ruth Joy (Environ. Sci., Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, rjoy@sfu.ca), Steven Bergner (Computing Sci., Simon Fraser Univ., Burnaby, BC, Canada), Dave Campbell (Mathematics and Statistics, Carleton Univ., Ottawa, ON, Canada), Mike Dowd (Mathematics and Statistics, Dalhousie Univ., Halifax, NS, Canada), Fabio Frazao (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Oliver S. Kirsebom (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Paul Nguyen Hong Duc (School of Mathematics and Statistics, Carleton Univ., Ottawa, ON, Canada), Bruno Padovese (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Marine Randon (Statistics and Actuarial Sci., Simon Fraser Univ., Burnaby, BC, Canada), Amalis Riera Vuibert (Biology, Univ. of Victoria, Victoria, BC, Canada), Sadman Sakib (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Scott Veirs (Orcasound, Seattle, WA), val veirs (Orcasound, Friday Harbor, WA), Jennifer Wladichuk (Univ. of Victoria/JASCO Appl. Sci., Victoria, BC, Canada), and Harald Yurk (Ecosystem Sci. Div., Fisheries and Oceans Canada, Vancouver, BC, Canada)

Shipping traffic continues to grow in the Salish Sea with considerable marine industrial developments planned for the region. With this comes an increase in the risk of cetaceans being disturbed, harassed, and potentially colliding with commercially operating vessels. One key conservation concern is these waters coincide with designated Critical Habitat for the endangered population of Southern Resident killer whales (SRKW). Monitoring technologies such as hydrophones provide the promise of real-time animal detection and localisation. To use these continuous streams of real-time whale location data, we are developing algorithms that automate acoustic detection and classification of SRKW calls using deep learning models trained on extensive new and updated annotated datasets from the region. We are developing open-source models for pod-level classification, while also differentiating SRKW from other ecotypes and species. We are developing a forecasting system to merge these whale detections with opportunistic visual observations. This is based on sequential data assimilation methods using advanced animal movement models to estimate SRKW pod locations with probabilistic predictions of whale directional movement. These methods will be transformative for shipping by providing forecasting with a lead time of a few hours allowing vessels to adjust their speed or pathway to minimize whale-vessel interactions.

Contributed Papers

2:35

4pAB4. Tracking killer whale movements in the Alaskan Arctic relative to a loss of sea ice. Brynn Kimber (CICOES, Univ. of Washington, 7600 Sand Point Way NE, Bldg. 4, Seattle, WA 98115, brynn.kimber@noaa.gov), Jenna Harlacher, Eric Braen (CICOES, Univ. of Washington, Seattle, WA), and Catherine Berchok (AFSC/MML, NOAA, Seattle, WA)

Killer whales (*Orcinus orca)* are highly intelligent and versatile predators. Though they have a consistent presence in southern Alaskan waters, they are not known to frequently or consistently venture into the typically ice-covered U.S. Arctic, where they risk ice entrapment. However, with the trend of decreasing sea ice in the Arctic Ocean, these top predators may be venturing into waters that were previously inaccessible to them. To expand our knowledge of killer whale movements in the Arctic, eight years of data (2010–2018) from four AFSC passive acoustic moorings (funded by NOAA S&T, the U.S. Navy, and BOEM) located in the Chukchi, Beaufort, and northern Bering Seas, were manually analyzed for the presence/absence of killer whale signals to examine interannual trends in the acoustic presence of killer whales. Results from our data indicate that killer whales spend more time than previously recorded in the Arctic, as far as the edge of the Chukchi Borderland, directly following the decrease in sea ice in these areas. These results speak to a changing Arctic, both in terms of the presence of killer whales themselves and in terms of the impact that increased killer whale predation could have on Arctic food webs.

2:50

4pAB5. Call repertoire and inferred ecotype presence of killer whales (*Orcinus orca*) recorded in the southeastern Chukchi Sea. Brijonnay Madrigal (Marine Mammal Res. Program, Univ. of Hawai'i at Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, bcm2@hawaii.edu), Jessica Crance, Catherine Berchok (Marine Mammal Lab., NOAA Alaska Fisheries Sci. Ctr., Seattle, WA), and Alison Stimpert (Bioacoustics and Vertebrate Ecology Lab., Moss Landing Marine Labs., Moss Landing, CA)

Killer whales occur in the Arctic but few data exist regarding the ecotypes present. Calling behavior differs among ecotypes, which can be distinguished based on pulsed call type, call rate, and bandwidth. Data included in this study were from a passive acoustic recorder deployed 75 km off Point Hope, Alaska, in the southeastern Chukchi Sea (sampling rate 16 kHz, ~30% duty cycle). A total of 1323 killer whale pulsed calls were detected on 38 of 276 days during the summers (June–August) of 2013–2015. Most calls (n = 804, 61%) were recorded in 2013 with the majority recorded in July (76% of total calls). Calls were manually grouped into six categories: multipart, downsweep, upsweep, modulated, single modulation, and flat. Most detections were flat (n = 485, 37%) or multipart calls (n = 479, 36%), which contained both high and low frequency components. Comparisons with calls reported in the published literature showed similarities with other transient populations in both fundamental frequency contour point distribution and median frequency. This study provides the first comprehensive catalog of transient killer whale calls in this region and reports on previously undescribed calls, providing new insight into transient acoustic behavior and call diversity in the Chukchi Sea.

3:05

4pAB6. Acoustic localization of resident killer whales for source-level estimation of echolocation clicks. Jack Lawson (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr., Victoria, BC V8W2Y2, Canada, jack.lawson.1313@gmail.com), Jennifer Wladichuk (Univ. of Victoria/JASCO Appl. Sci., Victoria, BC, Canada), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Jens Koblitz (Max Planck Inst., Constance, Germany)

To study the efficacy of resident killer whale echolocation clicks for foraging and other activities requires knowledge of the acoustic characteristics of their clicks, in particular, source levels. To compute apparent source level (ASL) from recordings of killer whales in the wild requires localization of the clicking animals, ideally including rigorous uncertainty estimates, to account for transmission losses from source to receiver. This paper considers acoustic localization of killer whales based on time-of-arrival-differences for clicks recorded at a 2×2 m^2 array of 23 hydrophones. To quantify uncertainties, a Bayesian localization approach is formulated and two methods of solution are considered, one based on a linearized approximation and the other a nonlinear three-dimensional (3D) grid search. Simulations indicate significant linearization errors in 3D uncertainty (probability) distributions, confirming the superiority of the nonlinear localization. Results of this localization approach are used to calculate ASLs (with uncertainties) for echolocation clicks from Southern and Northern Resident killer whales determined to be directed approximately at the array (i.e., on-axis). Comparison of ASL values across varying ranges indicates a roughly logarithmic range dependence, consistent with whales adjusting their click levels based on distance to the target, approximately accounting for two-way spherical-spreading loss.

3:20–3:35 Break

Invited Papers

3:35

4pAB7. Echolocation click characteristics of two fish-eating killer whale populations. Jennifer Wladichuk (Univ. of Victoria/JASCO Appl. Sci., 3393 Robson Pl., Victoria, BC V9C 0J2, Canada, jwladichuk@uvic.ca), Jack Lawson, Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Jens Koblitz (Max Planck Inst., Univ. of Constance, Constance, Germany)

Acoustic disturbance has been identified as a threat to the survival of the endangered Southern Resident killer whale (SRKW) population whose critical habitat in the Salish Sea overlaps with busy commercial shipping lanes and popular recreational boating areas. Vessel noise has the potential to mask important sounds used by the whales for navigating, communicating, and foraging. To investigate the impact on foraging, acoustic characteristics of the whales' echolocation clicks must be known. We collected data on SRKW in the wild using a 2×2 m^2 23-hydrophone array deployed at short ranges (<500 m). We also acquired data on a neighboring population of Northern Resident killer whales (NRKW), whose numbers are increasing. Comparable results were obtained for most parameters between the two populations. Average peak-to-peak apparent source levels were 203 and 196 dB re 1 μPa at 1 m for SRKW and NRKW, respectively. The two populations had similar mean centroid and peak frequencies ranging between 26–29 kHz and 19–21 kHz, respectively. However, root-mean-square bandwidths differed significantly between the populations, with averages of 52 and 39 kHz for SRKWs and NRKWs, respectively. These results will be used to investigate masking potential under varying ambient noise conditions.

3:55

4pAB8. Orcasound: Open source software for catalyzing conservation of endangered Southern Resident killer whales in real-time. Scott Veirs (Beam Reach, Seattle, WA, sveirs@gmail.com) and val veirs (Beam Reach, Friday Harbor, WA)

Over the last 20 years, Orcasound has evolved from a cooperative network of real-time hydrophones to an open source software and hardware project. Orcasound connects cabled hydrophones deployed in critical habitat of the Southern Resident killer whales (SRKWs) with a web app that enables community scientists to listen and tag audio data in real-time. The live audio feeds are also monitored by automated algorithms (e.g., OrcaHello), archived in Amazon data "buckets" which are publicly accessible for free and governed by a Creative Commons license that promotes non-commercial data sharing and specifies appropriate attribution. Many of these open data have been annotated through crowd-sourcing tools built by the Orcasound community, like Pod.Cast and OrcaAL, generating open test and training data sets needed for development of more advanced classifiers. We present the latest version of Orcasound's live-listening app as a novel conservation tool, and describe our evolving architecture for: generating audio streams with low-cost hydrophones and hardware, and free software; deploying diverse algorithms, including on non-Orcasound hydrophone data streams; validating, annotating, and analyzing acoustic detections; aggregating them in an open data cooperative along with regional data from sighting networks; and issuing notifications to network members as well as conservation stewards.

Contributed Papers

4:15

4pAB9. Recent developments, observations and lessons learned in the use of real-time passive acoustic monitoring from ocean gliders in order to mitigate harm to North Atlantic Right Whales and Southern Resident Killer Whales. John E. Moloney (Eng., JASCO Appl. Sci., Ste. 202 - 32 Troop Ave., Dartmouth, NS B3B 1Z1, Canada, john.moloney@jasco.com), Katie Kowarski (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), S. B. Martin (Halifax, JASCO Appl. Sci., Halifax, NS, Canada), Braind Gaudet (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), Art Cole (Eng., JASCO Appl. Sci., Dartmouth, NS, Canada), and Emily Maxner (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada)

JASCO's novel passive acoustic monitoring (PAM) system, OceanObserver was deployed in Canada's Gulf of Saint Lawrence (GoSL) in autumn 2018 aboard a Slocum G3 glider. The purpose of this mission was to assist in the search for endangered North Atlantic Right Whales and other threatened cetaceans, and to report detections in near-real-time so that appropriate mitigation actions could be taken. This proposed presentation will provide results and lessons learned based upon the rigorous analysis and comparison of the real-time, *in situ* detections with the automated and manual detections derived from post-trial analysis of the recorded raw acoustic data. This post analysis has permitted an assessment of the performance of the automated detectors deployed at sea and the effectiveness of the real-time communication management software in getting marine mammal detections to shore-based stakeholders in a timely manner. It also has permitted the evaluation of the effectiveness of shore-based acoustic analysts tasked with the validation of the automated detections sent to shore. Whether the mission objective of "no false positives" in order to ensure the correctness of mitigation decisions will be discussed. The method of assessing performance will be described and empirical results presented. The application of this technology to SRKW conservation will be discussed.

4:30

4pAB10. Open-source deep learning models for acoustic detection and classification of orcas. Sadman Sakib (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Steven Bergner (Computing Sci., Simon Fraser Univ., Burnaby, BC, Canada), Dave Campbell (School of Mathematics and Statistics, Carleton Univ., Ottawa, ON, Canada), Mike Dowd (Dept. of Mathematics and Statistics, Dalhousie Univ., Halifax, NS, Canada), Fabio Frazao (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Ruth Joy (School of Environ. Sci., Simon Fraser Univ., Burnaby, BC, Canada), Oliver S. Kirsebom (Comput. Sci., Dalhousie Univ., 6050 University Ave., Comput. Sci., Halifax, NS B3H 4R2, Canada, oliver.kirsebom@dal.ca), Paul Nguyen Hong Duc (School of Mathematics and Statistics, Carleton Univ., Ottawa, ON, Canada), Bruno Padovese (Comput. Sci., Dalhousie Univ., Halifax, NS, Canada), Marine Randon (School of Environ. Sci., Simon Fraser Univ., Burnaby, BC, Canada), Amalis Riera Vuibert (Comput. Sci., Dalhousie Univ., Victoria, BC, Canada), Scott Veirs, val veirs (Beam Reach SPC, Seattle, WA), Jennifer Wladichuk (Univ. of Victoria/JASCO Appl. Sci., Victoria, BC, Canada), and Harald Yurk (Ecosystem Sci. Div., Fisheries and Oceans Canada, Vancouver, BC, Canada)

Current efforts to acoustically study and monitor killer whales in British Columbia (BC), including the endangered population of Southern Resident killer whales (SRKW), are hampered by the lack of sufficiently accurate sound detection and classification algorithms. Recently, several research groups have reported significant improvements in algorithm performance utilizing deep neural networks. However, for most practitioners, these novel tools remain out of reach. To bridge this gap, we are developing a set of open-source deep learning models for acoustic detection and classification of BC's killer whales. These models will be made publicly available along with expert-curated training and test sets to facilitate further development and applications. We are collaborating with Orcasound to deploy the models on their live data. We are also working together with the PAMGuard developer team to ensure that our models can be seamlessly imported and used in PAMGuard, a widely used open-source software platform for passive acoustic monitoring. In this contribution, we will provide an overview of our deep learning methodology and present preliminary results on model performance, with particular attention to the model's ability to handle diverse and variable acoustic environments and generalize to new "unseen" environments.

4:45

4pAB11. Characterizing Southern Resident killer whale calls using a particle filter for frequency contour extraction. Paul Nguyen Hong Duc (School of Mathematics and Statistics, Carleton Univ., 1125 Colonel By Dr., Ottawa, ON K1S 5B6, Canada, p.nguyenhongduc@gmail.com), Dave Campbell (School of Mathematics and Statistics, Carleton Univ., Ottawa, ON, Canada), and Ruth Joy (School of Environ. Sci., Simon Fraser Univ., Burnaby, BC, Canada)

With only 74 Southern Resident killer whales (SRKWs) remaining in the waters of the eastern Pacific, understanding these marine mammals is vital to their conservation. Invasive techniques such as focal follows or tagging whales could introduce stress to SRKWs. Instead, passive acoustic monitoring that uses a network of hydrophones one preferred tracking method, particularly in their critical habitat that overlaps the shipping lanes in the Salish Sea. The acoustic data, if processed in real-time using modern deep learning methods, can be used to detect whales and alert ships of the potential for spatial overlap to reduce risks of collision. In this project, we consider the differences in SRKW whale call types from archived recordings of each of J, K, and L pods recorded on regional hydrophones. The whale calls are extracted as functional observations in the time and frequency space from underwater recordings using a particle filter. Functional data analysis (e.g., functional clustering) are performed to characterize the whale calls and represent the variations in call types. Such statistical insights between pod-level call types should be useful for improving machine learning whale detection algorithms, and for identifying SRKW movement in a high ship traffic region of their critical habitat.

Session 4pBA

Biomedical Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, Underwater Acoustics, and Physical Acoustics: Novel Ultrasound Imaging Approaches for Muscle Characterization

Kang Kim, Cochair
Cardiology/Medicine, Bioengineering, University of Pittsburgh, 623A Scaife, 3550 Terrace St., Pittsburgh, PA 15213

Siddhartha Sikdar, Cochair
Bioengineering, George Mason University, 4400 University Drive, MS 1J7, Peterson Hall, Rm 3303, Fairfax, VA 22030

Chair's Introduction—1:30

Invited Papers

1:35

4pBA1. Muscle elastography: Stress versus stiffness. Joseph Crutison, Dieter Klatt (Univ. of Illinois Chicago, Chicago, IL), Thomas G. Sandercock (Northwestern Univ., Chicago, IL), Eric J. Perreault (Northwestern Univ., Evanston, IL), and Thomas Royston (Univ. of Illinois Chicago, 851 S. Morgan St., Rm. 218 SEO (MC 063), Chicago, IL 60607, troyston@uic.edu)

Dynamic elastography imaging, whether based on magnetic resonance, ultrasound or optical modalities, attempts to reconstruct quantitative maps of the viscoelastic properties of biological tissue, properties that are altered by disease and injury, as well as response to therapy. Reconstruction often assumes isotropy and homogeneity and neglects the effects of nonhomogeneous boundary conditions. Skeletal muscle violates all these assumptions, posing challenges and opportunities for developing better imaging biomarkers. Muscle activation and varying tensile loads can influence measurement interpretation. For bulk shear waves in an isotropic viscoelastic material not influenced by boundary conditions the measured wavelength correlates with the square root of the tissue's shear elastic modulus and viscosity, important rheological parameters. But, under significant tensile loading the measured transverse wavelength will also be affected by the induced quasistatic stress field, making identification of the muscle's inherent rheological properties seem impossible. Acoustoelastic theory, previous studies addressing this issue specifically for muscle, as well as new studies by our group are reviewed, identifying remaining challenges, potential strategies and opportunities for a more comprehensive understanding of both muscle structure and function based on noninvasive measurements. [Support acknowledged: NSF #1852691, NIH #AR071162.]

1:55

4pBA2. Role of anisotropy in elastographic tissue characterization. Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN 55905, urban.matthew@mayo.edu), Maureen Pelissero (Winona State Univ., Winona, MN), and Hsiao-Chuan Liu (Dept. of Radiology, Mayo Clinic, Rochester, MN)

The structure and function of soft tissues are intimately connected. Some soft tissues are organized in such a way that they have material properties that are anisotropic, that is, the measured properties depend on which direction they are measured. Some such tissues are the brain, kidney, arteries, and cardiac and skeletal muscle. We will provide a brief overview of work targeted at characterization of the anisotropic mechanical properties in kidney and cardiac and skeletal muscles. In many cases, these tissues can be assumed to be transversely isotropic, which allows for simplification in estimating their mechanical properties. The use of propagating shear waves can be used to determine these mechanical properties. As a comparison example, we will show data from characterization of renal tissue with ultrasound shear wave elastography (SWE) and optical coherence elastography. We will also provide an overview of evaluation of skeletal muscle mechanical properties using shear wave elastography techniques and use of model fitting to estimate mechanical properties. Special care must be taken in evaluation of such anisotropic tissues to ensure that measurements are standardized across practices and studies.

2:15

4pBA3. Shear wave elastography for muscle biomechanics. Jean-Luc Gennisson (Université Paris Saclay - CNRS, BioMaps, 4 Pl. du Général Leclerc, Orsay 91401, France, jean-luc.gennisson@universite-paris-saclay.FR)

In vivo muscle biomechanical properties are classically deduced from inverse dynamics or measures of joint torque performed using ergometers. However, this provide information about the combined behavior of several structures. Thus, it is complicated to isolate the behavior of an individual muscle. Ultrasound elastography techniques have been developed with the aim to non-invasively assess localized stiffness. In this presentation we focused on the supersonic shear imaging technique that allows to recover stiffness by looking at the propagation of shear waves. In the case of isotropic, homogeneous and quasi-incompressible biological tissues, the propagation velocity is directly linked to the shear elastic modulus. However, this link is not so obvious in muscles due to their anisotropy. Here, we present how stiffness as well as viscosity can be estimated in fusiform muscles but also in pennate muscles by using specific ultrasound sequences that allows to recover directly anisotropy in a 2D imaging plane. Finally, we show how it is possible to estimate nonlinear elastic properties of muscle by deriving the acoustoelasticity theory in a transverse isotropic media. Our hope is to combine all of these mechanical parameters to understand the behavior of a single muscle and deduce its force during movement.

2:35

4pBA4. Noninvasive direct assessment of the skeletal muscle function during neuromuscular electrical stimulation. Kang Kim (Cardiology/Medicine, Bioeng., Univ. of Pittsburgh, 623A Scaife, 3550 Terrace St., Pittsburgh, PA 15213, kangkim@upmc.edu), Zhiyu Sheng (Medicine, Univ. of Pittsburgh, Pittsburgh, PA), and Nitin Sharma (Biomedical Eng. Dept., North Carolina State Univ., Chapel Hill, NC)

Neuromuscular electrical stimulation (NMES) artificially activates skeletal muscle contraction by sending electrical pulses either to nerves or muscle belly. NMES has been used in various applications in rehabilitation and sports medicine. However, muscle function assessment has been limited to electrical activity measurements by an electromyography, indirect mechanical measurements using a force sensor, or ultrasound imaging that only provides anatomy and structure. In this paper, dynamic contractility of the targeted muscle in response to the NMES stimulation on human subjects is assessed using ultrafast plane wave imaging and speckle tracking. The change of the strain in the targeted muscle due to the continued NMES, resulting in muscle fatigue, was continually measured. By correlating the transient muscle strain with the force measurement as the direct gait function, it has been shown that muscle strain as measure of muscle contractility obtained by ultrasound elastography can be used as surrogate biomarker of muscle fatigue due to NMES. This technology can be used in several applications including NMES integrated exoskeletal for those who lost gait function as feedback to the motion control and NMES dose and monitoring the muscle strength and functional gain by NMES induced muscle treatment and training.

2:55

4pBA5. Viscoelastic response ultrasound for interrogating dystrophic muscle degeneration. Caterina M. Gallippi (Joint Dept. of Biomedical Eng., The Univ. of North Carolina at Chapel Hill, 10010 Mary Ellen Jones Bldg., CB 7575, Chapel Hill, NC 27599-7575, cmgallip@email.unc.edu), Joseph B. Richardson (Elec. and Comput. Eng., North Carolina State Univ., Chapel Hill, NC), Christopher J. Moore (SonoVol, Inc., Durham, NC), and James F. Howard, Jr. (Neurology, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Dystrophic muscles undergo repeating cycles of inflammation, necrosis, fibrosis, and fatty deposition, with associated changes in viscoelastic property. Thus, a noninvasive approach to monitoring muscle viscoelasticity is relevant to monitoring disease progression and responses to therapy. Viscoelastic Response (VisR) ultrasound is an acoustic radiation force (ARF) imaging method that executes consecutive, colocated ARF excitations and evaluates the induced on-axis tissue displacement to decipher elastic and viscous properties. Further, by rotating an asymmetrically shaped ARF excitation beam to evaluate elasticity and viscosity along and across the axis of symmetry in transversely isotropic materials, such as muscle, VisR interrogates viscoelastic anisotropy. Finally, machine learning models have been developed to enable quantitative VisR assessments of elastic and viscous moduli. VisR has been demonstrated *in vivo* for monitoring dystrophic muscle degeneration in boys with Duchenne muscular dystrophy (DMD). Relative to age-matched controls, boys with DMD had higher VisR-derived elasticity and viscosity in early disease stages, when inflammation is present, and higher viscosity in late disease stages after cycles of fatty deposition. Boys with DMD also had lower VisR-derived longitudinal than transverse elastic and viscous moduli, while the opposite was true for controls. VisR is a promising approach to monitoring dystrophic muscle degeneration in DMD.

3:15–3:30 Break

3:30

4pBA6. Using ultrasound to measure Achilles Tendon kinematics, kinetics, and material properties when walking. Samuel A. Acuña (Bioengineering, George Mason Univ., 4400 University Dr., Ste. 3100, Fairfax, VA 22030, sacuna2@gmu.edu)

Ultrasound has emerged as a fundamental tool to investigate the biomechanics of muscles and tendons during locomotion. Ultrasound can directly and non-invasively assess neuromuscular activity *in vivo* and can thus inform researchers how muscles and tendons interact during a wide variety of functional tasks such as walking. For example, loading from the triceps surae muscles (i.e., calf muscles) interacts through distinct fascicles within the Achilles tendon, which can give rise to unique biomechanical behavior detectable via ultrasound. This presentation will discuss how we have used ultrasound to monitor kinematics and material properties of the Achilles tendon during walking, as well as motivate investigations of tendon kinetics using wave speed. This work has broad application to clinical gait analyses as well as to further our understanding of the neuromuscular activity underlying healthy gait. For example, we will investigate the hypothesis that age-related changes in the Achilles Tendon can influence propulsive forces in older adults. Finally, we will discuss the broad opportunities and challenges of using ultrasound during clinical gait analysis.

3:50

4pBA7. New applications of ultrasound-based muscle activity sensing for rehabilitation engineering. Siddhartha Sikdar (Bioengineering, George Mason Univ., 4400 University Dr., MS 1J7, Peterson Hall, Rm 3303, Fairfax, VA 22030, ssikdar@gmu.edu), Ahmed Bashatah, Joseph Majdi (George Mason Univ., Fairfax, VA), and Parag V. Chitnis (Bioengineering, George Mason Univ., Fairfax, VA)

Traditionally, electromyography using surface electrodes has been the dominant method for sensing muscle activity. Major challenges with surface electromyography include the difficulty in resolving signals from overlying muscles and low signal to noise ratio. In recent years, ultrasound has been investigated as an alternative and complementary modality for sensing functional muscle activity that overcomes several limitations of surface electromyography. Ultrasound imaging can non-invasively resolve individual muscles, including deep and overlying muscles, and detect dynamic activity within different functional compartments in real-time. While the use of ultrasound in the biomechanics community has a long history, the continuing miniaturization of ultrasound instrumentation has opened up new opportunities for using ultrasound in rehabilitation engineering as a biosignal sensing modality for assistive devices. This talk will describe results from our current work on developing miniaturized low-power ultrasound instrumentation for wearable systems, including next generation prostheses and exoskeletons. I will also describe opportunities and limitations for applications of ultrasound in rehabilitation engineering.

Contributed Papers

4:10

4pBA8. Phase velocity imaging using acoustic radiation force-induced shear wave with a multi-frequency excitation pulse. Md Murad Hossain (Biomedical Eng., Columbia Univ., 630 West 168 St., P&S 19-418, New York, NY 10032, mh4051@columbia.edu) and Elisa Konofagou (Biomedical Eng., Columbia Univ., New York, NY)

The viscoelastic properties of tissues are often assessed by analyzing the phase velocity dispersion curve computed using a 2-D Fourier transform on the single fixed duration acoustic radiation force excitation pulse (FD-ARFEP)-induced shear wave with broadband frequencies. This study proposes a multi-frequency ARFEP (MF-ARFEP) using a single imaging transducer to generate shear waves at target frequencies of 100:100:1000 Hz to reduce noise in the phase velocity images. The MF-ARFEP is generated by summing sinusoids with target frequencies and then sampled to collect tracking pulses in-between the sampled MF-ARFEP. The MF-ARFEP is validated by imaging 6 and 70 kPa inclusions in a 18 kPa background, *ex vivo* canine liver, and *in vivo* 4T1 breast cancer mouse tumor with comparison to the FD-ARFEP. The median phase velocity was similar between MF-ARFEP versus FD-ARFEP. However, the CNR was on an average 1.3 times higher in MF-ARFEP versus FD-ARFEP -derived images of inclusions and the coefficient of variation was on an average 1.1 and 2.2 times lower in MF-ARFEP versus FD-ARFEP-derived images of tumor and liver. These results demonstrate the feasibility of the MF-ARFEP excitation pulse to generate phase velocity images at 100:100:1000 Hz frequencies with reduced noise compared to the FD-ARFEP. Ongoing studies entail the translation of this new viscoelasticity imaging method for the characterization of breast cancer tumors in patients.

4:25

4pBA9. A miniature ultrasound M-Mode imaging system based on time delay spectrometry. Ahmed Bashatah (Bioengineering, George Mason Univ., Fairfax, VA, abashat2@gmu.edu), Paul Otto (Bioengineering, George Mason Univ., Fairfax, VA), Biswarup Mukherjee (Ctr. for Biomedical Eng., IIT Dehli, Delhi, India), Parag V. Chitnis, and Siddhartha Sikdar (Bioengineering, George Mason Univ., Fairfax, VA)

For wearable ultrasound applications, power and safety are a concern. We have developed an imaging method based on time-delay spectrometry (TDS) that employs a low-voltage, wideband chirp to establish a relationship between reflector depth and frequency by mixing the transmit and receive signals. We will show a low power, 4 channel ultrasound M-Mode system capable of tracking *in vivo* muscle interfaces in the forearm. The battery powered imaging system consists of power regulation subsystem, a RF chirp subsystem, and four channels consisting of a transmit RF amplifier, a mixing unit, and a receive amplifier which are sampled at 40 KSPS by a processor with a four channel ADC. The processor performs an FFT on each demodulated return to get 4 scanlines that are tracked over time to produce M-Mode images that are sent to a host computer. The wearable system can provide M-Modes images of up to 4 cm in forearm at a framerate of 50 Hz for $\sim$7 h with a total power consumption of 1.92 W. We demonstrate the ability of a miniaturized TDS ultrasound imaging system to collect *in-vivo* M-Modes that can be used for long term monitoring. The system can be used for wearable applications due to its low power consumption. [United States Department of Defense, Congressionally Directed Medical Research Program, award number W81XWH-16-1-0722 (PI: S Sikdar). National Institute of Biomedical Imaging and Bioengineering of the National Institutes of Health under Award No. U01EB027601.]

4:40

4pBA10. Quantification of forearm muscle activation by ultrasound imaging and high-density electromyography. Zhiyu Sheng (Medicine, Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 624, Pittsburgh, PA 15213, zhs41@pitt.edu), Ernesto Bedoy (Ctr. of Neurosci., Univ. of Pittsburgh, Pittsburgh, PA), Douglas J. Weber (Mech. Eng. and the Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), Brad E. Dicianno (Physical Medicine and Rehabilitation, Univ. of Pittsburgh, Pittsburgh, PA), and Kang Kim (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Assessment of skeletal muscle function is an active research area in rehabilitation and sports medicine. An effective assessment methodology can be used to evaluate muscle recovery or response to treatments or interventions, and for early diagnosis of skeletal muscle diseases when combined with corresponding biomarkers. The main objective of this study is to develop an engineering technique for detecting muscle activation by combined ultrasound imaging and high-density electromyography (HDEMG). An ultrasound speckle tracking algorithm was employed to measure the impulsive kinetic energy released by muscle contraction while the processed HDEMG depicted the spatial-temporal electrical response of the activated muscle. The location and time course of the muscle activation can then be quantified by detecting the precise onset of the mechanical and electrical signals from ultrasound imaging and HDEMG, respectively. To demonstrate the conceptual feasibility, experiments were performed, with synchronized dual-probe ultrasound imaging and HDEMG recorded, on the forearm of one human participant without an injury or disability. To test detections of different activation patterns, single electrical pulses were sent to the flexor carpi radialis (FCR), flexor digitorum superficialis (FDS) and the median nerve, respectively, to intentionally create cases where FCR and FDS can contract individually or simultaneously.

4:55

4pBA11. Externally induced shear waves in the right ventricular free wall during the cardiac cycle. Luxi Wei (Erasmus MC, Doctor Molewaterplein 40, EE23, Rotterdam 3015 GD, The Netherlands, l.wei@erasmusmc.nl), Rahi Alipour Symakani (Erasmus MC, Rotterdam, The Netherlands), Annette Caenen (Ghent Univ., Ghent, Belgium), Lana B. H. Keijzer, Daphne Merkus, Beatrijs Bartelds, Yannick Taverne, Antonius F. van der Steen, Hendrik Vos, and Mihai Strachinaru (Erasmus MC, Rotterdam, The Netherlands)

Increased right ventricular (RV) diastolic stiffness is linked to adverse prognosis in pulmonary arterial hypertension. Currently, a non-invasive

method for measuring RV stiffness is lacking. In the left ventricle, shear wave (SW) speed is related to diastolic stiffness. We show for the first time that SWs in the RV free wall (RVFW) can be induced and imaged transthoracically with ultrasound. SW imaging was performed using an L7-4 array (ATL, USA) with a Vantage 256 system (Verasonics, USA) in a 5-weeks-old Yorkshire-Landrace pig ($\pm$120 beats/min). SWs were induced by a push beam ($f_0 = 4.5$MHz, push duration $= 800\mu$s) on the RVFW and its propagation was imaged ($f_0 = 5.2$MHz) using plane wave compounding (-12 deg, 0 deg, +12 deg) at 3 kHz framerate. Three acquisitions of 1 s were performed, where fourteen SWs were sequentially induced during each acquisition. SW propagation speeds were calculated using a semi-automatic pipeline that includes tissue velocity estimation along a manually traced spline on the RVFW, and Radon transform to estimate SW speed. At least 85% of the waves were tracked successfully for all acquisitions. The SW velocities varied during the heart cycle as expected, ranging from 0.58 ± 0.09 m/s at end-diastole to 1.8 ± 0.2 m/s during systole. Pathological increase in stiffness may be demonstrated in longitudinal case/control studies.

5:10

4pBA12. Ultrasound measurement of fiber orientation and tissue strain in excised myocardium under biaxial tension. John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA 15261, jmc345@pitt.edu), Marc A. Simon (Div. of Cardiology, Dept. of Medicine, Univ. of California San Francisco, San Francisco, CA), and Kang Kim (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Biaxial mechanical testing is a useful tool for determining the mechanical properties of myocardium tissues, which are anisotropic due to their layered and fibrous microstructure. Mechanical properties of the tissue are largely determined by those of the laminar myofiber sheets, whose orientations vary in a helix throughout the tissue thickness. Fiber architecture of tested tissues is usually determined *post hoc* by destructive histological sectioning, and the contribution of fiber kinematics to tissue-level mechanical behavior is accomplished by assuming a continuum model for the myocardium. In this work is presented simultaneous ultrasound-based measurement of fiber orientation and tissue strain in excised myocardium under biaxial tension. Two-protocol (1:1 and 1:6) testing is employed in order to probe tissue anisotropy and fiber kinematics. An ultrasound linear array (15 MHz) is scanned by rotation about the imaging axis to obtain three-dimensional backscatter data from the tissue during quasi-static biaxial deformation, from which transmural fiber orientation and tissue strain are determined by backscatter tensor imaging and 3D speckle tracking, respectively. Results illustrate feasibility of the simultaneous measurement and future utility for studying fiber mechanics in diseased tissues. Fiber rotation and elongation measured directly from ultrasound is compared to that calculated under the continuum assumption.

Session 4pED

Education in Acoustics: Preview of Next JASA Special Issue on Education in Acoustics

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Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802

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Chair's Introduction—1:30

Invited Papers

1:35

4pED1. Acoustics and vibrations as an integral part of the engineering curriculum at the University of Hartford. Christopher M. Jasinski (Mech., Aerosp., and Acoust. Eng. Dept., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, jasinski@hartford.edu) and Robert Celmer (Mech., Aerosp., and Acoust. Eng. Dept., Univ. of Hartford, West Hartford, CT)

The University of Hartford is home to two unique undergraduate engineering majors in acoustics, both sharing a core course layout of acoustics, vibrations, and projects. These include Vibrations 1/2, Fundamentals of Architectural and Musical Acoustics, Acoustics by Design (Project), Advanced Acoustical Topics, and Acoustic Capstone Design Project. The B.S.M.E. with Acoustics Concentration and the B.S.E. in Acoustical Engineering and Music programs allow for two different tracks within the acoustics field, providing cohesive plans of study on many facets of listening and design. Additionally, all Mechanical Engineering majors are required to take Vibrations 1, and a course in Engineering and Environmental Acoustics. The department philosophy for this inclusion is that acoustics and vibration considerations are an essential component for the development of the complete mechanical engineer. Examples of the wide range of career paths that program graduates pursue will be addressed. This presentation will discuss the details and history of the acoustics program, components of the Vibrations I and Engineering and Environmental Acoustics courses, example research and design projects based on work in these courses, and a review of educational goals and outcomes, along with pedagogical development of a unique Vibrations Concept Inventory for assessment.

1:55

4pED2. Teaching acoustics through entertainment: Why and how? Olivier Robin (Faculté de Génie - Dept. Génie Mécanique, Université de Sherbrooke, 2500, bd de l'université, Sherbrooke, QC J1H1L2, Canada, olivier.robin@usherbrooke.ca) and Luc Jaouen (Matelys, Vaulx-en-Velin, France)

In our modern and highly connected world, we are fed with large and growing amounts of information. As an example, the yearly number of published scientific articles exceeded the astounding number of 1 500 000 papers.in 2015. More and more time is spent on numerical media or supports, with texts that tend to be scanned rather than read. The way we acquire knowledge is also deeply modified by these changes. In this communication, some alternatives to classical acoustics and vibration teaching materials are discussed, with corresponding perspectives for an effective information transmission. Among other examples, the presented works include video games, comics and reactive web documents that can be all used for educational purposes with entertainment as a keyword.

2:15

4pED3. Knocking over LEGO minifigures with time reversal focused vibrations: Understanding the physics and developing a museum demonstration. Brian E. Anderson (Dept. of Phys. & Astron.,Brigham Young Univ., N245 ESC, Provo, UT 84602, bea@byu.edu), Lucas A. Barnes (Phys. & Astronomy, Brigham Young Univ., Provo, UT), Pierre-Yves Le Bas (Detonator Technol. (Q-6), Los Alamos National Lab., Los Alamos, NM), Adam D. Kingsley, Aaron C. Brown (Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Henrik R. Thomsen (Dept. of Earth Sci., ETH Zürich, Zurich, Switzerland)

Time reversal (TR) of impulse responses can be used to focus wave energy, even in a reverberant environment. A demonstration of TR focusing was developed to knock over a targeted LEGO minifigure, standing among other minifigures (who remain standing), by focusing plate vibrations under the target minifigure\rquote s feet. In order to achieve a high degree of repeatability, the physics of the demonstration needed to be understood. A laser Doppler vibrometer was used to compare the motion of the minifigure and the plate directly beneath it. Speaker shakers are used to generate motion of the plate. If their induced vibration was too large the minifigure loses contact with the plate prematurely, causing it to be in the air when the main focused vibrations arrive. If the minifigure is in contact with

4p THU. PM

the plate when the focused vibrations arrive then the minifigure is launched into the air before toppling on its side. This demonstration was optimized to be a two-player-game museum exhibit in a wave propagation museum hosted by ETH Zurich in Switzerland. This demonstration illustrates the power of focused vibrations.

2:35

4pED4. Comics and acoustics education: A drawn example around the decibel. Olivier Robin (Faculté de Génie - Dpt Génie Mécanique, Université de Sherbrooke, 2500, bd de l'université, Sherbrooke, QC J1H1L2, Canada, olivier.robin@usherbrooke.ca)

While having been the object of various criticisms and considered to be a childish litterature for a long time, comics are now a legitimate art form, the ninth art. Comics are also considered as a fully fledged tool for disseminating and explaining science. More and more researchers advocate for a larger use of comics in science, and creating comics is generally found to have a positive effect on students. This presentation will first briefly recall main facts concerning comics, and their use in dissemination and education. The genesis of an educational "comics paper" explaining the origin and history of the decibel, together with examples from acoustics and other domains will be then thoroughly described. This "comics paper" will be put into perspective with its companion "classical paper," in order to appreciate the transition from an almost exclusively written form to a predominantly drawn form.

2:55–3:10 Break

3:10

4pED5. Creating a mentored research environment in an underwater acoustics lab. Cameron T. Vongsawad (Phys. and Astronomy, Brigham Young Univ., 820 W. 555 N, Orem, UT 84057, cvongsawad@byu.edu), Tracianne B. Neilsen, Kaylyn N. Terry, Scott P. Hollingsworth, Corey E. Dobbs, and Gabriel H. Fronk (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Mentored research experiences for undergraduate students yield an increased interest in STEM fields, are better prepared for the professional world, and develop enhanced cognitive and personal skills. These experiences help them become better students with improved ability for independent thought and problem solving who are more likely to pursue graduate degrees. The new Brigham Young University underwater acoustics laboratory has been built to facilitate active mentoring undergraduate and graduate students in order promote quick involvement in original research. New students require significant scaffolding in order to meet the steep learning curves in many topics and skills. This scaffolding includes regular group training, peer mentorship, and active laboratory resources. Advisors facilitate this through efficient laboratory design, structured procedures, and regular communication. This process supports student-led research and mentorship by promoting effective transfer of knowledge and allowing students to develop into independently driven research scientists with the technical and professional skills to be successful in a professional world.

3:30

4pED6. One problem, two solution approaches: Using the impedance translation theorem and equivalent circuits for graduate level homework. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N245 ESC, Provo, UT 84602, bea@byu.edu) and Scott D. Sommerfeldt (Brigham Young Univ., Provo, UT)

Research applications such as the study of musical instruments, human vocal tract effects on speech, automotive exhaust pipes, HVAC ducts, and others often utilize one-dimensional models of the acoustic systems to determine the resonance behavior and sound radiation properties. The impedance translation theorem and the equivalent circuits method both include all the physics associated with these one-dimensional systems, although one approach or the other may be more appropriate or straightforward to use depending on the result desired. Both these methods are taught at Brigham Young University in two graduate level courses. One of the most challenging and memorable homework assignments from each of these courses is based on using these techniques to analyze a particular acoustic system and compare its response with the real thing. This paper will overview these two techniques and outline how they are applied in homework assignments for these two courses to analyze phonemes produced by altering the human vocal tract.

3:50

4pED7. Using Impedance to explore resonance in mechanical, acoustical, and electrical analog systems. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

Resonance is usually defined as the frequency at which a measured quantity reaches a maximum value with any change in frequency resulting in a decrease in the amplitude of the measured quantity. Sometimes, but not always, this resonance frequency corresponds to the undamped natural frequency, ω_o, of the system. This paper explores the responses of analogous mechanical, acoustic, and electrical oscillating systems, comparing the typical measured response curves (displacement, pressure, and current, respectively) used to showcase the concept of resonance for each. Driving point impedance (real, imaginary, magnitude, and phase) is used to compare the three types of systems. Frequency dependence of the input mechanical impedance (measured with an impedance head) for three mechanical systems (damped mass-spring, tuning fork, wineglass) reveals differences depending on where the system is driven. Measurements of the input acoustic impedance (measured with a microflown p-u probe) for a Helmholtz resonator allows for extraction of the undamped natural frequency in the presence of damping which cannot otherwise be ignored. The electrical impedance of series and parallel RLC circuits and a loudspeaker illustrate differences in the definition of "resonance" when compared with analogous mechanical and acoustic systems.

4:10

4pED8. An overview of the upcoming Special Issue on Education in Acoustics. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA) and Preston S. Wilson (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

The current special session includes presentations by authors who have submitted (or will be submitting) papers to an upcoming special issue of The Journal of the Acoustical Society of America on education in acoustics. Quoting from the call for papers for the special issue: "The scope of this Special Issue would include all types of papers that would be of interest to people who teach courses in acoustics (at all levels), or who might write textbooks on acoustics (at all levels), or to people who enjoy learning and relearning the basics of acoustics." This talk will provide an overview of the special issue and highlight some of the issue's content not represented by speakers in the current special session.

THURSDAY AFTERNOON, 2 DECEMBER 2021 — 402 (L)/403 (O), 1:00 P.M. TO 4:55 P.M.

Session 4pNS

Noise, Architectural Acoustics, ASA Committee on Standards, Psychological and Physiological Acoustics, and Education in Acoustics: Sound Walks

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, #121, Pompano Beach, FL 33062

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, MS 38655

Chair's Introduction—1:00

Contributed Papers

1:05

4pNS1. Development of a multi-level predictive soundscape model to assess the soundscapes of public spaces during the COVID-19 lockdowns. Andrew Mitchell (Inst. for Environ. Design and Eng., Univ. College London, Central House, 14 Upper Woburn, London WC1H 0NN, United Kingdom, andrew.mitchell.18@ucl.ac.uk), Tin Oberman, Francesco Aletta, and Jian Kang (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom)

The recent developments in the standardization of soundscape as both a research and engineering field have highlighted the need for models which can predict likely soundscape assessment from objective measurements. Such a need was highlighted during the COVID-19 lockdowns. The unprecedented restrictions in human activity presented a unique opportunity to investigate the urban noise impacts of drastic reductions in traffic noise and human sounds, but simultaneously made it impossible to carry out standard methods of soundscape assessment (i.e., in-person surveys or soundwalks). To address this, a multi-level linear regression model was developed based on an existing database of soundscape surveys and binaural recordings to predict how the soundscapes of 13 locations in London and Venice would have likely been perceived during the lockdowns based on objective measurements. To build this model, a feature selection process was applied to an extended suite of psychoacoustic metrics and a variable characterising the context of each location to identify a minimum set of input features and model structure. This presentation will demonstrate the development of this model, its application in the COVID case study and corresponding results, and will discuss the potential for future applications of a similar predictive soundscape modelling framework.

1:20

4pNS2. Sound of the city: Creating a balanced sound composition in urban green spaces. Lauren Gray (A. Morton Thomas and Assoc., 4291 Fieldhouse Dr., College Park, MD 20742, lreedgray@gmail.com)

Sound in the landscape is an important and often-ignored aspect of the human experience. In urban landscapes, seemingly cacophonous sounds can create a symphony, combining the beloved sounds of nature and humans with the often less desirable, but no less important, sounds of traffic and sirens. The work of this thesis puts that symphony, and its relationship to the landscape, under a microscope. Investigations into the theories of composers John Cage and R. Murray Schafer led to the creation of a new design theory and a new methodology for surveying sound. The theory and method were applied to the re-making of an urban soundscape, allowing further exploration into the impact of sound on the perception of place and a close examination of the conscious, subconscious, beautiful, and necessary in the design of landscape.

4p THU. PM

Invited Papers

1:35

4pNS3. Soundwalk overview and its application to design. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and David s. Woolworth (Roland, Woolworth & Assoc., Oxford, MS)

How an acoustical environment is perceived by the occupants is a key element in the analysis and successful design of that environment. The goal of such a design effort would be to improve the perception of the acoustical environment and so to enhance the quality of life for the residents, workers and visitors in a locale. The soundscape technique is a powerful method for determining how an environment is perceived under existing conditions, and for developing positive design and planning outcomes. An important and useful tool in the soundscape toolbox is the soundwalk. The protocols for setting up a soundwalk investigation, and the detailed practices which may be followed are described, with reference to the perception data collection procedures as outlined in the ISO 12913 Standard series. Examples of soundwalk data collection are presented. The application of the soundwalk method in the design process is encouraged through case studies presented in this session. Attendees are invited to participate in a live soundwalk conducted at the conclusion of the session.

1:55

4pNS4. Soundwalks in a botanical garden as a part of the planning process. Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein, Marylin Roa, Matthew Vetterick, Jennifer Miller, Hyun Paek, and Ian Jones (Siebein Assoc.,Inc., Gainesville, FL)

Soundwalks were taken over a one-year period of time in an existing botanical garden and in the community surrounding the garden as part of the process for a planned expansion of the garden. Soundwalks were taken so that sounds propagating from special events and typical activities in the gardens could be experienced from the perspectives of guests at the gardens and residents who live nearby. Special events included a special Christmas celebration with music playing and large exhibits that lasted for several weeks as well as mock-ups of weddings, meetings and dinners at several locations where these would normally occur. A house sound system was designed to include monitoring of sound levels and controls over processors so that agreed upon sound levels could not be exceeded. Full scale aural mock-ups of sounds propagating from special events were conducted so that residents could sit on their patios and listen to sounds propagating from the gardens and then walk into the gardens to listen to the sound levels played in the event areas also. This process allowed a dialogue to begin between residents, planners and the gardens to develop sound control strategies for the planned expansion of the facilities.

2:15

4pNS5. Soundwalks in new and existing intensive care units. Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Jennifer Miller, Matthew Vetterick, and Ian Jones (Siebein Assoc.,Inc., Gainesville, FL)

Soundwalks were taken in 3 existing ICU's in a hospital. New ICU's were designed to address acoustical and other issues identified in the 3 existing ICU's. Sound walks were taken in the 3 new ICU's after they were constructed. While sounds of individual acoustical events were similar in the new and existing ICU's, the overall sound levels and frequency of sounds decreased in the new ICU's compared to the existing ICU's. In other words, the sounds of people speaking were measured similarly in both ICU's as were the sounds of doors closing, the ice machine operating and other specific acoustical events. However, the design of the new ICU removed the sounds of nurses speaking at the nurses' station from proximity to patient rooms so the sound levels of people speaking who were not communicating directly with the patient or family were reduced in the new ICU. Additionally, the sounds of medical equipment were largely removed from the patient rooms in the new ICU. Equipment such as ice machines, coffee pots and other items were also removed from patient areas in the new ICU. Overall, the new ICU had a quieter environment for patients, nurses and visitors.

2:35

4pNS6. The soundwalk is a tool. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

Soundwalk protocols and procedures are formally established in ISO Standard 12913; the intention is a general reproducibility of results and to be able to compare soundwalk data for different walks, and even to bring the experience to the lab where it can be further examined. In general standards themselves require a reasonable level of technical expertise to execute, may require resources not available to laypeople, or may not align with the needs of a project. This paper will examine the soundwalk as a tool adaptable for given purposes and draw from soundwalks to entertain some of the questions related to the human elements such as perception, preconceived notions on sound, and repeated walks. The use of soundwalks to develop community sound ordinances will also be discussed.

Contributed Papers

2:55

4pNS7. Urban soundscape complexity. Dick Botteldooren (Information Technol., Ghent Univ., Technologiepark 126, Technologiepark 126, Gent 9052, Belgium, dick.botteldooren@ugent.be)

The soundscape approach has become common practice in urban sound design. ISO12913 provides methods for assessing it within context. One of the proposed approaches is grounded in the affect that the sound environment may trigger. In addition we have proposed that in everyday situations—that is when persons are not instructed to listen to the sound environment—saliency and attention should be considered and explain the existence of non-influential sound environments. However, the underlying mechanism causing a soundscape to be calming, stimulating, or disruptive are not completely explained by classical psycho-acoustic. We argue that complexity could be the missing factor. Complexity at the sensorial level has been suggested many years ago and it is now common in bio-acoustics. However, semantic complexity determined by the understanding of the interplay between sounds, has seldomly been considered. Yet, in an urban context, several auditory streams will likely co-exist. This conceptual idea is complemented by a measurement method relying on sound recognition by artificial intelligence to quantify the variation in likelihood over time of multiple sounds. Complexity indicators such as entropy and spectral slope can be calculated on this likelihood trend. It can be shown that they are related to effects of environmental sound.

3:10–3:25 Break

Session 4pPA

Physical Acoustics and Biomedical Acoustics: Session in Honor of David T. Blackstock II

Mark F. Hamilton, Cochair
Walker Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1591

Thomas G. Muir, Cochair
Applied Research Labs, University of Texas, P.O. Box 8029, Austin, TX 78713

Chair's Introduction—1:30

Invited Papers

1:35

4pPA1. David T. Blackstock—An international perspective. Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

The international acoustics community looks back with deep appreciation on the great lifetime achievement of David T. Blackstock. Always interested in international scientific exchange, he was especially involved in the International Commission for Acoustics (ICA) since the 1980s. Without his contributions to understanding across the Iron Curtain, we would not have such harmonious cooperation in acoustics today. This applies not only to the open and trusting cooperation at international conferences, but also to the establishment of firmly rooted support for individual researchers and students who have to work under difficult conditions. This paper looks at David Blackstock's involvement in the ICA, and describes the structures of international collaboration that he was instrumental in initiating.

1:55

4pPA2. David T. Blackstock and kidney stone lithotripsy. James A. McAteer (Dept. of Anatomy, Cell Biology, and Physiol., Indiana Univ. School of Medicine, Indianapolis, IN), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), and Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Bellievue, WA)

For the past 25 years, David Blackstock served as External Advisor to an NIH Program Project Grant (PPG) dedicated to improving shock wave lithotripsy (SWL) of kidney stones. Invented in the 1980s, SWL revolutionized stone management, eliminating the need for open surgery. As a truly noninvasive surgery, SWL remains the flagship of medical acoustic therapies. Study of its engineering, bioeffects, and physical mechanisms of action helped make SWL safer and more effective and led to new modalities of diagnostic and therapeutic ultrasound. Dr. Blackstock's role in this effort cannot be overstated. He was a regular participant in our series of in-depth, two-day, semiannual meetings where in addition to reviewing new data related to his areas of expertise in shock wave propagation and nonlinear acoustics he absorbed, dissected, and eloquently critiqued research in the broad array of fields (e.g., clinical urology, renal vascular physiology and pathology, fracture mechanics, and more) otherwise unrelated to his experience. Dr. Blackstock had the ability to learn on the spot and then add incisive, valuable suggestions based on his expertise, intuition, and critical thinking. He kept us focused. He elevated our thinking. He stimulated our creativity and encouraged us toward new ideas. Work supported by NIH-P01-DK043881.

2:15

4pPA3. David Blackstock: Distortion of shock waves and breaking of kidney stones. Robin O. Cleveland (Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, robin.cleveland@eng.ox.ac.uk)

This talk will discuss how the seminal work that David Blackstock carried out in nonlinear acoustics for underwater and atmospheric applications contributed to the understanding of the medical use of shock waves for breaking of kidney stones. David's expertise lead to him acting as an external advisor for two decades to a NIH funded programme in shock wave lithotripsy; that contribution will be covered elsewhere in the session. I will present the weak shock theory that describes how nonlinear pulses distort when undergoing geometrical spreading. The theory will then be applied to the first clinically approved lithotripter, that employed a spark to generate a shock wave and an ellipsoidal reflector to focus that into the body. Inside the reflector the shock wave undergoes spherical spreading, reflection and then spherical convergence. It will be shown that nonlinear distortion inside the reflector results in variations in phase and amplitude across the mouth of the reflector which then impact the focusing of the shock wave onto a kidney stone. I will reflect on David's role as a Ph.D. supervisor and his mentorship of acousticians young and old.

2:35

4pPA4. Bridging shock: Blackstock's general solution for propagation of finite-amplitude sound waves. Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu), Allan D. Pierce (Retired, East Sandwich, MA), and Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Reflections are offered on D. T. Blackstock ["Connection between the Fay and Fubini solutions for plane sound waves of finite amplitude," *J. Acoust. Soc. Am.* **39**, 1019–1026 (1966)]. This work treats the propagation of planar finite-amplitude sound waves. It nominally links the Fourier-series-based solution of Fubini for the pre-shock condition, and the distant, post-shock solution of Fay. However, addressing the transition region between these two required solving the entire problem, which Blackstock did with aplomb. This additionally enabled specification of undetermined factors in the Fay solution, statement of an explicit bound on the applicability of the solution, and generalization to cylindrical and spherical waves. Hallmarks of Blackstock's work are evident: scholarly mindfulness of historical precedent, meticulous analysis that inevitably advanced understanding of nonlinear acoustic phenomena, and elegance in presentation, which was, moreover, pedagogically convincing.

2:55

4pPA5. How graduate work with David T. Blackstock led to a lifelong obsession with wave equations. Thomas L. Szabo (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

As one of David T. Blackstock's first graduate students at the University of Rochester (1966), I quickly admired his elegant style of normalizing and solving equations and was baffled by his frequent allusions to great acousticians and mathematicians of the past. Shortly after my arrival, he submitted his transient solution to the viscous wave equation to J.A.S.A. We planned a summer research project (1967) to validate his solution experimentally. He also introduced me to advanced methods for solving equations. David told me there was no general transient solution to this equation, which inspired my determination to solve it. My data confirmed his predictions leading to my first published paper. I left Rochester with an M.S. and afterwards, I filled several notebooks with unsuccessful equations. Eventually, after ten years in ultrasound imaging, I chose the University of Bath for its excellent PhD program in nonlinear acoustics. David visited me there in 1990 and we discussed his 1985 general solution to the Burgers equation. Later at Bath, twenty-three years after beginning, aha moments led me to more general and causal solutions to both linear and nonlinear viscous wave equations for frequency power law media and papers, cited a thousand times.

3:15–3:30 Break

3:30

4pPA6. David Blackstock and the experimental explanation of the spikes on sonic boom waveforms. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

The present paper discusses an extremely clever experiment perforrmed by David Blackstork and his student Bruce Davy at Rochester in the early 1970s. In the late 1960s people were puzzled by some anomalous features in the measured waveforms of sonic booms recorded at the ground during flyovers of supersonic aircraft. The waveforms were supposed to look like the letter N, but that was not always the case. Sometimes there were strange upward reaching spikes just behind the leading and trailing shocks and in other instances the wavefront was rounded. There were a lot of explanations kicking around, and the present author and David Blackstock came up with the idea that it was somehow caused by imperfect focusing of wavefonts after traveling through regions of higher sound velocity. During our discussions, Blackstock came up with the remark, ''Let's talk experiment.'' What resulted was an experiment that would have been worthy of Lord Rayleigh. The paper that reported this, titled ''Measurements of the Refraction and Diffraction of a Short *N*-Wave by a Gas-Filled Soap Bubble,''appeared in *J. Acoust. Soc. Am.* in March 1971. Besides really nailing down the physical phenomena responsible for the spikes on sonic boom waveforms, it illustrated a wealth of physical concepts and experimental techniques. The present talk discusses the background of the paper, the physics that it used, and the influence it had on subsequent research.

3:50

4pPA7. From laboratory-scaled experiments to numerical predictions of sonic boom level variability due to atmospheric turbulence. Philippe Blanc-Benon (CNRS, LMFA UMR 5509 CNRS Ecole Centrale Lyon, 36 Ave. Guy de Collongue, Ecully 69134, France, philippe.blanc-benon@ec-lyon.fr)

Supersonic flights of aircraft through the atmosphere create shock waves called "sonic booms." Atmospheric turbulence affects the perceived loudness of sound at the ground level, mainly by changing its amplitude and rise time, which are significant factors in determining the acceptability of supersonic flight. Some authors compared theoretical predictions of sonic boom distortion to sonic boom recordings, but such analyses are limited because the parameters of turbulence cannot be measured accurately enough to allow detailed quantitative comparisons. In the 1990s David Blackstock showed that laboratory scaled experiments using N-waves produced by electrical sparks and a downscaled atmosphere offer an attractive alternative to field measurements since both the acoustic source and the turbulence can be controlled (Lipkens and Blackstock, JASA 1998, Lipkens and Blanc-Benon, CRAcadSci Paris 1995). Following this research, we developed at Ecole Centrale de Lyon two experimental setups to study separately the influence of temperature or random velocity fluctuations on the waveform distortion in relation to the occurrence of random caustics at large distances of propagation (Blanc-Benon, Lipkens, Dallois, Hamilton, and Blackstock, JASA 2002). The aim of this paper is to present recent experimental results obtained using optical interferometry and numerical predictions based on the nonlinear KZK propagation equation.

4:10

4pPA8. David Blackstock and the reflection and refraction of finite amplitude waves. Victor W. Sparrow (Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

In honor of David Blackstock, this presentation recalls his contributions regarding nonlinear acoustics in the areas of reflection and refraction. Some of this is well documented and other pieces are not. For example, in his 1960 Ph.D. dissertation David was the first to recognize Pfriem's pressure amplification equation would apply to finite amplitude plane-wave reflection from a rigid surface. Acoustic pressure MORE than doubles for finite amplitude rigid surface reflection, and it is the variational sound speed that exactly doubles. And this result provides a basis for an understanding of oblique reflection and the formation of finite amplitude Mach stems. Further, the work of David's Ph.D. student Frederick Cotaras on finite amplitude refraction from 1989 will be highlighted in addition to other contributions as time permits.

4:30

4pPA9. David T. Blackstock's *Fundamentals of Physical Acoustics*. Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 03722, Republic of Korea, ohm@yonsei.ac.kr)

One can hardly imagine that a book, written on a subject as complicated as physical acoustics, could serve as both an introductory text for beginning students and a serious reference for experts. *Fundamentals of Physical Acoustics* by David T. Blackstock is one such book. Nearly 40 years in the making, the book has its roots in the class notes that Blackstock had written for courses on acoustics throughout his academic career. Sitting in his Acoustics I course as a first-year graduate student at University of Texas at Austin in 1994, I was (gratefully in retrospect) one of those many guinea pigs whom Professor Blackstock tested his manuscript on. What came out of this long and arduous journey is a book that is not only battle tested and combat proven, but also praised nowadays as a modern classic in physical acoustics literature. In this talk, I share my experience with this book, now as a professor who has used the book for graduate-level acoustics courses for the last 14 years. Its pedagogical philosophy, unique structure, real-world drawn problems, adaptability to a MOOC (massive open online course), and last but not least, his lucid and elegant writing style will be highlighted.

4:50–5:10
Open Mic

Session 4pPPa

Psychological and Physiological Acoustics and Speech Communication: Challenges and Opportunities in Technology for Remote and Virtual Testing

Axel Ahrens, Cochair
Technical University of Denmark, Ørsteds Plads 352, Kgs. Lyngby, 2800, Denmark

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

Chair's Introduction—1:30

Invited Papers

1:35

4pPPa1. Improved accuracy and computational efficiency in virtual acoustic rendering using principal components-based amplitude panning. 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 (Univ. of Louisville, Louisville, KY)

Virtual loudspeaker arrays can be rendered over headphones using a head-related impulse response (HRIR) dataset, but these techniques often introduce undesirable spatial and timbral artifacts that are associated with distortions in the interaural level difference (ILD) cue. Here, a new and efficient virtual acoustic rendering technique that minimizes ILD distortions is described and evaluated. The technique uses principal components analysis (PCA) performed on the time-domain minimum-phase representation of the directionally dependent HRIRs. The resulting principal components (PCs) act as virtual acoustic filters, and the PCA scores are implemented as panning gains. These panning gains map the PC filters to source locations, similar to vector-base amplitude panning (VBAP). Thus, the technique is termed principal components-based amplitude panning (PCBAP). A spatial and spectral error analysis of ILD shows similar reconstruction accuracy is obtained using PCBAP with 15 filters compared to a 256-loudspeaker VBAP array. PCBAP is also scalable to simulate large loudspeaker arrays with a fixed set of PC filters and only small increases in computational requirements. Results from psychophysical testing suggest that perceptual equivalence between a PCBAP-rendered source and a reference source rendered with the original HRIR dataset is directionally dependent but can be achieved with as few as 10–20 PCs.

1:55

4pPPa2. Remote measurement of spectral weighting in sound localization using virtual reality headsets. Jwala P. Rejimon (Ctr. for Hearing Res., Boys Town National Res. Hospital, Chapel Hill, NC), Monica L. Folkerts (Hearing and Speech, Vanderbilt Univ., Nashville, TN), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, cstecker@spatialhearing.org)

Unequal weighting of binaural information across frequency can reduce sensitivity in the presence of competing but uninformative cues ("binaural interference"), a potentially serious problem for listeners who use combined electric and acoustic (EAS) hearing. Here, we used virtual-reality techniques to measure spectral weighting functions (SWF) during localization of simulated EAS stimuli [see van Ginkel *et al.*, *JASA* **145**, 2445 (2019)]: low-frequency "acoustic" noise bands and high-frequency "electric" click trains. This study was conducted remotely at each participants' location, using sanitized and calibrated research equipment (Oculus Quest 2 headsets and Sennheiser HD 280 Pro headphones) delivered by lab personnel. Sounds were presented with random binaural-cue jitter across seven components on each trial. SWFs were computed by statistical regression of localization responses onto binaural cue values. SWFs confirmed the dominance of 400–1000 Hz components reported previously [Stecker, Folkerts, & Stecker 2019(A), JASA 145:1720]. Introducing a gap between noise and click-train components increased weight on neighboring components, consistent with reduced interference in such conditions [van Ginkel *et al.*, 2019]. Finally, presenting noises and click trains from competing locations reduced weight on non-target components for most but not all participants, a potential marker of susceptibility to binaural interference. [Work supported by NIH R01-DC016643, NIH T35-008757.]

2:15

4pPPa3. Web-based remote testing as a viable option for measuring binaural hearing abilities. Ellen Peng (Boys Town National Res. Hospital, 1500 Highland Ave., Madison, WI 53705, ellen.peng@boystown.org), Emily Burg (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI), Tanvi Thakkar (Waisman Ctr., Univ. of Wisconsin-Madison, Lacrosse, WI), Shelly Godar, Won Jang, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

Web-based remote testing can increase access to clinically diverse populations in hearing research. However, whether remote testing affects data quality compared to in-lab testing is unknown. Data for binaural hearing tasks may be particularly vulnerable to commercial-grade hardware limitations and background noise from the testing environment. We replicated two published studies on binaural hearing abilities using a web-based behavioral experimental platform (Gorilla.sc). In Experiment 1, we replicated an experiment by Goupell *et al*. [*JASA* **133(4)** (2013)] that tested intra-cranial lateralization to interaural time and level difference cues. In Experiment 2, we replicated an experiment by Goupell *et al*. [*JASA* **140(3)** (2016)] that investigated binaural and contralateral unmasking of speech. Two groups of college-aged young adults were tested: one with verified normal hearing (NH) and one with self-reported NH. Testing took place in participants' homes using their own computer and audio hardware. Performance was compared between the two groups of participants as well as with published data collected in-lab. For both experiments, performance was similar for in-lab and web-based remote testing. We found no effect of verified versus self-reported NH. These results suggest that remote testing may be a feasible option for binaural hearing experiments. [Work funded by NIH-NIDCD.]

2:35

4pPPa4. Moderated versus unmoderated remote audiovisual speech perception tasks in children. Liesbeth Gijbels (Speech & Hearing Sci., Univ. of Washington, 1715 NE Columbia Rd., Seattle, WA 98195, lgijbels@uw.edu), Jason D. Yeatman (School of Medicine, Div. of Developmental-Behavioral Pediatrics, Stanford Univ., Stanford, CA), and Adrian KC Lee (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Virtual adaptations for auditory/audiovisual research have received relatively little attention, especially considering developmental work. The recent pandemic provided opportunities to explore different aspects of remote behavioral testing, ranging from adapting the types of stimuli used, to adjusting the methods of behavioral assessment. When working with children online, an additional consideration is the experimenter's use of moderation, which may also influence the results. Our lab evaluated an online audiovisual speech perception task in 6–7-year-old children, in sessions that were moderated ($N = 37$) and unmoderated ($N = 47$) by an experimenter. Differences in performance show the importance of choice of behavioral measures when performing virtual audiovisual research. Additionally, we found differences in attention, drop-out rate, and the number of unusable results. We found that both types of virtual assessments can be used with sufficient considerations taken into account, but the differences in testing procedures should be explicitly stated to facilitate cross-study comparisons. Both moderated and unmoderated remote testing setups have their particular benefits depending on the research question, caregiver involvement, and requirements for sustained attention.

2:55–3:10 Break

3:10

4pPPa5. Migrating in-lab testing to at-home testing: Setup and procedures. Langchen Fan (Oregon Health & Sci. Univ., 677 S. Lowell St., Apt. 562, Portland, OR 97239, langchen@ohsu.edu), Holden D. Sanders, Morgan S. Eddolls (Oregon Health & Sci. Univ., Portland, OR), Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., Portland, OR), and Lina A. Reiss (Oregon Health & Sci. Univ., Portland, OR)

To reduce the risk of spreading COVID-19, we developed an experimental setup that allows for testing of participants at their home with rigor comparable to the laboratory setting. Testing equipment included a touchscreen laptop, high-quality portable sound card, professional headphones, a hotspot USB for internet connection, and a microphone. All equipment was protected in a custom-molded case. The equipment was dropped off and picked up at the participant's home before and after the experiment. The experimental programs were coded in Matlab and compiled into Matlab executables. During the experiment, the researcher conducted experiments on the laptop via remote control software. Simultaneous to testing, the researcher gave instructions and monitored the participant through video call. Participants were screened for normal hearing thresholds using a modified audiogram, between 0.25 and 8 kHz, with one-up, one-down procedure. Participants were next tested on dichotic vowel perception. Ambient sound level of the participant's home was monitored during the experiment. Overall, participants reported that the procedure was easy to understand and follow. Results from remote testing will be compared with those from standard in-person booth testing to validate accuracy of remote testing. [This research was funded by NIH-NIDCD-R01-DC013307.]

3:30

4pPPa6. Speech intelligibility in a virtual restaurant: Using a calibrated system to collect data remotely. Gregory M. Ellis (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Chicago, IL 60626, gregory.ellis@northwestern.edu) and Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Many individuals with hearing-impairment struggle to understand speech in complex environments, even when using appropriate amplification. A restaurant was simulated to better understand the effects of signal-to-noise ratio (SNR), reverberation time (T_{60}), wide-dynamic range compression (WDRC), and digital noise reduction (DNR) on speech intelligibility for listeners with hearing impairment. A virtual restaurant was created with six competing talkers in the environment and target talker in front. Spatialized audio was processed with a 16-band hearing aid simulation. SNR was jittered following a normal distribution (Mean = 0 dB, SD = 2). Two T_{60}'s (0.8 and 1.8 s), two WDRC speeds (the fastest and slowest commercially available) and two DNR conditions (on or off) were the other factors. Three participants had participated in a sound booth when the COVID-19 pandemic affected ability to gather in-lab data. A new protocol was created to strike a balance between remote data collection and experimental control. Data for 25 additional participants were collected using a remote method with calibrated headphone and tablet system provided for the participant to use at home. The remote testing method will be discussed and data obtained using the headphone and tablet will be compared to in-booth data. [Work supported by NIH.]

3:50

4pPPa7. A within-subject comparison of remote and in-laboratory methods for auditory detection studies. Margaret H. Ugolini (Ball Aerosp. Technologies Corp., 2610 Seventh St., Bldg 441, Wright Patterson Air Force Base, OH 45433, margaret.ugolini.ctr@us.af.mil), Eric R. Thompson (711th Human Performance Wing, Human Systems Directorate, Air Force Res. Lab, Wright-Patterson AFB, OH), and Frank S. Mobley (711 Human Performance Wing, Air Force Res. Lab., Wright-Patterson AFB, OH)

While the gold standard of psychoacoustic research involves tightly controlled, laboratory-based inquiry, it has been necessary to develop protocols to conduct experiments on auditory perception outside of the lab without sacrificing experimental rigor and data quality. Outside of the lab, there is the potential for differences in acoustic environment across subjects, as well as increased auditory and visual distractions. The present study compares the difference in performance on a masked auditory detection experiment within the same cohort of subjects across two platforms of varying experimental control: (1) an in-lab, computer-based experiment conducted within a sound booth; and (2) a remote, tablet-based experiment conducted with provided noise-isolating headphones in an uncontrolled—but measured—acoustic environment. Comparisons in performance across experimental platforms and settings will be discussed, including the influence of a concurrent visual task to encourage task focus, and the effect of background noise levels on performance in the remote setting. Results suggest that there was little influence of the setting and equipment on subjects' performance in this masked auditory detection study, as similar results were obtained in both settings. This work aims to provide guidance on best-practices for conducting remote acoustic studies without a detriment to data quality.

4:05

4pPPa8. Remote auditory testing: Test-retest and user impressions. Kaitlin D. Rink (Commun. Sci. & Disord., Western Washington Univ., 516 High Street/ AI 386, MS 9171, Bellingham, WA 98225, dooleyk2@wwu.edu), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and Anna C. Diedesch (Commun. Sci. & Disord., Western Washington Univ., Bellingham, WA)

The Portable Automated Rapid Testing (PART) application was developed to measure basic auditory processing abilities and has been shown to effectively measure these abilities in research and audiologic clinical settings. Here, the effect of testing environment and participant perception on testing location were evaluated comparing the performance of normal-hearing and hearing-impaired participants in their homes versus a clinical setting. Subjective data on participant impressions of testing environments was included to assess feasibility of testing outside a clinical setting, as patient comfort completing tests in environments without clinical supervision may play a role in their overall performance. An important component of implementing PART tests to the clinical test battery is preventing additional test time to an already lengthy audiologic clinical appointment. Though testing time in clinics is limited, the implementation of administering auditory tests from the PART test battery in the waiting room or at the patients' home could provide critical additional information on their auditory processing ability while adding minimal time to appointments. Use of PART may enable access to a broader collection of testing that an audiologist can use to improve patients' outcomes and address patients' hearing concerns beyond a pure-tone hearing test without increasing current clinical test time.

Session 4pPPb

Psychological and Physiological Acoustics: Binaural Hearing and Speech Perception (Poster Session)

Ellen Peng, Cochair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68131

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

All posters will be on display from 2:30 p.m. to 5:30 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m. and authors of even-numbered papers will be at their posters from 4:00 p.m. to 5:30 p.m.

Contributed Papers

4pPPb1. The relationship between spatial release from masking and the ventriloquist effect. Ann Holmes (Dept. of Psychol. Sci., Univ. of Louisville, Louisville, KY 40292, ann.holmes@louisville.edu), Matthew Neal, and Pavel Zahorik (Otolaryngol. and Comm. Disord., Univ. of Louisville and Heuser Hearing Inst., Louisville, KY)

Speech is often multisensory and the perception of auditory-visual cues may not always match reality. This experiment explored how spatial release from masking (SRM) can be influenced by perceptual shifts in spatial location through the ventriloquist effect, and whether such shifts impact speech perception. Normal hearing participants completed two tasks: a localization task and an intelligibility task. Both tasks employed audio-video recordings of the Coordinate Response Measure [Kishline *et al.*, *J. Acoust. Soc. Am.* **148**, 492–495 (2020)] and participants were asked to indicate the location of a talker or to report keywords spoken by one talker in the presence of a competing talker. Participants were exposed to auditory-only and auditory-visual conditions of each task, and in the auditory-visual conditions, audio location was either matched or mismatched with the video location. Listeners showed strong SRM on keyword identification in auditory only conditions and strong visual capture of the auditory target to the talker irrespective of true auditory location. Despite this combination of results predicting a strong perceived SRM elicited by ventriloquism, only moderate perceived SRM was observed from mismatched auditory-visual conditions. Further discussion will address the relative contribution of visuals in matched conditions and effects of perceptual SRM.

4pPPb2. Smartphone-derived 3D head and pinnae modeling for personalized 3D virtual audio. Kirsti Pajunen, Shakti Davis (MIT Lincoln Lab., Lexington, MA), Gabriel Alberts (Speech and Hearing Biosciences and Technol., Harvard Med. School, Lexington, MA), Hannah Wright (MIT Lincoln Lab., Lexington, MA), Heath Jones (US Army Aeromedical Res. Lab., Fort Rucker, AL), and Christopher J. Smalt (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, Christopher.Smalt@ll.mit.edu)

3D audio is commonplace augmented or virtual reality environments, and is used simulate the direction of arrival for sounds presented over headphones. The Head-Related Transfer Function (HRTF) enables 3D audio by characterizing how sound is received from a sound source in every direction. Optimal sound localization performance with HRTF-based 3D audio is thought to require personalized HRTFs because the localization cues are dependent on the size and shape of the head and pinnae. In this work, we propose a prototype pipeline to generate HRTFs from high resolution smartphone videos of the head and ears, and we validate the localization performance of the resulting HRTFs through human-subject behavioral experiments. To create a personalized 3D HRTF, a video was taken of a subject's torso, left ear and right ear using a smartphone camera. Acoustic simulation with Boundary Element Modeling was performed in COMSOL, taking an approximately 1 h to complete. Preliminary virtual sound localization results were obtained on a pilot subject for each of 3 different HRTF types: A generic HRTF, a gold standard acoustically measured HRTF, and the smartphone video HRTF. We found a mean angle error of 36.6, 21.1, and 14.0 deg, respectively, indicating that the smartphone-based HRTF developed in this study provides higher localization accuracy than other methods on this pilot subject. Further testing will add additional subjects, and compare to free-field localization over loudspeakers.

4pPPb3. Are there binaural consequences of speech envelope enhancement? Lucas S. Baltzell, Daniel Cardosi, Jayaganesh Swaminathan (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), and Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu)

Fluctuations in the amplitude envelope play a critical role in the spatial perception of sounds. For example, listeners place increased perceptual weight on binaural cues occurring at onsets, and the steepness of onset slopes can influence binaural sensitivity. For speech stimuli, we recently showed that the temporal weighting of binaural cues is non-uniform and depends in part on the detailed behavior or the envelope. Here, we examined the potential *binaural* consequences of two *monaural* speech enhancement strategies that operate by altering the speech envelope: consonant enhancement (CE) and envelope expansion (EE). While CE and EE improve speech intelligibility under certain conditions, their effect on binaural perception has not been considered. We measured sensitivity to interaural time differences (ITDs) carried by single-word stimuli that were either unprocessed or processed with CE/EE. Thresholds were measured in quiet, in the presence of multi-talker babble, or for a vocoded condition designed to limit the availability of fine-structure ITDs. For ten listeners with normal hearing, while sensitivity depended on the specific word token, there were no systematic effects of speech enhancement on ITD sensitivity. Ongoing work is considering supra-threshold binaural tasks and listeners with hearing loss who have reduced spatial sensitivity. [Work supported by NIDCD.]

4pPPb4. Effects of entropy in real-world noise on speech perception. Erik Jorgensen (Commun. Sci. and Disord., Univ. of Iowa, SCH 307, 250 Hawkins Dr., Iowa City, IA 52246, erik-j-jorgensen@uiowa.edu), Yu-hsiang Wu, and Inyong Choi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA)

Traditional tests of speech-in-noise perception use artificial background noise, leading to results that do not generalize to real-world scenarios. There is a significant need for improved ecological validity in laboratory and clinic speech-in-noise tests, particularly with respect to the types of noise used. However, the acoustic complexity and random nature of real-world noise poses challenges to its use and the interpretation of results. In this study, the complexity of real-world noise from 8 environments was systematically quantified in the time and frequency domains using entropy. Then, segments of noise distributed in entropy from low to high were extracted and combined with target sentences at fixed signal-to-noise ratios. Ten normal-hearing subjects were tested on 200 sentences-in-noise across the entropy distribution in both the time and frequency domains in a trial-by-trial design. Linear mixed effect models showed a significant effect of entropy in the time and frequency domains on number of words correct per trial. There was a systematic improvement in speech perception with increases in entropy variance in both time and frequency domains. Entropy in real-world noise affects speech perception even when the overall signal-to-noise ratio is fixed. Experiments using real-world noise should consider this effect in their design and interpretation.

4pPPb5. Fusion of dichotic vowels with level differences across ears in normal-hearing listeners. Langchen Fan (Oregon Health & Sci. Univ., 677 S Lowell St., Apt. 562, Portland, OR 97239, langchen@ohsu.edu), Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., Portland, OR), and Lina A. Reiss (Oregon Health & Sci. Univ., Portland, OR)

Fundamental frequency (voice pitch) is an important cue for sound segregation. Previous studies show that normal-hearing (NH) listeners fuse pure tones with similar pitches. When two different vowels with the same fundamental frequencies are presented dichotically to the two ears, NH listeners often fuse the two vowels and perceive one vowel. Sometimes the perceived vowel is not one of the two vowels presented. Interaural level difference is an important cue for sound localization, which also helps with sound segregation. The current study introduced level differences into the dichotic vowel pair to examine how level differences affect vowel fusion. Dichotic vowel pairs were selected from synthetic vowels /i/, /ae/, /ɔ/, and /ʊ/ with the fundamental frequency of 109.9Hz. The level difference varied between -8 and 8 dB. The results showed that there was no significant effect of level difference on fusion in NH listeners. However, for some vowel pairs, the single fused vowel perceived changed significantly with level difference, indicating that internal representations of fused vowels can shift with level differences. This research was funded by NIH-NIDCD-R01-DC013307.

4pPPb6. Recalling vowel sequences in completing backgrounds: Effects of rhythmic regularity and tempo. Dylan V. Pearson (Speech and Hearing Sci., Indiana Univ.–Bloomington, 1625 Pinestone Ct., Bloomington, IN 47401, dylpears@iu.edu), Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Gary R. Kidd (Speech and Hearing Sci., Indiana Univ. - Bloomington, Bloomington, IN), and J Devin McAuley (Psych., Michigan State Univ., East Lansing, MI)

Previous studies have demonstrated that the rhythmic regularities embedded in naturally produced speech are essential for speech understanding in the presence of competing backgrounds. In the current study, the facilitative effect of rhythmic regularity was further investigated using a vowel-recall task. Listeners recalled the last three vowels from a sequence of 9 to 18 synthetic vowels presented at an average rate of 2 vowels/second along with a competing sequence of harmonic complexes. Through this task, the rhythmic regularity (isochronous or anisochronous) and tempo of the target and competitor sequences as well as the duration of exposure to target rhythmic context (i.e., the target sequence length) were independently manipulated without affecting the intelligibility of individual target vowels. Better recall performance was found for isochronous target sequences compared to anisochronous target sequences and with longer exposure to the rhythmic context prior to recall. The rhythmic regularity of the competing signal had no significant effect on recall performance; however, the tempo of the competing signal was significant with recall performance improving as the background tempo increased from 1 to 4 vowels/second. These effects of rhythm and tempo are consistent with theories of selective listening based on attentional entrainment.

4pPPb7. Binaural amplitude modulation fusion as a function of modulation depth. Briana N. Martinez (Commun. Disord., California State Univ., Los Angeles, 5151 State University Dr., Los Angeles, CA 90032, bmarti113@calstatela.edu), Langchen Fan, and Lina A. Reiss (Oregon Health & Sci. Univ., Portland, OR)

Binaural fusion occurs when two different signals presented dichotically are perceived as one sound. In previous studies, binaural fusion was studied for stimuli differing in pitch. In this pilot study, binaural fusion was measured for amplitude-modulated tones differing in amplitude modulation (AM) depth in normal-hearing listeners. AM fusion ranges were measured with dichotic AM tones of matched carrier and AM rate but different modulation depths: specifically a reference signal with a fixed AM depth presented at the left ear, and a comparison signal with AM depth varied across trials presented at the right ear. Subjects were asked if they heard one or two sounds. AM fusion ranges were calculated as the range of comparison AM depths that fused with the reference AM depth. Carrier frequencies tested were 500 and 1600 Hz and AM rates tested were 16 and 64 Hz. Preliminary data showed that the closer the comparison depth is to the reference depth, the more likely the subject is to perceive one sound. In other words, the AM fusion range is centered on the reference AM depth. [This research was funded by NIH-NIDCD-R01-DC013307 and supported by the SURIEA program.]

4pPPb8. The aperture problem for binaural hearing can explain ambiguous lateralization of complex binaural stimuli. Daniel J. Tollin (Dept. of Physiol. & Biophys., Univ. of Colorado School of Medicine, RC1-N, Rm 7106, 12800 E 19th Ave., Aurora, CO 80045, daniel.tollin@ucdenver.edu), Matthew J. Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Although we perceive a spatially contiguous, integrated world, sensory systems sample information discretely via small spatial and narrow frequency receptive fields (RFs) in vision and audition, respectively. Undersampling of stimuli by narrow RFs produces ambiguous representations (e.g., the classic "aperture problem" that visual motion cannot be uniquely determined by one RF). Here, we posit a similar problem for encoding low-frequency (<1500 Hz) interaural time differences (ITDs). Current models of ITD processing propose large ensembles of ITD-sensitive neurons spanning wide ranges of preferred ITD and characteristic frequency (CF). To account for anomalous lateralization of narrowband but robust lateralization of broadband stimuli, such models require two paradoxical assumptions that have recently come into question; (1) the existence of neurons preferring larger than physiologically possible ITDs, and (2) population biases overemphasizing small ITDs. Here, we consider an alternative account based on a narrow and weighted ITDxCF aperture through which the stimuli are perceived. ITDs are bounded by $\pm\pi$ interaural phase difference and CFs by dominance weighting, where $\sim$600–750 Hz contributes most to lateralization. A more detailed consideration of this "binaural aperture" sheds light on how lateralization data emerge from limited ensembles of neurons constrained to physiologically relevant ranges of ITD and frequency.

4pPPb9. A rapid measurement for the effect of interaural mismatch on interaural time difference sensitivity. Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, jaronoff@illinois.edu), Hannah E. Staisloff, Simin Soleimanifar, Mona Jawad (Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Leslie R. Bernstein (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., Farmington, CT)

Cochlear implant (CI) users often have difficulties utilizing binaural cues such as interaural time difference (ITD) because of a mismatch in

terms of the cochlear stimulation location in each ear. Methods for measuring this mismatch often rely on psychophysical approaches that can require hours of testing. The goal of this study was to evaluate a rapid method for measuring the effects of mismatch on ITD sensitivity. Seven normal hearing listeners were tested with a vocoder that simulated current spread. Participants were tested with a traditional adaptive ITD protocol and a rapid descending ITD protocol with fixed ITDs starting at 2000 μs and decreasing in two dB steps to approximately 50 μs. In both protocols, four stimuli were presented. The first and last stimuli always had an ITD of 0 μs. One of the two center stimuli had a non-zero ITD. Participants had to identify which had the non-zero ITD. The carriers were altered to simulate varying mismatch across ears. The preliminary results indicated a similar effect of mismatch for both protocols, with considerably shorter testing time for the descending ITD protocol. This suggests that the rapid descending ITD protocol is a valid method for assessing the effects of interaural mismatch.

4pPPb10. Do humans integrate auditory and text information in a statistically optimal fashion? Gavriel D. Kohlberg (Otolarynology - Head and Neck Surgery, Univ. of Washington, Virginia Merrill Bloedel Hearing Res. Ctr., 1701 NE Columbia Rd., CHDD Clinic Bldg., Rm CD176, Seattle, WA 98115, kohlberg@uw.edu), Eric M. Prater (School of Medicine, Univ. of Washington, Seattle, WA), Yi Shen, Adrian K. C. Lee (Speech & Hearing Sci., Univ. of Washington, Seattle, WA), Jay T. Rubinstein (Otolarynology - Head and Neck Surgery, Univ. of Washington, Seattle, WA), Les E. Atlas (Elec. Eng., Univ. of Washington, Seattle, WA), and Richard A. Wright (Linguist., Univ. of Washington, Seattle, WA)

Decreased speech perception in noise is a major problem, especially for older adults and for those with hearing loss. In order to improve speech perception in noise, the acoustic speech information can be supplemented with visually displayed speech text derived from the output of an automated speech recognition program. While such supplementation of auditory speech information with visual text has been shown to increase speech perception in noise, it is relatively unknown how listeners combine these two modalities of speech information to achieve improved speech perception. We aim to evaluate how listeners combine visual speech text and auditory speech information. We will evaluate subjects' speech perception based only on auditory information in background noise. We will then evaluate subjects' speech perception based only on visual text information that is derived from an automated speech recognition system. Using these two assessments of speech perception (auditory alone, visual text alone), we will predict listeners' combined performance based on a maximum likelihood estimate model. We hypothesize that listener's performance on joint presentation of auditory speech information with visual text information can be predicted based on the listener's performance on separate presentations of auditory speech information alone and visual text information alone.

4pPPb11. Spectral weighting in sound localization: Effects of competing noise. Monica L. Folkerts (Hearing and Speech, Vanderbilt Univ., 1215 21st Ave. South, Nashville, TN 37232, monica.l.folkerts@vanderbilt.edu), Erin M. Picou (Hearing and Speech, Vanderbilt Univ., Nashville, TN), and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

The localization of complex sounds relies upon spatial cues, namely the interaural time differences (ITDs) and interaural level differences (ILDs) which are distributed across the frequency spectrum of the stimulus. Spectral weighting functions (SWFs) measured using free-field and binaural methods [Stecker *et al.*, *JASA* **145**, 1720 (2019)] have confirmed the spectral "dominance region" [Bilsen and Raatgever, *Acustica* **28**, 131–2 (1973)] in which listeners utilize the ITD cue around 400-1000 Hz. However, the salience of the spectral "dominance region" in the presence of competing noise is unknown. Previous studies demonstrate decreased localization performance in the presence of a white-noise masker, especially when the stimulus does not contain high-frequency components [Lorenzi *et al.*, *JASA* **105**, 3454–3463 (1999); Brungart and Simpson, *JASA* **126**, 3199–3208 (2009)]. The current study measures SWFs in the presence of two uncorrelated lateral white-noise maskers (at –90° and +90° azimuth) for three signal-to-noise ratios (SNRs: +9, 0, –6 dB). Contrary to our hypothesis that the "dominance region" would shift to higher frequency components, the "dominance region" remained salient around 400–1000 Hz across all SNRs. [Work supported by NIH R01-DC016643.]

4pPPb12. Binaural model analysis of wave-field-synthesis auralization over extended listening areas. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu), Samuel Chabot, Mincong Huang, Jonathan Mathews (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), M. Torben Pastore (College of Health Solutions, Arizona State Univ., Tempe, AZ), and E. Ellington Scott (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

This paper evaluates the performance of a recently proposed method to create auralizations with a 2D ray-tracing algorithm and publicly available floor plans. Using this method, acoustic landmarks were simulated for a 128-channel wave field synthesis (WFS) system. Late reverberation parameters are calculated using additional volumetric data. The approach allows the rapid sonic recreation of historical concert venues with adequate sound sources. The listeners can walk through these recreations over an extended user area (12 $\times$ 10 sqm), and the software suite can be used to calculate room acoustical parameters for various positions directly using a binaural rendering method or via the WFS simulation. To evaluate the performance of the auralization system binaural sound fields are rendered for various grid positions across the listening area and analyzed using a binaural model algorithm. The Binaurally Integrated Cross-correlation Auto-correlation Mechanism (BICAM) [*J. Acoust. Soc. Am.* **140**(1), EL143] was selected for this purpose because it can estimate binaural activity patterns from a running signal with a 2-layer cross-correlation algorithm. Using this algorithm, the reflection patterns of early reflections and traditional room acoustical parameters (ISO 3382) were compared between the WFS auralization and direct binaural rendering method. [Work support by NSF IIS-1909229.]

4pPPb13. Effects of target level on release from masking by voice-gender difference and spatial separation between talkers. Yonghee Oh (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2128, Gainesville, FL 32610, yoh@phhp.ufl.edu), Hannah Schoenfeld, Allison O. Layne, and Sarah E. Bridges (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

In multi-talker listening situations, spatial and voice-gender separation of the target from the masking speech are two major acoustic cues leading to substantial release from masking in normal-hearing listeners. A recent study demonstrated that the masking release by those two cues elicits an unequal perceptual weighting and a single point of intersection where the magnitude of masking release was the same for the two cue types, with a gender-specific difference of transition between two types of masking release [Oh *et al.*, *JASA-EL* (2021)]. The goal of this study was to investigate how the interactions of these two cues are affected by various levels of the target (30, 40, and 50 dB SL). Speech recognition thresholds in the competing speech were measured, and masking release benefits by either voice-gender difference or spatial separation cues were calculated. Results revealed that the weighting of masking release from two cues is non-linearly related to the target levels. In addition, gender-related masking release was maximized when the target was presented at 40 dB SL. These results imply that a target level can be one major factor associated with maximal speech perception in noise in normal-hearing listeners. [Work supported by ASHA New Century Scholars Research Grant.]

4pPPb14. Classifying and quantifying individual differences in phonetic categorization patterns. Michael L. Smith (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, smit8854@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

A key component to speech perception is the accurate categorization of phonemes, which is classically measured by labeling functions in forced-choice tasks. The common assumption is that poor categorization drifts toward chance performance at 50%. However, the current study shows that individuals cluster in distinct patterns that diverge from that assumption. In an experiment where listeners categorized phonemes whose frequency

spectra were acutely or gradually shifted upwards (reminiscent of cochlear implants), some listeners were better able to maintain distinct perceptual phonetic categories despite increasing amounts of frequency mismatch, while others could recalibrate their perception commensurate with the shift, up to a certain point before categorization ultimately broke down in a variety of ways. This study highlights the importance of classifying categorization patterns at the individual level, rather than averaging across groups, and invites reconsideration of how to quantify and interpret overlooked behavioral response patterns that signify when and how phonetic categorization breaks down. This study emphasizes provides novel ways to mathematically analyze qualitatively unique signatures of categorization breakdown (asymmetry, bias, all-or-none judgments, etc.) that are potentially ripe for studying individual differences in strategy and capacity for overcoming specific challenges relating to hearing or adaptation to acoustic degradation.

4pPPb15. Exploring a listener's ability to "eavesdrop" on two simultaneous conversations. Nathaniel J. Spencer (Air Force Res. Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433, nathaniel.spencer.1.ctr@us.af.mil), Eric R. Thompson, and Brian D. Simpson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Most speech perception studies assume that the talkers are speaking to the subject, but little is known regarding subject ability to monitor exchanges between interlocutors (i.e., "eavesdrop"). The current study measured subject ability to monitor two concurrent, artificially generated "conversations" using phrases from the Coordinate Response Measure corpus. Individual conversations consisted of a "prompt" phrase of the form: "Ready [call sign] go to [color]-[number] now" and a "response" phrase of the form: "[same call sign] going to [color]-[number]". A color and/or number prompt-response mismatch occurred within a conversation on 50% of trials. The listener was asked to identify whether a mismatch took place and, if so, identify in which of the two conversations the mismatch occurred. The (virtual) conversations were either co-located or spatially separated ($\pm 15°$, $\pm 90°$). All talkers were of the same gender, or competing prompters and competing responders were of different genders. Group-mean performance ranged from about 65–80% correct in mismatch detection, and about 70%–85% correct in call sign identification (for mistake-ID correct trials). Performance improved when competing talkers were separated to ± 15 deg in space, and/or in gender. This suggests that, on average, listeners can follow simultaneous conversations over time, particularly when competing talkers differ in pitch or space.

4pPPb16. Individual differences in personality traits affect performance, frustration, and effort in a speech-in-noise task. Paola Medina Lopez (Univ. of Puerto Rico, Mayagüez, Puerto Rico), Timothy Stump (Linguist, Univ. of Texas, Austin, TX), Nicole Kirk (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN), Vincent Jung (Mt. Hebron High School, Ellicott City, MD), Jane E. Clougherty (Dornsife School of Public Health, Dept. of Environ. & Occupational Health, Drexel Univ., Philadelphia, PA), and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu)

Individual personality traits may influence psychological and physiological response to background noise. Here, we assessed participants' ($N = 93$) performance on a speech-in-noise arithmetic task along with self-rated effort and frustration and several emotion/personality-related questionnaires. We also manipulated the feeling of control that participants had over noise level by allowing half to choose task difficulty (easy" or "hard) while assigning the others to a difficulty level without agency. On each trial participants heard three spoken digits and added them. On half of the trials, one digit was masked by noise. To better assess the impact of choice and perceived difficulty, signal-to-noise ratio and spoken digits were identical across all conditions. Participants who scored higher on the Extraversion scale of the Big-5 personality inventory expressed significantly more frustration when they had no control over task difficulty, and less frustration when they had control (interaction, $p = 0.03$). Similarly, participants with a more external (higher) Locus of Control reported putting significantly more effort into the hard task (whether assigned or chosen) but less effort into the easy task (interaction $p = 0.01$). Results will be discussed in terms of implications for future research on noise sensitivity and long-term health.

4pPPb17. Differences in sound-source localization in reverberant environments that depend on stimulus-type and mode of presentation. M. Torben Pastore (College of Health Solutions, Arizona State Univ., PO Box 870102, ASU, Tempe, AZ 852870102, m.torben.pastore@gmail.com), Yi Zhou (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ), Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), and William A. Yost (College of Health Solutions, ASU, Tempe, AZ)

One interesting aspect of sound-source localization in reverberant environments is that different stimuli (e.g., speech versus noise) can elicit different spatial sensations at the same delays between the direct sound and its reflections. For example, a given delay between a single simulated direct sound and a delayed copy presented from a different location may elicit the sensation of either a single sound located at or near its source for speech while a noise stimulus under otherwise the same conditions might be perceived as a direct sound with echoes. Furthermore, presenting stimuli over headphones versus loudspeakers can further modify spatial perception for the same stimulus conditions. This modeling study will explore reasons for these two interesting differences of behavioral outcome, using simulation tools developed in Montagne and Zhou (*JASA*, 2018) and Pastore and Braasch (JASA, 2020).

4pPPb18. Prediction of speech recognition in background noise and competing speech from suprathreshold auditory and cognitive measures. Jonathan H. Venezia (VA Loma Linda Healthcare System, Loma Linda, CA, Jonathan.Venezia@va.gov), Nicole Whittle, Christian Herrera Ortiz, Marjorie R. Leek (VA Loma Linda Healthcare System, Loma Linda, CA), Caleb Barcenas, and Grace Lee (Loma Linda Univ., Loma Linda, CA)

Previous studies have struggled to identify measures beyond the audiogram to reliably predict speech-in-noise scores. This may owe to: (i) different mechanisms mediate performance depending on materials and task; and (ii) effects are not reproducible. Here, 38 listeners with normal/near-normal audiograms completed batteries of temporal auditory and cognitive tests, and speech recognition ("Theo-Victor-Michael" test) in speech-shaped noise (SSN), speech-envelope modulated noise (envSSN), one (1T) and two (2T) competing talkers. A two-stage Bayesian modeling approach was employed. In Stage 1, speech scores were corrected for target-word frequency/neighborhood density, psychometric function parameters were extracted from temporal tests, and cognitive measures were reduced to three composite variables. Stage 2 then applied Gaussian process models to predict speech scores from temporal and cognitive measures. Leave-one-out cross-validation and model stacking determined the best combination of predictive models. Performance in SSN/envSSN was best predicted by temporal envelope measures (forward masking, gap duration discrimination), while performance in 1T was best predicted by cognitive measures (executive function, processing speed). Temporal fine structure measures (frequency-modulation, interaural-phase-difference detection) predicted the number of 1T distractor responses. All models failed on 2T. These results show that prediction of speech-in-noise scores from suprathreshold "process" measures is highly task dependent.

Session 4pSA

Structural Acoustics and Vibration, Engineering Acoustics, Physical Acoustics, and Musical Acoustics: Additive Manufacturing and Acoustics

Christina Naify, Cochair
Applied Research Laboratories, The University of Texas at Austin,

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Kathryn Matlack, Cochair
University of Illinois at Urbana-Champaign, 1206 W Green St., Urbana, IL 61801

Chair's Introduction—1:30

Invited Papers

1:35

4pSA1. Tracking the evolution of structural resonances during the build phase of a wire-arc additively manufactured part. Karl A. Fisher (Mater. Eng., Lawrence Livermore Natl. Lab., 7000 E. Ave., Liveremore, CA 94551, fisher34@llnl.gov), John Elmer (Mater. Eng., Lawrence Livermore Natl. Lab., Livermore, CA), and James Candy (Eng., AD Office, Lawrence Livermore Natl. Lab., Livermore, CA)

Acoustic resonance tracking is investigated for a wire arc additive manufacturing process as a possible real time *in situ* diagnostic during the build. A small 40kHz acoustic emission sensor was attached to one end of a 5 × 1.5 in. cylindrical 316 SS rod. A series of layers (weld beads) were programmed into the control software of the robot to build a thin bar upward off the end of the cylinder. A high-speed long-duration digitizer-recorder was used to capture the transient acoustic signals (100 Hz–70 kHz) during the build process. Typical build durations were on the order of 1–2 h. The transient data were analyzed using short time window FFTs, as a function of build parameters. The model and experiment share similar features, however there are some notable differences. In particular, the number of actual modes recorded in the experiment is less than predicted due to the sensors low in-plane sensitivity. The current results indicate that the evolving spectral response of a part during the build process has the potential to provide real time process monitoring information. The time-frequency analysis of the diffuse sound field in the part, reveals a unique spatial and temporal perspective of temperature, stress and geometric features.

1:55

4pSA2. Quantitative characterization of microstructure in additively manufactured metals with nonlinear ultrasound. Aurelio Bellotti (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 790 Atlantic Dr., Atlanta, GA 30332-0355, aurelio.bellotti@gatech.edu), Jin Yeon Kim (School of Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA), Joseph Bishop (Sandia National Labs., Albuquerque, NM), Bradley Jared (Dept. of Mech., Aerosp., and Biomedical Eng., Univ. of Tennessee, Knoxville, TN), Kyle Johnson, Donald Susan, Philip Noell (Sandia National Labs., Albuquerque, NM), Carly Donahue (Los Alamos National Lab., Los Alamos, NM), and Laurence Jacobs (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Our work employs nonlinear ultrasonic techniques to track microstructural changes in additively manufactured metals. One study is based on the use of the second harmonic generation technique with Rayleigh surface waves used to measure the acoustic nonlinearity parameter, β. Stainless steel specimens made through laser-powder bed fusion and laser engineered net shaping were compared with traditional wrought manufactured specimens. The β parameter is measured through successive steps of an annealing heat treatment intended to decrease dislocation density. In agreement with fundamental material models for the dislocation-acoustic nonlinearity relationship in the second harmonic generation, β drops for each specimen throughout the heat treatment before recrystallization. Geometrically necessary dislocations (GNDs) are measured from electron back-scatter diffraction as a quantitative indicator of dislocations; average GND density and β are found to have a statistical correlation coefficient of 0.852 showing the sensitivity of β to dislocations in additively manufactured metals. Moreover, β shows an excellent correlation with hardness, which is a measure of the macroscopic effect of dislocations. Ongoing work continues trying to characterize additively manufactured stainless steels with nonlinear ultrasound through noncollinear mixing.

2:15

4pSA3. Ultrasonics for monitoring melt pool dynamics and *in situ* sensing of microstructure during powder bed fusion additive manufacturing. Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu), Nathan Kizer (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Abdalla Nassar, Edward Reutzel (Penn State Appl. Res. Lab., University Park, PA), Haifeng Zhang (Univ. of North Texas, Denton, TX), Anthony Rollett (Carnegie Mellon Univ., Pittsburgh, PA), and Tao Sun (Univ. of Virginia, Charlottesville, VA)

Ultrasonics are viable tools to monitor internal microstructure of polycrystalline metals. Recently, the need for such tools has dramatically increased as metallic additively manufactured (AM) materials are rapidly becoming common in engineered structures. Nondestructive techniques like X-ray CT and ultrasonics are increasingly being used as a quality control step to help assure performance. Ultrasonics can also be used to help improve and develop AM processes by integration as an *in situ* method to assess the relationship between part microstructure and process parameters. This presentation will cover the development of two *in situ* ultrasonic techniques aimed at improving the state of the art in metal AM. First, ultrasound is employed to monitor keyhole melt pool dynamics in Al6061 samples. In this application, a high-frequency ultrasonic shear wave scatters from the liquid-solid boundary of the melt pool, which allows us to monitor keyhole initiation, melt pool depth, fluctuations in depth, and solidification. The sensitivity was confirmed through ground truth measurements using simultaneous synchrotron x-ray imaging. Then, progress on using *in situ* diffuse ultrasonic scattering on Gr91 powder bed fusion parts will be highlighted. The microstructure sensitivity of diffuse scattering is thought to be able to discern the formation of different microstructures during the manufacturing process.

2:35

4pSA4. Ultrasonic scattering predictions for metal additive manufacturing. Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu), Cody Pratt, Luz D. Sotelo, and Jazmin Ley (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Additive manufacturing (AM) of metals offers great promise for the creation of new engineering solutions because of the expanded design space offered by these new approaches. However, several challenges remain with respect to control of the spatial organization of the microstructure. The complex thermal cycles that drive the densification of material can result in dendritic grain growth, material texture, and residual stresses that must be quantified if a certified component is desired. Ultrasonic techniques have been shown to have the sensitivity and resolution necessary to play a major role in the metal AM inspections. The microstructural challenges present in AM samples have been examined, in many cases individually, but AM samples are especially complex. In this presentation, AM samples of 316 stainless steel created using laser powder bed fusion (LPBF) are studied ultrasonically in terms of their wave speed, attenuation, and diffuse backscatter characteristics. Electron backscatter diffraction (EBSD) measurements are used to create digital synthetics with statistically equivalent properties. The synthetics are then used to make ultrasonic predictions that are compared with the measurements. Prospects for *in situ* sample validation are then discussed.

Contributed Papers

2:55

4pSA5. Effect of architecture and process on the ultrasonic mapping of hybrid additively manufactured materials. Luz D. Sotelo (Univ. of Nebraska-Lincoln, Mech. and Mater. Eng., Lincoln, NE 68588, luz.sotelo@huskers.unl.edu), Jazmin Ley, Cody Pratt, Kossi Loic Avegnon, Michael P. Sealy (Univ. of Nebraska-Lincoln, Lincoln, NE), and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Hybrid additive manufacturing enables the creation of materials with functional microstructure and mechanical property patterns. While samples with functional patterns provide unique advantages in the overall performance of the material, they present a challenge for nondestructive evaluation. In particular, the nondestructive mapping of the properties imparted by hybrid AM processes is critical. In this work, hybrid 316 stainless steel samples are created using laser powder bed fusion (LPBF) with varying hybrid architectures and with a variety of hybrid processes. The ultrasonic responses of these samples, in terms of wave speed, attenuation, and backscatter amplitude, are evaluated for each architecture and process combination. These responses are compared with those from conventionally manufactured samples and traditional AM samples. This evaluation provides information on the sensitivity of ultrasonic responses to hybrid architecture and process, highlighting the potential of ultrasonic nondestructive methods for the characterization of hybrid AM components. [This material is based upon work supported by the National Science Foundation Graduate Research Fellowship Program under Grant No. 1610400. Any opinions, findings, and conclusions or recommendations expressed in this material are those of the author(s) and do not necessarily reflect the views of the National Science Foundation.]

3:10–3:25 Break

3:25

4pSA6. Statistical characterization of low-porosity, additively manufactured metal foams measured in an acoustic impedance tube. Stephanie G. Konarski (US Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, stephanie.konarski@nrl.navy.mil), Christina J. Naify, Charles A. Rohde (U.S. Naval Res. Lab., Washington, DC), and Scott Roberts (Jet Propulsion Lab., California Inst. of Technol., Pasadena, CA)

This work focuses on the acoustic response of aluminum foams with different porosities additively manufactured using powder bed fusion. Emerging voxel-by-voxel additive manufacturing, which further expands the achievable microstructure design space through localized spatial control of the printing properties, allows for direct fabrication of samples with varying percent porosities. Unlike traditional, rigid-frame foams with very high porosities, the samples range from fully dense to approximately 50% porosity. This work utilizes normal incidence acoustic impedance tube measurements and the Johnson–Champoux–Allard–Lafarge analytic model, which are common characterization methods for traditional rigid-frame porous media. For the reflection and transmission coefficients measured in an acoustic impedance tube, Bayesian statistical inversion is performed with the Johnson–Champoux–Allard–Lafarge model to obtain distributions of the possible values for the six required model parameters. By using a statistical characterization approach, the applicability of these common acoustic measurement and modeling methods for lower porosity additively manufactured foams is also assessed. [This work was supported by the Office of Naval Research.]

3:40

4pSA7. Double phononic crystal lens-based enhancement of underwater power transfer. Ahmed Allam (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW (Love Building), Rm. 204, Atlanta, GA 30332-0405, a.allam@gatech.edu), Karim G. Sabra, and Alper Erturk (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Acoustic waves are widely used for underwater communication and also investigated for power transfer owing to their low attenuation compared to electromagnetic waves. We investigate enhancing the performance of an underwater Acoustic Power Transfer (APT) system using Gradient-Index Phononic Crystals (GRIN-PCs). The GRIN-PCs consist of air inclusions in a 3D-printed polymer and are used to construct 3D acoustic lenses capable of collimating and focusing acoustic waves. The lenses are used to improve the directivity of a point source transmitter and focus the acoustic waves at a point receiver to reduce spreading losses in the APT system. The developed lenses allow for using lower frequency transducers to transmit power, hence reducing wave attenuation, and enhancing the efficiency of the system. The enhancement is verified experimentally using hydrophones as point transmitters and receivers in a double-lens system. Additionally, the lenses were used to enhance the performance of 100 kHz piezoelectric transducers with limited directivity, increasing the output power by an order of magnitude. Numerical models based on the finite element method were used to study the effect of varying the separation distance, lens aperture, and frequency on the pressure field. The developed APT concept could find applications for power delivery to wireless devices used in ocean monitoring, underwater navigation, marine life tracking, among others.

3:55

4pSA8. Additive manufacturing of architected ceramics foam for noise reduction. David Nevarez-Saenz (Dept. of Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, dxnevarez-saenz@shockers.wichita.edu), Ted Adler (School of Art, Design and Creative Industries, Wichita State Univ., Wichita, KS), Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS), and Wei Wei (Dept. of Mech. Eng., Wichita State Univ., Wichita, KS)

Porous foams are routinely used for noise control applications; their open cellular structure provides good sound dissipation performance over a broad frequency range while retaining low bulk density. However, conventional acoustic foams cannot satisfy the mechanical and flammability constraints frequently encountered in aerospace applications. While fiberglass blankets are currently used for high temperature noise dissipation applications, their batch-to-batch performance variability and lack of mechanical stiffness remains an engineering hurdle. In this study, we investigate the viability of additively manufacturing ceramic foams for high temperature noise dissipation applications. We demonstrate the use of a low-cost, material extrusion 3D printer to fabricate ceramic foams with controlled porous architectures that can be tuned to provide exceptional sound absorption properties. The printed and sintered ceramic foams are tested using a normal incidence impedance tube and the effect of various print, sintering, and geometrical parameters on the acoustical impedance are studied. Our work shows that low-cost additive techniques present a viable step toward scalable fabrication of ceramic foams with tailored mechanical and acoustical properties.

4:10

4pSA9. 3D printed fibrous porous structures with enhanced noise reduction properties. Will Johnston (Aerosp. Eng., Wichita State Univ., 1845 Fairmount St., Wichita, KS 67260, wjjohnston@shockers.wichita.edu), Pulitha G. Kankanamalage (Aerosp. Eng., Wichita State Univ., Wichita, KS), Siddharth Pathak, Mary Drouin (Spirit Aerosystems, Wichita, KS), and Bhisham Sharma (Aerosp. Eng., Wichita State Univ., Wichita, KS)

Additive manufacturing allows the cost-effective fabrication of cellular porous structures with tailorable geometries. Structures with small pore sizes offer improved acoustical performance; however, they often result in increased structural weights and are difficult to produce using low-cost additive manufacturing methods. Here, we leverage our recent work demonstrating the fabrication of fibrous structures using 3D printing, to develop multifunctional porous structures with enhanced acoustical properties. Our method allows the addition of fibers to a structural host, which helps improve the acoustical performance without a significant weight or flow-reduction penalty. In this presentation, we outline the fabrication method and study the effect of adding fibers to the acoustical performance of porous structures with periodic gyroid unit cells. The effect of fibers on the acoustical impedance and flow resistance is studied using an impedance tube and a flow bench, respectively. We then use an inverse characterization method to analyze the effects of changing fiber and structural parameters on the acoustical performance. Our results show that adding fibers to porous host structures can drastically improve their broadband noise reduction potential.

4:25

4pSA10. Binder jet printing barium titanate piezoelectric ceramic discs. David Schipf (U.S. Naval Res. Lab., NRC Postdoctoral Researcher, 4555 Overlook Ave. SW, Washington, DC 20032, david.schipf.ctr@nrl.navy.mil), Gregory Yesner, and Matthew D. Guild (U.S. Naval Res. Lab., Washington, DC)

In this study, we measure the properties of barium titanate ($BaTiO_3$) piezoceramic discs that have been binder jet printed and infiltrated with epoxy. Binder jet printing is a scalable and relatively inexpensive additive manufacturing technique. With binder jet printing, it is possible to produce atypical geometries, and large quantities of transducers on demand without expensive tooling. However, there are many practical challenges with binder jet printing ceramics, and sintered parts tend to have relatively low densities. We seek to address these challenges and explore the suitability of binder jet printing piezoelectric transducers and active materials. In this presentation we will discuss our techniques for printing and post-processing $BaTiO_3$ ceramic discs. We will discuss porosity, and how epoxy infiltration can improve the functionality of discs. We will present measurements of discs fabricated using a traditional pressing method and measurements of printed discs. These measurements include density, dielectric permittivity, piezoelectric coefficient d33, impedance and dielectric loss. From our measurements, we will attempt to model relationships between printing parameters and final properties. Printing and powder processing techniques that can increase relative density of sintered parts will also be discussed. [Work sponsored by the Office of Naval Research Distribution A: Approved for public release.]

4:40

4pSA11. *In situ* monitoring of polymer 3D printing using ultrasonic Lamb waves generated by a 3D printed wedge. Nathan Kizer (Penn State Univ., University Park, PA 16802, njk19@psu.edu) and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

The additive manufacturing (AM) of polymers has become increasingly popular in industry to create complex geometries for parts while maintaining low manufacturing costs. Rapid nonuniform cooling introduced during the printing process can lead to part warping and delamination which are precursors to part failure and/or poor part quality. Previous researchers have integrated online monitoring techniques such as optical or thermal sensors. These techniques are limited to monitoring exposed areas of the part leaving unexposed areas unmonitored. In this work, ultrasound has been implemented to monitor the inaccessible areas where defects can originate. An *in situ* guided wave-based ultrasonic inspection system was implemented into the test setup to detect delamination during printing. A 3D printed ultrasonic wedge was designed to generate S0 guided waves that were received by a broadband pinducer. The wedge was printed onto a plate of 6061 aluminum acted as the print bed for the experiments. A 3D printed part was printed between the transducers, which was continuously monitored for signs of delamination. This presentation will report on precursors to part delamination that are difficult or impossible to sense using previous methods. This work seeks to reduce 3D part failures and, thus, polymer waste associated with 3D printing.

Session 4pSC

Speech Communication: Perception and Social Evaluation (Poster Session)

Drew J. McLaughlin, Chair
Psychological & Brain Sciences, Washington University In St. Louis, 1 Brookings Drive, St. Louis, MO 63130

All posters will be on display from 1:30 p.m. to 4:30 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m.

Contributed Papers

4pSC1. Investigation of possible Near-merger in Mandarin nasal codas. Suyuan Liu (Univ. of BC, HuanShan Rd., 54-901 Yantai, Shandong 264000, China, suyuan.liu@ubc.ca)

The two codas /n/ and /ŋ/ in Mandarin are known to undergo merger in production of speakers from various dialect groups such as Shanghai Mandarin (Faytak *et al.*, 2020) and Taiwan Mandarin (Chiu *et al.*, 2019). A study by Faytak *et al.* (2020) revealed that even for Beijing Mandarin speakers, who do not show such a merger in production, they still could not distinguish /n/ and /ŋ/ in perception. This indicates a possible instance of near merger in Standard Mandarin, meaning speakers can distinguish between two sounds in production but not in perception (Labov *et al.*, 1991). This study aims to validate whether Mandarin nasal codas demonstrate patterns of near merger with consideration of potential influence by dialect groups. Both speakers of Beijing Mandarin and speakers of Shanghai Mandarin will be recruited. This study consists of one production section where subjects produce minimal pairs in Mandarin that only differ in coda position, and a perception task involving a phoneme categorization task and a goodness rating task of various productions of /n/ and /ŋ/. The hypothesis is that listeners' perceptual performance will correlate with their production pattern (e.g., Fridland and Kendall, 2012). However, if this study validates near merger in Mandarin coda nasals, this will be the first case of a non-tonal near merger in Mandarin. If otherwise, this study still sheds light on how speakers from various Chinese dialect groups might exhibit different cue-weighting strategies.

4pSC2. Predicting asymmetries in vowel perception: Formant convergence succeeds where peripherality fails. 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 Ocke-Schwen Bohn (Aarhus Univ., Aarhus, Denmark)

Vowel discrimination is often asymmetric such that discriminating the same vowel contrast is easier in one direction compared to the opposite direction. According to the Natural Referent Vowel (NRV) framework, these asymmetries reveal a perceptual bias favoring acoustic vowel signals produced with more extreme vocalic gestures, which act as *natural referent* vowels. The NR vowel within a contrast typically falls in a more peripheral location within articulatory/acoustic vowel space (defined by F_1 and F_2) and with F_1 and F_2 in closer proximity. However, these properties do not always align, as in the case of the /e/-/Ø/ contrast. We here compared findings across three studies where asymmetries were observed during discrimination of this contrast. Peripherality predicts an asymmetry such that perception would be better in the /Ø/□/e/ direction. However, all three studies showed better performance in the opposite direction, which also aligns with the prediction based on formant convergence (derived from reported acoustic measures for each stimulus set). These findings suggest that this perceptual bias is shaped by formant convergence, rather than peripherality *per se*, which is presumably tied to the extent of vocal-tract constriction. A follow-up study will obtain articulographic recordings of these vocalic gestures to quantify this articulatory-acoustic relation.

4pSC3. The untapped potential of human language: Investigating the perception of typologically unattested and rare consonants. Catherine Brislane (Dept. of Lang. and Linguistic Sci., University of York, York, North Yorkshire Yo10 5DD, United Kingdom, clb628@york.ac.uk)

Though research into this area is scarce, the perception of rare and unattested consonants has potential for research in speech phonotactics and pronunciation, especially in relation to the creation of ConLangs which typically use non-native' sounds in their phonological inventories. Previous studies explored the "nativeness" of rare/ non-English phonemes (Davidson, 2005; Linzen and Gallagher, 2017) or conducted ABX tests to gauge perceptual similarity (Noguchi and Hudson Kam, 2017; Pisoni *et al.*, 1982). This paper explores why rare and unattested phonemes may be less common in speech today by using a variation of the ABX methodology. Using information from PHOIBLE (Moran and McCloy, 2019), each question contained three distinct sounds: a common phoneme, an English phoneme, and a rare/unattested phoneme (e.g., a palatal tap). The participant would then pick which sound (either 1 or 2) was most like the third "hook" phoneme (sound 3). Sounds 2 and 3 would then swap, creating a new "hook." Similarity identification relative to sound 1 was analysed via logistic regression in R. Notably, several unattested phonemes were chosen over the opposing sound to a significance rate of $p > 0.05$, with a proportion of rare phonemes also being chosen to a degree of significance over English sounds.

4pSC4. Perception of checked tones in Xiapu Min. Yuan Chai (Dept. of Linguist., Univ. of California San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, yuc521@ucsd.edu)

Xiapu Min is a variety of Min Chinese spoken in Xiapu, Fujian, China. The language has seven tones, two of which (T53 and T21) are checked tones that only appear in syllables with a glottal stop coda (Chai and Ye, 2019). Compared with their unchecked counterparts, checked tones have distinct f0 contours, glottalization at the end of the syllable, and are of shorter duration. In this study, we investigate whether those acoustic features are used by listeners of Xiapu Min to identify whether a tone is checked or not. We ran a forced-choice identification task with resynthesized audio stimuli. Stimuli consisted of a natural unchecked token resynthesized with five distinct f0 contours. Each vowel was further modified to have a short versus long duration and presence versus absence of glottalization. The results indicate that the listeners are more likely to identify a stimulus as having a checked tone when it has a high-falling f0 contour (similar to checked T53), shorter duration, or glottalization, among which the

duration has the largest coefficient. Thus, all three phonetic parameters found in checked tone production also influence checked tone identification. However, duration is more likely to matter for the identification of checkedness than f0 or glottalization.

4pSC5. The prioritization of consonants, vowels, and tone in Cantonese word recognition. Fion Fung (Linguist., Univ. of BC, Vancouver, BC, Canada), Rachel Soo (Linguist., Univ. of BC, 100 St. George St., Toronto, ON M5S 3G3, Canada, soorache@gmail.com), and Molly Babel (Linguist., Univ. of BC, Vancouver, BC, Canada)

Sound contrasts are redundantly cued in the speech stream by acoustic features spanning various time scales. Listeners are presented with evidence for a particular category at various temporal intervals, and must coalesce this information into a coherent percept to accurately achieve recognition. Previous work on tone languages has shown that listeners prioritize consonants, then vowels, then lexical tone during phonological and word processing, despite lexical tone being a suprasegmental cue that unfolds with the vowel. We present an online eye-tracking study to assess the time course of Cantonese listeners' recognition of a target word (e.g., 包 /pau͡55/ 'bun') with competitors for rime (北 /pak55/ 'north'), onset (敲 /hau͡55/ 'to knock'), and tone (爆 /pau͡u33/ 'to explode') co-present on the screen. This design allows a test of the role of relative prioritization and contribution of consonant, vowel, and tone information in phonological processing. If vowels are prioritized before tones, we predict increased looking times to tone competitors. If vowels and tones are processed jointly, we predict equal looking times to both vowel and tone competitors. Data collection with Gorilla is ongoing. Data analysis will focus on overall proportions of looking-time to the target and competitors.

4pSC6. Stimulus onset asynchrony effects on perceptual learning. Rachel Soo (Linguist., Univ. of BC, 100 St. George St., Toronto, ON M5S 3G3, Canada, soorache@gmail.com) and Molly Babel (Linguist., Univ. of BC, Vancouver, BC, Canada)

Listeners use lexical knowledge to guide the perception of phonetically ambiguous speech sounds and for retuning phonetic category boundaries. While phonetic ambiguity has many sources, diachronic sound changes represent a more systematic form of phonetic variation. Sound changes may neutralize the phonetic and phonological contrast between two lexical competitors, rendering lexical knowledge in perceptual learning paradigms an unreliable scaffold for boundary retuning, particularly in languages with a high proportion of monosyllabic lexical items (e.g., Cantonese). We present pilot results of an auditory-object-to-picture matching task for perceptual learning of English vowels in monosyllabic words. Seven-step [æ]-[ɑ] continua were created for three minimal pairs (e.g., "gnat"—"knot"). Images corresponding to the real word endpoints were presented at three different stimulus onset asynchrony (SOA) intervals: −250, 0, +250 ms. Participants responded as to whether the visual image matched the auditory token. Preliminary findings indicate that listeners deem the auditory token and picture a congruent match at higher proportions in the 0 ms SOA condition. Our future work analyzes the subsequent learning in response to this exposure method and uses this SOA in a Cantonese perceptual learning study focused on adaptation to lexical tone variation.

4pSC7. Examining the link between speech perception and production through categorical boundaries. Kuniko Nielsen (Linguist, Oakland Univ., Human Health Bldg., 433 Meadow Brook Rd., Rochester, MI 48309-4452, nielsen@oakland.edu)

The link between speech perception and production is one of the central topics in speech perception theories. Newman (2003) examined the correlations between acoustic measures of listeners' perceptual and production prototypes for a given speech category, and found significant correlations for stop consonants (VOT) and for voiceless fricatives (spectral peaks). Shultz *et al*. (2012) also investigated the possible link between perception and production by examining speakers' relative cue weighting of VOT and f0 in word initial stops. Although their results showed no significant correlation between weighting of VOT in production and perception, they reported a trend toward a slightly positive correlation between production and perception for VOT values. The current study extends these findings and aims to explore the perception-production link by examining the relationship between the location of categorical boundaries (between /p/ and /b/) in perception and production. If mental representations for perception and production are directly linked, then speakers with higher VOT values in both categories in production would be expected to also show a higher categorical boundary in perception.

4pSC8. Investigating the perceptual magnet effect in speech perception: An event-related potential study. Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu), Sharon Miller (Audiol. and Speech-Lang. Pathol., Univ. of North Texas, Denton, TX), Xiaojuan Zhang, and Bing Cheng (School of Foreign Studies, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

The "Perceptual Magnet Effect" (PME) is characterized by perceptual warping in favor of stronger assimilation near the prototypical sound against other nonprotypical variants. This ERP study adopted a modified oddball paradigm to examine the PME predictions. The participants were 18 normal adults. The stimuli were computer-synthesized /a/ sounds and their acoustic controls by replacing the formant frequencies with sinewaves. Prior to the ERP sessions, the prototype (P) and nonprototype (NP) were chosen based on behavioral identification and goodness rating. Then 4 variants were created with ±40 mels apart in reference to P or NP, respectively, in either the first formant (F1) or second formant (F2). Mismatch negativity (MMN) responses were derived with the P (or NP) as the standard and the 4 variants as the deviant. The same ERP procedure was used for the nonspeech stimuli. Behavioral results showed that individual subjects consistently rated vowel category goodness based on the F2/F1 ratio. MMN amplitudes in the P and NP conditions showed a pattern consistent with the PME model. This pattern was observed for both speech and nonspeech stimuli. These results provide neural evidence in support of the PME that may extend to general auditory processing of spectral patterns.

4pSC9. Neural processing of phoneme, affect, and gender information in spoken words: A cross-modal priming study. Yang Zhang (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu), Jo-fu Lotus Lin (Inst. of Linguist., National Tsing Hua Univ., Hsinchu, Taiwan), Keita Tanaka (School of Sci. and Eng., Tokyo Denki Univ., Ishizaka, Hiki-gun, Saitama, Japan), and Toshiaki Imada (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Human speech contains not only the linguistic content but also important information about speaker identity and affect. This study employed whole-head magnetoencephalography (MEG) to examine how brain activities were modulated by selective listening of phoneme, affect and gender information with different degrees of task difficulty. The participants were 10 male Japanese adults with normal hearing. The words were 'right' and 'light' recorded from native English speakers and binaurally presented at 50 dB SL. The participants were asked to judge congruency between the visual prime and the spoken word for each trial. The experiment started with a familiarization phase, which was immediately followed by the test phase with 200 trials in each condition. Behavioral results confirmed an increasing order of difficulty from gender to affect to phoneme conditions. Significant priming effects were found only for the affect and gender conditions. In line with the behavioral results, the MEG data revealed distinct patterns of hemispheric and regional involvement and neural oscillatory activities for evaluating the cross-modal congruency in the three conditions. These results demonstrate the neural dynamics and complexity in processing linguistic and paralinguistic information in spoken words with differential influences of language experience.

4pSC10. Neural representations of speech: Decoding bottom-up acoustics and examining top-down effects using electroencephalography. McCall E. Sarrett (Psychol. and Brain Sci., Villanova Univ., Tolentine Hall 334, 800 E Lancaster Ave., Villanova, PA 19085, mccall.sarrett@villanova.edu), Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), and Joseph C. Toscano (Psychol. and Brain Sci., Villanova Univ., Villanova, PA)

The acoustics of spoken language are highly variable, and yet most listeners easily extract meaningful information from the speech signal. Psycholinguistic work has revealed which acoustic dimensions are relevant when listeners categorize speech sounds, and how listeners use higher-level expectations to shift their categorization responses. However, the real-time neural mechanisms subserving such processes are not well understood. Two questions of interest include which perceptual distinctions are detectable in neural responses and whether higher-level information influences perceptual encoding directly. We present three electroencephalography (EEG) studies that address these issues. First, we examine perceptual encoding of speech sounds using machine learning techniques (N = 27). We contrast two approaches with machine learning and EEG and discuss methodological considerations for researchers using such techniques. We find that this approach can reveal neural sensitivity to phonetic contrasts that are indistinguishable in traditional EEG analyses. Second, we examine how top-down influences, such as sentence context (N = 31) or visual information (N = 33), affect perceptual encoding. We show that neural representations of speech sounds are influenced by listeners' context-based expectations in limited cases, specifically, when acoustic cues are ambiguous. We compare how task design may impact linguistic processing, and how more naturalistic tasks may lead to richer processing dynamics.

4pSC11. The influence of sociophonetic knowledge on lexical encoding across dialects. Ellen Dossey (The Ohio State Univ., 108A Ohio Stadium East, 1961 Tuttle Park Pl., Columbus, OH 43210, dossey.1@osu.edu)

The goal of this study was to determine if the strength of lexical encoding differed between regional dialects as a function of the explicit social knowledge listeners held about the dialects or talkers. Participants heard lexical items produced in the Midland, Northern, and Southern dialects of American English in an explicit lexical recognition memory task. Some participants were told the regional backgrounds of the talkers while others were not. It was predicted that encoding would be most robust for tokens produced in the ideologically standard Midland dialect and would be more robust for the socially stereotyped Southern dialect compared with the non-stereotyped Northern dialect. Being told regional information about the talkers was predicted to increase the robustness of encoding. Analysis of response times provided some evidence of a standard dialect benefit across listeners from the three regions, though this effect varied depending on the vowel contained in the stimulus item. Interestingly, being given regional information about the talkers did not increase the robustness with which tokens were encoded; for some listeners, it hindered performance. These results indicate that social knowledge about linguistic varieties can be highly integrated with speech processing, but that it is not facilitative in all situations.

4pSC12. Manipulating listener attitudes toward accented talkers influences speech perception. Melanie Cammarata Philbrick (School of Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL) and Erin Ingvalson (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98105, eingvals@uw.edu)

We previously demonstrated that listeners' attitudes toward talkers with foreign accents is related to listeners' ability to perceive foreign-accented speech. In this study, we sought to determine if listeners' attitudes toward talkers with foreign accents are malleable and, if so, whether changing listeners' attitudes toward talkers with foreign accents would change the strength of the relationship between listeners' attitudes and listeners' speech perception accuracy. Specifically, we hypothesized that supporting listeners' biases regarding non-native speech would lead to better speech perception performance. To test this hypothesis, we created short vignettes that either conformed to or violated listeners' biases regarding non-native talkers. Native English-speaking listeners rated and transcribed speech produced by native and non-native English speakers. Half of the listeners read the vignette prior to rating the speech, half received no information about the talkers; vignettes were randomly paired with talkers. Contrary to our hypothesis, there was no effect of vignette type. Listeners rated non-native listeners as more accented and less likeable when vignettes were present. Interestingly, listeners were also more accurate transcribing speech by both native and non-native talkers when the vignettes were present.

4pSC13. Effect of relative timing of target and background words on speech understanding using the coordinate response measure paradigm. Toni Smith (Psych., Michigan State Univ., 316 Phys. Rd., East Lansing, MI 48824, smit3094@msu.edu), Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA), Gary R. Kidd (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Anusha Mamidipaka, and J Devin McAuley (Psych., Michigan State Univ., East Lansing, MI)

Recognition of target keywords in CRM sentences in the presence of competing CRM sentences has been shown to depend upon both the rhythmic context of target and background speech and F0 differences between target and background speakers. The present study investigated the role of the relative timing of target and background keywords for background keywords that were presented either in isolation (Experiment 1) or embedded within their associated background context (Experiment 2) for target and background material that was spoken by the same male talker. In Experiment 1, onset asynchrony of target and background keywords varied between –200 ms and 200 ms. Results showed an asymmetric U-shaped performance curve where (1) target recognition improved with increasing deviation of background keywords from the expected onset timing of target keywords and (2) target words were better recognized when they began before background keywords compared to when they began after. In Experiment 2, where background keywords were embedded within their original CRM context, performance was reduced to chance for both an intact background rhythm condition and a range of altered background rhythm conditions. Results will be discussed in terms of a selective entrainment hypothesis and the role of F0 differences in speech segregation.

4pSC14. Abstract withdrawn.

4pSC15. Length, speech style, and acoustic characteristics of speech impact perceptions of personality. John Culnan (Linguist., Univ. of Arizona, Communications Bldg., Rm. 109, Tucson, AZ 85721, jmculnan@email.arizona.edu)

People naturally make judgments about those around them when they see others or hear them speak. The present work examines the impact of length of speech, speech style (read speech or conversational speech), and acoustic characteristics of that speech upon the Big Five personality traits (agreeableness, conscientiousness, extroversion, neuroticism, and openness) attributed to the speaker by unknown listeners. Personality trait ratings were collected from 222 listeners who heard audio clips from different speakers producing different lengths of speech in two speech styles. Linear mixed effects models examining length and speech style were then calculated, which demonstrate that short segments of roughly four to five seconds are generally judged differently from longer segments of around 15 s, while speech style only makes a difference in judgments of conscientiousness. Pearson's correlation coefficients reveal that listeners make use of intensity information more than other acoustic features, with intensity measures shown to correlate with conscientiousness, extroversion, and neuroticism. Other correlations of acoustic features with personality ratings demonstrate that extroversion is particularly easy to identify acoustically, whereas openness and agreeableness are more challenging to capture.

4pSC16. Comparing Levenshtein distance and dynamic time warping in predicting listeners' judgements of accent distance. Holly Lind-Combs (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, lind-combs.1@osu.edu), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Rachael F. Holt (Speech and Hearing Sci., Ohio State Univ., Columbus, OH), and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Non-native accent ratings are related to segmental and holistic acoustic deviations from listeners' home accents (Bartelds *et al*., 2020). This study builds on prior work by including both non-ambient native and nonnative accents, using a different perceptual task in which listeners ranked talkers based on their perceived distance from Standard American English, and employing analyses that account for collinearity between distance metrics. Listeners ($n = 52$) completed two rankings where all talkers produced the same sentence and one where each talker produced a unique sentence. Phonemic and holistic acoustic distances between the nine non-ambient accents and Midland American English were quantified using Levenshtein distances and dynamic time warping (DTW), respectively. Results from separate linear mixed effects models showed that both DTW and Levenshtein distances contributed to perceptual distance. Sentence and its interaction with DTW/Levenshtein distance were not significant. The model predicting perceptual rankings from DTW was a better fitting model than the model with Levenshtein distances. Because DTW captures both segmental and suprasegmental distance, it may better predict listeners' similarity judgements among English varieties than a metric only capturing phonemic distance. [Funded by the National Science Foundation (1941691; 1941662) and The Ohio State University Center for Cognitive and Brain Sciences.]

4pSC17. Generalization and specificity in adaptation to unfamiliar speech. Melissa M. Baese-Berk (Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu), Shiloh Drake (Linguist., Univ. of Oregon, Lewisburg, PA), Kurtis Foster (Linguist., Univ. if Oregon, Eugene, OR), Dae-yong Lee (Dept. of Linguist., Univ. of Oregon, Eugene, OR), Cecelia Staggs, and Jonathan Wright (Linguist., Univ. of Oregon, Eugene, OR)

Unfamiliar accents are often harder for listeners to understand than familiar accents; however, previous work has demonstrated that listeners are able to quickly adapt to unfamiliar speech. Further, when listeners are trained on speech from multiple unfamiliar accents, they are able to generalize to a talker with a novel unfamiliar accent. However, tasks used in previous studies are not sensitive enough to determine whether this increased generalization also comes with costs for the specificity of learning. That is, when a listener is trained on multiple talkers and multiple accents, do they demonstrate costs for adaptation to an accent as compared to listeners who are trained on multiple talkers from a single accent? Here, we present data from multiple tasks designed to investigate both whether listeners generalize after exposure to multiple accents, but also whether this more diverse training results in a cost to specificity of learning for a single accent.

4pSC18. How intonation and articulation cues impact gender perception for cisgender and transgender speakers. Brandon Merritt (Speech, Lang. and Hearing Sci., Indiana Univ., 945 Basswood Circle, Bloomington, IN 47403, bmmerrit@iu.edu) and Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Gender identities beyond cisgender male and female are poorly represented in speaker gender perception research. Incorporating transgender speakers may reveal speech characteristics that extend beyond hetero-cis-normative standards and better reflect listeners' real-world experiences and cognitive representations of gender. This study compared how intonation and articulation impact listeners' judgements of speaker gender identity and masculinity/femininity using cisgender male and female, transfeminine, and transmasculine speakers. Cisgender speakers' fundamental and formant frequencies were shifted to gender-ambiguous range and transfeminine and transmasculine speakers to prototypically cisgender female or male range, respectively. Listeners judged speaker gender in conditions that either removed fundamental frequency variation, frequencies above 500 Hz, or both cues. There were nonuniform effects of intonation and articulatory modifications on listener perceptions of gender across speaker groups. However, articulation influenced listener perception more than intonation. When listeners heard articulatory as compared to intonation cues, gender identification scores were generally higher for all speaker groups and masculinity/femininity scores more aligned with cisgender female and transfeminine speakers' gender identities. Although most prior work with transgender speakers has focused on intonation, articulation can substantially impact gender perception and should be considered in gender affirming communication training.

4pSC19. Social priming for nonnative-accented speech. Drew J. McLaughlin (Psychol. & Brain Sci., Washington Univ. in St. Louis, One Brookings Dr., St. Louis, MO 63130, drewjmclaughlin@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

Prior work indicates that a speaker's race/ethnicity can prime a listener to expect native versus nonnative (foreign) accents. In the present study, we replicate the findings of McGowan (2015) and then address novel topics including the effect of social primes on perceptual adaptation. Using a matched-guise design, we examined the effect of race information on the perception of, and adaptation to, Mandarin Chinese-accented English in babble. Subjects randomly assigned an East Asian female's face during the speech perception task showed an advantage over subjects assigned a White female's face or a silhouette (control) face (as measured by transcription accuracy). Performance across trials indicated that although subjects in the control group initially perform worse than those in the East Asian prime group, they perceptually adapt more rapidly to the Mandarin accent, eventually reaching a similar level of performance. Subjects in the White prime group, however, do not adapt as rapidly as those in the control group. In a second experiment, we aim to expand on this work by examining similar priming effects for Arabic-accented English (with Middle Eastern, White, or control primes), and uncommon combinations of minority race/ethnicity primes with nonnative accents (e.g., an East Asian face with an Arabic accent).

4pSC20. Perceptual assimilation of regionally accented Mandarin lexical tones by native Beijing Mandarin listeners. Yanping LI (tMARCS Inst., Western Sydney Univ., Locked Bag 1797, Penrith, New South Wales 2751, Australia, yanping.li@westernsydney.edu.au), Catherine T. Best (MARCS Inst., Western Sydney Univ., Sydney, New South Wales, Australia), Michael D. Tyler (School of Psych., Western Sydney Univ., Sydney, New South Wales, Australia), and Denis Burnham (tMARCS Inst., Western Sydney Univ., Penrith, New South Wales, Australia)

Pronunciations of Beijing Mandarin are those of Standard Mandarin (hereafter Mandarin), which developed from the Beijing dialect. Native speakers of other regional dialects in China learn Mandarin as an early second language, and produce its four lexical tones (level, rising, dipping, and falling) with regional accents. This study investigated how native Beijing Mandarin listeners perceptually assimilate regionally accented Mandarin lexical tones. Native Beijing Mandarin listeners ($M_{\text{age}} = 21.31$ years) were recruited to identify 16 Mandarin real words (4 consonant-vowel syllables: *ba*, *di*, *du*, *gu* $\times$ 4 tones) produced by speakers of Beijing, Shanghai, and Guangzhou dialects and to rate how similar the productions were to the Beijing accent. Stimuli were blocked by syllable, maintaining minimal contrasts between the tones. The Beijing listeners identified Mandarin words in all three accents with high accuracy (>90%), indicating that they reliably assimilated the regionally accented tones into the four Mandarin tones. However, their reaction times were longer and their rating scores were lower for the regional accents than for the Beijing stimuli. This indicates that while Mandarin listeners are sensitive to accent differences in Mandarin lexical tones, they show phonological constancy across regional accents, which is crucial for tone categorization and discrimination of tone contrasts.

4pSC21. Judgments of self-identified gay and heterosexual male speakers of American English: Which vowels do listeners rely on to form their sexual orientation judgments? Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Tracy *et al.*, 2015) demonstrated that, upon hearing a single phone, listeners differentiated between gay and heterosexual male talkers of American English. For instance, listeners relied on seven vowels (e.g., /æ/, /eɪ/, /ɛ/, /iː/, /oʊ/, /ɑ/, and /uː/) to form their judgments. From this particular finding, it is unclear whether the results could be replicated and whether these are the only vowels that listeners rely on; it is possible that listeners might rely on additional vowels. To further explore these issues, the present study examined a wider range of vowels (e.g., /iː/, /ɪ/, /eɪ/, /ɛ/, /æ/, /ɑɹ/, /ɑ/, /ɔ/, /oʊ/, /uː/, /ai/, /aʊ/, /ɔɪ/, /ɪr/, /ˆ/, and /ju/) and, in most instances, presented two different tokens of each vowel to listeners. Some tokens of a vowel (e.g., /iː/, /ɛ/, /æ/, /oʊ/, /uː/, and /ju/) came from a stressed syllable and some tokens came from an unstressed syllable. Only four vowels (e.g., /ɔ/, /aʊ/, /ɔɪ/, and /ˆ/) were presented once. Across all instances of every vowel, participants were able to significantly discriminate between gay and heterosexual speakers. Thus, listeners use a wide range of vowels to form their sexual orientation judgments of male talkers.

THURSDAY AFTERNOON, 2 DECEMBER 2021 501 (L)/504 (O), 1:00 P.M. TO 4:45 P.M.

Session 4pSP

Signal Processing in Acoustics, Physical Acoustics, Animal Bioacoustics, Noise, and Biomedical Acoustics: Signal Processing Methods for Source Classification and Localization in Real Acoustic Environments II

Kay L. Gemba, Cochair
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Invited Papers

1:00

4pSP1. Experimental evaluation of a universal covariance matrix tapered adaptive beamformer. Savas Erdim (ECE, Univ. of Massachusetts, 285 Old Westport Rd., North Dartmouth, MA 02747, serdim@umassd.edu), C. J. Berg, and John R. Buck (ECE, Univ. of Massachusetts, North Dartmouth, MA)

Covariance Matrix Tapers (CMT) widen the beampattern notches of an adaptive beamformer to improve the suppression of moving interferers. If the bearing rate of a moving interferer is known, then the ideal notch width can be found. In practice, the bearing rate is an unknown and often time-varying parameter. To address this challenge, we previously proposed a universal CMT beamformer that blends the array weights across a set of fixed notch width CMT beamformers. This universal beamformer suppressed moving interferers well in simulations. This talk presents experimental results from a 31 microphone uniform line array with a 750 Hz moving interferer at a range of 10 meters. We evaluated two scenarios with the interferer moving at different speeds. In the first experiment, the interferer moved 40 deg in 814 snapshots (approximately 0.05 deg/snapshot) while in the second experiment, the interferer moved 30° in the same time. Both experiments had 25 dB INR at a single sensor. The experiments confirm that the universal CMT beamformer adapts to the best notch width for moving interferers. The results also show that the universal beamformer matches or exceeds the performance of the best fixed notch width CMT beamformer. [Work supported by ONR 321US.]

1:20

4pSP2. Robust direction-of-arrival estimation for a target speaker based on multi-task U-net based direct-path dominance test. Hao Wang, Zhaoyi Gu, Kai Chen, and Jing Lu (Inst. of Acoust., Nanjing Univ., 504 Acoust. Bldg., 22th Hankou Rd., Nanjing 210093, China, lujing@nju.edu.cn)

A dedicated multi-task network for direct-path dominance test has been proposed recently, based on which the robustness of the direction-of-arrival estimation for a target speaker tends to be significantly improved. In this letter, the network is further refined to avoid the original two-stage processing. Moreover, the benefit and generalization of the multi-task network are confirmed by comparison with a single-task network on a significantly larger database. The efficacy of the proposed method in high noisy environments is also validated in real application environments.

1:40

4pSP3. The effect of seabed roughness on passive fathometer measurements. Derek Olson (Oceanogr., Naval Postgrad. School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943, dolson@nps.edu)

The seafloor is rough on a variety of horizontal scales spanning several orders of magnitude. This roughness, together with subbottom roughness and volume heterogeneities can play a role in scattering ambient sound. When the ambient sound pressure field is used to remotely sense the seafloor using passive fathometer techniques, scattering may cause a departure from the plane wave models that are commonly fit to passive estimates of the bottom reflection coefficient. Here, a model for the effect of scattering on passive reflection measurements is developed for an isovelocity water column. Results are shown for a rough half-space, and a rough layered seafloor. The bias introduced into geoacoustic inversions by the presence of seafloor scattering is investigated. [This work was supported by the U.S. Office of Naval Research.]

2:00

4pSP4. Probabilistic focalization for shallow water localization. Florian Meyer (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093, florian.meyer@ieee.org) and Kay L. Gemba (Code 7162, U.S. Naval Res. Lab., Washington, DC)

We consider passive localization and tracking of an underwater acoustic source from measurements provided by a vertical line array. A focalization approach is introduced that probabilistically associates detected directions of arrival (DOAs) to modeled DOAs and jointly estimates the time-varying source location. Modelled DOAs are provided by an embedded ray tracing processor that makes use of environmental parameters to characterize the acoustic waveguide. Due to its statistical model, the proposed Bayesian method can provide robustness against inaccurate environmental parameters as well as an uncertainty characterization for estimated source locations. We demonstrate performance advantages compared to matched field processing using data collected during the SWellEx-96 experiment.

2:20

4pSP5. Signal processing ocean ambient sound for environmental awareness. Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1600 SW 4th Ave., Ste. 260, Portland, OR 97201, siderius@pdx.edu) and John Gebbie (Adv. Mathematics Appl., Metron, Inc., Portland, OR)

Signal processing methods have been developed that use ocean ambient sound to estimate environmental parameters such as the seabed properties. The techniques are straightforward, but in realistic environments there are a variety of complications. The methods considered use a vertically oriented hydrophone line array and practical issues arise such as array motion and deformation. These array parameters are often unknown and represent mismatch with the processing assumptions (e.g., a stationary, straight array). If ignored, performance can degrade significantly so robust signal processing techniques are needed to mitigate. Further, when using ambient sound, as opposed to a sound projector, additional averaging time is needed; this is typically on the order of one or two minutes. Stability in the measurements is essential to obtain the necessary coherent gain. Motion-correcting averaging methods, Doppler correction, adaptive beamforming and supergain are applied to ambient sound processing and will be discussed. Further, data will be analyzed to illustrate the relevant issues and potential remedies in real acoustic environments. Measurements from recent experiments will be presented to illustrate the performance including data collected on the New England Shelf Break in April/May 2021. [Work supported by the Office of Naval Research.]

2:40–3:00 Break

Contributed Papers

3:00

4pSP6. Using a library of shipping sources for data-driven localization of nearby sources. Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., 2788 Defoors Ferry Road, Apt. 325, Atlanta 30318, ndurofchalk3@gatech.edu), Jihui Jin, Justin Romberg (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Underwater source localization is often achieved with a purely model-based approach such as matched-field processing with simulated replica-field. However, such approaches only yield reasonable predictions if the complex and dynamic ocean environment is sufficiently known – often a daunting task. Alternatively, it has been suggested that channel impulse responses (CIRs) estimated from measurements of sources of opportunity (such as commercial shipping vessels) can feed a data-driven approach to source localization that forgoes the need for precise model-parameters [Durofchalk *et al.*, *JASA* **146**(4), 2691–2691 (2019)]. In this presentation, multiple vertical line array (VLA) data from the SBCEx16 experiment conducted in the vicinity of shipping lanes in the Santa Barbara channel (580 m depth, downward refracting profile) are first used to a construct a library of estimated CIRs between selected locations along opportunistic shipping tracks and VLA receivers using ray-based blind deconvolution (RBD) [Byun *et al.*, *JASA* **141**(2), 797–807 (2017)]. Subsequently, this library of data derived CIRs is used to localize other surface sources with traditional

matched-field processing techniques and as training data for a machine learning algorithm. The average localization error and computational efficiency of the different methods are compared.

3:15

4pSP7. An infrasound signal classification method based on non-negative matrix factorization. Zixuan Meng (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, mengzixuan@mail.ioa.ac.cn) and Pengxiao Teng (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Non-negative matrix factorization (NMF) is a principal component analysis method. Previously, several researchers have applied it to the field of infrasound noise reduction. This paper presents a method of combining the results of non-negative matrix factorization with information entropy theory as a signal feature. Firstly, the short-time Fourier spectrum of the original infrasound signal is factorized by NMF, then the entropy of the coefficient matrix's singular value is taken as the signal feature. This feature is combined with several classical features to form a feature vector and input it into the classifier for classification. The classification results show that the proposed NMF singular entropy is an effective signal characterization method.

3:30

4pSP8. Using RMS differentials for robust onset time estimation of impulses. Trevor W. Jerome (NSWCCD, GTWT Water Tunnel, State College, PA 16804, twj115@psu.edu)

Various methods exist for estimating the precise onset time of broadband impulses in a time-domain signal. The method presented here offers a high robustness, with speeds relativley equal to or surpassing currently existing methods. The method is capable of dealing with impulses that have been corrupted by dispersion. Onset time correlates well with the maximum of the lower envelope bound curve of the derivative of the RMS in low-noise signals. In noisy signals, the onset time is found using the same method and a noise correction procedure can be applied for more precision.

3:45

4pSP9. Seabed classification and source localization from ship spectrograms using deep learning models. Jhon A. Castro-Correa (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, 3rd Fl., Newark, DE 19716, jcastro@udel.edu), Christian D. Escobar-Amado, Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

An 18-layer residual neural network (ResNet-18) and a convolutional neural network (CNN) are used for source localization and seabed classification using broadband spectrograms obtained from ship-radiated noise. The propagation of the broadband noise from those ships of opportunity (SOO) contains significant information about the waveguide and seafloor. In this work, a ResNet-18 and a CNN are trained using multi-task learning to estimate the closest point of approach, ship speed, and seabed type. Training data are simulated using a range-independent normal modes model and several sediment profiles obtained by inversions in different regions. The input data samples correspond to 20 min long broadband spectrograms in the 300–1500 Hz band. During the training stage, data augmentation techniques are also applied to evaluate their effects on the predictions. The metrics used to evaluate performance are accuracy for seabed classification and root mean square error for source localization. Results demonstrate the ability of deep learning models to estimate source localization and seabed classification along with the impact of data augmentation techniques using SOO spectrograms. [Work supported by ONR, Contract No. N00014-19-C-2001.]

4:00

4pSP10. Computational Bayesian inference of range and depth of a mobile scatterer in a refractive ocean waveguide with a limited vertical aperture array. Abner C. Barros (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, abarros1@umassd.edu) and Paul J. Gendron (ECE, Univ. of Massachusetts Dartmouth, Dartmouth, MA)

A computational Bayesian method is presented for inference from a limited vertical aperture regarding the range, depth and speed of a submerged large mobile object. Closely spaced multipath arrivals in Doppler and vertical angle off of the target in a refractive environment are addressed in the presence of uncertainty in the ambient acoustic noise power. Vertical angles and Doppler frequencies of the arrival structure are jointly inferred and their posterior density is mapped to the object's range, depth, and speed through acoustic ray interpolation. A case study with the classic Munk sound speed profile is presented to lend credence to the approach.

4:15

4pSP11. Residual physical modeling and semi-supervised source localization. Michael J. Bianco (Univ. of California, San Diego, 9500 Gilman Dr., MC 0238, La Jolla, CA 92037, mbianco@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Source localization in reverberant environments remains an open challenge which ML techniques have shown promise in addressing. Real-world acoustic environments contain irregular boundaries, scattering, and diffracting elements (e.g., furniture and uneven surfaces) which are impractical to model using acoustic propagation software. We develop a hybrid approach to acoustic source localization which utilizes both analytical, signal processing-based localization, and machine learning (ML) to provide a data-driven source localization model. In this approach, the output of an analytical source localization model (e.g., SRP-PHAT or MUSIC) is augmented by a neural network (NN) system. This augmented system consists of a variational autoencoder and localization network, which is designed to correct the analytic estimate. The networks are trained via semi-supervised learning, on both labeled and unlabeled inputs. The NNs in this approach parameterize stochastic functions, which facilitate a statistically principled learning approach via variational inference (VI). The stochasticity also helps quantify source location uncertainty under the VI approximate posterior. For simulated and real acoustic data, the hybrid approach generalizes better and is more efficient than ML or analytic approaches alone.

4:30

4pSP12. Bayesian maximum entropy for geoacoustic inversion using ship-radiated noise. Christian D. Escobar-Amado (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Newark, DE 19716, escobarc@udel.edu), Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

Several acoustic signals from merchant ships, also known as ships of opportunity (SOOs), are employed for geacoustic inversion of the seabed on the New England Mud Patch Area using a Bayesian maximum entropy (BME) approach. The SOOs were recorded by three vertical line arrays deployed at different locations in the Seabed Characterization Experiment in 2017 (SBCEX2017). An effective seabed composed of a mud layer on top of a sandy layer is used for this work. For each SOO, the space of possible hypotheses H is uniformly sampled using Monte Carlo. For each hypothesis in H containing a set of geoacoustic parameters, a SOO spectrogram is simulated using the range independent normal mode model ORCA where the squared error between the measured and simulated SOO is calculated. BME is then applied to compute the marginal conditional posterior probability distribution of the geoacoustic parameters where the probability density function (PDF) provides the current state of knowledge. The inferred PDFs are consistent across all the measured SOOs used for this research and agree with previous geoacoustic inversions in the SBCEX2017 [Work supported by ONR Contract No. N00014-19-C-2001.]

Session 4pUWa

Underwater Acoustics, Signal Processing in Acoustics, Computational Acoustics, Acoustical Oceanography, and Physical Acoustics: Waveguide Invariant Theory and its Applications II

Heechun Song, Cochair
SIO, UCSD, 8820 Shellback Way, Spiess Hall, Rm. 448, La Jolla, CA 92093-0238

Julien Bonnel, Cochair
Woods Hole Oceanographic Institution, 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050

Altan Turgut, Cochair
Naval Research Laboratory, Acoustics Division, Washington, D.C. 20375

Invited Papers

1:10

4pUWa1. Interferometric method of the signal processing in shallow water. Sergey A. Pereselkov (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Russia, Voronezh, Universitetskay Pl. 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Hydrophysics Lab., General Phys. Inst. RAS, Moscow, Russian Federation), and Elena Kaznacheeva (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Voronezh, Russian Federation)

The interferometric signal processing method for localizing a broadband moving sound source in shallow water is proposed and studied theoretically and experimentally in the paper. The field of a moving sound source creates in waveguide a stable interference pattern of the intensity distribution (interferogram) in the frequency-time domain. Sound intensity is accumulated along interference fringes over the observation time. The two-dimensional Fourier transform (2D-FT) is applied to analyze the interferogram. The result of the 2D-FT of the interferogram is called the Fourier-hologram (hologram). The mathematical theory of hologram structure is developed in the present paper. It is shown that the hologram allows to coherently accumulate the sound intensity of the interferogram in a relatively small area as focal spots. The presence of these focal spots is the result of interference of acoustic modes with different numbers. The main result of our paper is a simple relationship between the focal spots coordinates on the hologram and the source range, velocity, and motion direction. The proposed interferometric signals processing method for the source localization is validated using the data of acoustic experiment and numerical modeling. [This research was supported by the RFBR (19-29-06075, 19-38-90326).]

1:30

4pUWa2. Numerical modeling of the sound field interference pattern in the presence of intense internal waves. Sergey A. Pereselkov (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Russia, Voronezh, Universitetskay Pl. 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Hydrophysics Lab., General Phys. Inst. RAS, Moscow, Russian Federation), and Elena Kaznacheeva (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Voronezh, Russian Federation)

The purpose of the paper is numerical modeling of the sound field in the presence of intense internal waves to study the interference pattern in inhomogeneous ocean waveguide. It is assumed that intense internal waves are the reason for the sound field modes coupling and horizontal refraction. The results of sound field modeling are processed to obtain acoustic intensity distributions in the frequency-time domain (interferogram) for unmoving sources. The two-dimensional Fourier transform (2D-FT) is applied to analyze the interferogram. The result of the 2D-FT can be called the Fourier-hologram (hologram). The hologram allows us to coherently accumulate the sound intensity of the interferogram in the narrow area as focal spots. It is demonstrated in the paper that the position of focal spots in the hologram depends on the distance between the source and the receiver. The relationship between the interferogram and hologram structures and the unperturbed and scattered fields parameters is obtained in the paper. The hologram spectral density consists of the two disjoint regions corresponding to the scattered and scattered fields. The filtering of these areas allows us to transmit the non-distorted information through an inhomogeneous ocean environment. The numerical simulation results of interferograms and holograms in the presence of the internal waves are considered. [This research was supported by the RFBR (19-29-06075, 19-38-90326).]

1:50

4pUWa3. Abnormal interference of very low frequency sound field in shallow water. Shengchun Piao (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145, Nantong St., Harbin 150001, China, piaoshengchun@hrbeu.edu.cn), Yang Dong (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), and Zhiqiang Wu (Qingdao Inst. of Marine Geology, China Geological Survey, Qingdao, China)

In a multi-channel seismic exploration experiment in the South Yellow Sea, a tuned air gun array with a shooting interval of 37.5m was used as broadband pulse source, and a horizontal line array with 336 elements is used to receive transmitted signals. The received single shot records in the range-time domain are transformed into an interference pattern in the range-frequency domain using Fourier transform. It is found that there is an abnormal interference pattern of the very low frequency (VLF) sound field in the stratified shallow water environment. It is a complex nonlinear structure, no longer regular striations in the range-frequency domain. The mechanism of this particular interference pattern is interpreted according to normal mode theory. In shallow water stratified leaky waveguide, VLF sound propagates as waterborne modes and bottom-trapped modes. These two kinds of normal mode interfere with each other and form the particular stable interference pattern. In the experimental environment, the group speeds for these two type normal modes change differently with frequency. And it causes slope of interference striations change from positive to negative with increase of frequency. The theoretical simulation results are in good agreement with the experimental data. It is shown that the structure of interference pattern is sensitive to bottom structure and geoacoustic parameters.

THURSDAY AFTERNOON, 2 DECEMBER 2021 — 502 (L)/503 (O), 2:20 P.M. TO 5:25 P.M.

Session 4pUWb

Underwater Acoustics: General Topics in Underwater Acoustics: Modeling and Measurement II

John Gebbie, Cochair
Advanced Mathematics Applications, Metron, Inc., 2020 SW 4th Ave., Suite 170, Portland, OR 97201

Aijun Song, Cochair
Electrical and Computer Engineering, University of Alabama, 245 7th Ave., Tuscaloosa, AL 35487

Contributed Papers

2:20

4pUWb1. The fluid bottom approximation and its implications for sound attenuation in underwater waveguides. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

The fluid-bottom model makes forward and inverse problems more amenable to theoretical analysis and numerical simulations but has some well-known limitations. The accuracy and weaknesses of the fluid approximation are elucidated in this paper by quantifying the effects of weak shear rigidity on acoustic normal modes. Mathematically, appearance of finite shear rigidity and attendant slow shear waves is a singular perturbation of the fluid-bottom model, which standard perturbation theories fail to capture correctly. The problem is addressed here by combination of an asymptotic technique and an analysis of exact solutions for idealized seabed models and placed in the context of earlier studies of the effects of shear rigidity. Due to coupling between compressional and shear waves at the seafloor and within the seabed, shear rigidity affects phase and group speeds of normal modes. By transferring the energy from compressional to shear waves, wave coupling makes a significant contribution to sound attenuation from low up to mid-frequencies. It is found that the effects of shear rigidity on the acoustic field in water are magnified by seabed stratification and especially by density variations. Implications of the results for seabed parameterization in geoacoustic inversions will be discussed. [Work supported by ONR.]

2:35

4pUWb2. Acoustic modeling in the Beaufort Sea using measurements and models of ocean sound speed. Jessica Desrochers (Ocean Eng., The Univ. of Rhode Island, 13 Gilroy St. Apt 2, Newport, RI 02840, jfothergill@uri.edu), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Sarah E. Webster (Adv. Phys. Lab., Univ. of Washington, Seattle, WA), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

Environmental changes in the Arctic over the last few decades have resulted in a subsurface sound speed duct located between 100- and 300-meters depth, known as the Beaufort Duct, which allows for long range acoustic propagation with little or no interference from the ocean surface or bottom. In 2017, two Seagliders traversed between five moored active acoustic sources transmitting linear frequency modulated (LFM) sweeps with frequencies around 250 Hz. During the experiment the Seagliders recorded acoustic arrivals and measured temperature, salinity, and pressure. These environmental measurements are used to calculate sound speed for use as an input to forward acoustic propagation models, including rays and

broadband parabolic equation predictions of acoustic time fronts. Results are compared with acoustic propagation predictions based on sound-speed profiles from ocean models as well as measured acoustic data. The measured and modeled acoustic arrival times along with the calculated sound-speed profiles of the region are used to explore the inverse problem.

2:50

4pUWb3. Ray method for complete sysstem of acoustic equations. Nikolai E. Maltsev (None, 1467 Leaftree cir, San Jose, CA 95131, maltsev_nick@msn.com)

Classic ray theory in acoustics represents approximate solutions of the Helmholtz equation. This report offers a new version of ray theory, derived for a complete system of acoustic equations as it was formulated by Lord Rayleigh for sound pressure and particle velocity. The immediate advantage of this approach is the regular field at caustics.

3:05

4pUWb4. Coherent reflection recovery in forward scattering from random rough surfaces with non-Gaussian correlation functions. Nicholas J. Joslyn (Appl. Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48109, njoslyn@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The coherence between incident and reflected acoustic waves, with wavenumber k, scattered from a random rough surface, with root-mean-square roughness height h, decreases exponentially as $khcos\theta$ increases. The incidence angle, θ, and the surface roughness properties are set by the geometry and characteristics of the environment while the wavenumber range is governed by the signal bandwidth. The frequency-difference autoproduct is a nonlinear field construction capable of recovering reflected-field coherence by downshifting recorded frequencies, effectively decreasing the wavenumber. However, due to the quadratic nature of the frequency-difference autoproduct, dependence on the surface correlation function is introduced. This effect, and other relevant features, appear in the analytic form of the frequency-difference autoproduct derived for Gaussian-distributed surface roughness and surface correlation function. Previous results from simulated isotropic Gaussian surfaces and simple laboratory experiments verify the validity of this form. This presentation provides new results for autoproduct-based coherence recovery in fields scattered from surfaces exhibiting non-Gaussian correlation functions. Coherence recovery is quantified by the coherent surface reflection coefficient (ensemble-average reflected-field amplitude divided by flat-surface reflected-field amplitude), and comparisons are made to the Gaussian-based theory. Experimental results may be provided. [Work supported by ONR and an NDSEG Fellowship.]

3:20

4pUWb5. Using energy flux methods to derive a 3D ocean acoustic propagation model. Mark A. Langhirt (Graduate Program in Acoust., Penn State, 201 Appl. Sci. Bldg., 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, RI), Sheri Martinelli (Appl. Res. Lab., The Penn State Univ., State College, PA), and Daniel C. Brown (Penn State Univ., State College, PA)

The energy flux model is a direct source-to-receiver calculation that avoids the need for finding eigenvalues, but it averages out the modal interference structure as originally derived. It has been shown possible in the two-dimensional model to include the near-neighbor modal interference that resolves large-scale caustic features and shadow zones. So far limited work has been done to develop three-dimensional energy flux models for ocean acoustic propagation, but such a model may have significant computational advantages in complex environments or at high frequencies. We present current progress towards developing three-dimensional underwater acoustic propagation models that have been derived using energy flux principles. In particular we will discuss a vertical normal mode and horizontal energy flux model used to solve for the acoustic propagation in a wedge environment.

3:35

4pUWb6. Rayleigh scattering of a cylindrical sound wave by parallel, infinite cylinders. Alexander B. Baynes (U.S. Navy, Dept. of Phys., Naval Postgrad. School, Spanagel Hall, Rm. 215, Monterey, CA 93943-5216, alex.baynes@gmail.com)

Scattering of sound by a target can be described as a wave radiated by virtual point sources inside the target. In the Rayleigh scattering regime, the strength of the virtual sources can be calculated analytically, and the number of linear sources is finite. When a target is located close to the ocean surface or another reflecting boundary, reflections of the incident and single-scattered waves from the boundary lead to multiple scattering from the target, with the target being insonified by nearby virtual sources. The virtual source method has recently been applied to study low-frequency acoustic fields in the vicinity of cylindrical targets near a pressure release surface or a hard bottom. This method is extended here to study the acoustic field in the near- and far-field of parallel cylindrical targets for arbitrary positions of the sound source. Multiple scattering solutions for scattering from soft, hard, and fluid targets are considered.

3:50–4:10 Break

4:10

4pUWb7. Scale-partitioned differential modeling of large-amplitude acoustic streaming. Jeremy Orosco (Mech. and Aerosp. Eng., Univ. of California San Diego, 13754 Mango Dr. Unit 306, Del Mar, CA 92014, jrorosco@ucsd.edu) and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

Classical techniques for modeling acoustic streaming rely on expansion of the flow about the relative slowness of the streaming velocity in comparison to the particle velocity. Rayleigh pioneered this approach nearly a century and a half ago for deducing streaming equations from the Navier-Stokes governing equations. Modern acoustofluidics applications do not generally admit the order of magnitude separation between flow components that is required in classical "slow streaming" approaches. We outline a theoretical technique that provides greater generality by using drastic spatiotemporal scale disparities to decompose the field equations. We achieve this with scale-partitioned differential operations. The framework is useful in general for modeling continuous nonlinear systems. We first provide brief examples of applying the framework to various media before exploring its use in the area of acoustofluidics specifically. We analyze large-amplitude ("fast") bulk acoustic streaming in a one-dimensional setting. The resulting model explains empirically observed characteristic nonlinear streaming phenomena. From the model, we deduce a variety of bulk streaming properties, including a simple, algebraic upper bound on the maximum achievable streaming velocity in relation to the acoustic forcing. A comparative analysis is undertaken using experimental data from the recent literature.

4:25

4pUWb8. Stochastic sound propagation using dynamically orthogonal parabolic equations. Wael H. Ali (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, whajjali@mit.edu) and Pierre F. Lermusiaux (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Accurate modeling of underwater acoustic propagation is challenging due to the complex ocean physics and acoustic dynamics and the need for resolving the wavelength of the propagating acoustic wave over large distances. These challenges are further amplified by the incomplete knowledge of the ocean environment and the acoustic parameters. These complexities thus lead to many sources of uncertainties in the governing models. In this work, we use our stochastic Dynamically Orthogonal (DO) framework to represent these uncertainties probabilistically in the acoustic Parabolic Equation (PE). These equations optimally represent the dominant uncertainties in the sound speed, density, bathymetry, and acoustic pressure fields. Starting from the governing PE, we derive range-evolution DO differential equations for the mean field, stochastic modes, and coefficients, hence preserving the nonlinearities and capturing the non-Gaussian statistics. The DO equations are implemented for the narrow-angle PE and higher-order Padé

wide-angle PEs and are applied in range-dependent canonical test cases and realistic ocean environments with uncertain source location, source frequency, sound speed, and/or bathymetry fields. We highlight the computational advantages of our framework by comparing it to Monte Carlo predictions and show convergence of the probability density functions as the number of samples and/or modes is increased.

4:40

4pUWb9. A look at reverberation received on a vertical array during the TREX13 experiment. Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com), Brian T. Hefner, Dajun Tang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

The TREX13 Target and Reverberation Experiment was conducted in shallow water off Panama City, Florida. Previously published work focused on reverberation from a fixed horizontal array, which resolves scattering in the azimuthal direction, but has very limited information about the vertical dependence of the scattering. Here, we look at the TREX13 datai—LFM pulses of various bandwidths in the 1.8–3.6 kHz frequency range—received on the vertical line arrays (VLAs) to see what information can be gleaned. The depth and time dependence of the reverberation and echo from a target are investigated. Beams are formed at various vertical angles, which should provide information about the grazing angle dependence of the scattering. A complication arises in determining the scattering strength in non-uniform environments, since the reverberation comes from all azimuths, and since the source and vertical array receivers were not co-located. In such bistatic geometry, the energy arriving at any instant comes from different ranges, azimuths, and vertical angles. However, if the source and receiver are close together, at longer times the geometry approximates the simpler monostatic situation. To help interpretation, results are compared with predictions from a range-dependent bistatic reverberation model. [Work supported by ONR, Ocean Acoustics.]

4:55

4pUWb10. Vessel noise modelling around the Port of Prince Rupert, British Columbia, and the potential effects on marine mammal listening distance. Elizabeth Ramsey (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, AB V8Z 77X8, Canada, elizabeth.ramsey@jasco.com), Graham Warner (JASCO Appl. Sci., Victoria, BC, Canada), and Alexander MacGillivray (JASCO Appl. Sci., Victoria, AB, Canada)

Underwater sound was modelled around the Port of Prince Rupert, British Columbia. The waters around the port region contain major international shipping lanes and important habitat for marine species, many of which require a quiet environment. Underwater noise calculations were carried out using JASCO's Acoustic Real-Time Exposure Model for In-motion Sources (ARTEMIS), a cumulative noise exposure and noise mapping model. Vessel traffic for three Periods were modelled using Automatic Identification System (AIS) vessel position reports from 2019: 1 to 31 January, 15 Jun to 15 July, and 15 September to 15 October. Vessel source levels were modelled using the JOMOPANS-ECHO reference spectrum model (MacGillivray and de Jong, 2021). Ambient noise was modelled using the Wind and Rain Ambient Sound Propagation (WRASP) model (Ainslie, 2010). Sound propagation loss was precomputed using the normal mode code, ORCA (Westwood *et al*., 1996). Vessel traffic projections for 2030 were created with AIS data from 2019 and 2020 to simulate underwater sound levels for 2030. Potential effects on marine mammals was assessed by calculating the relative reduction in distance (listening distance ratios) to which they could communicate, forage, and detect predators due to vessel noise. Listening distance ratios were assessed for 2030 using 2019 as a baseline.

5:10

4pUWb11. Long short-term memory neural network assisted peak to average power ratio reduction for underwater acoustic orthogonal frequency division multiplexing communication. Waleed Raza (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 150001 China, waleedraza93@gmail.com), Xuefei Ma, and Muhammad Bilal (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 150001, China)

The underwater acoustic wireless communication networks are generally formed by the different autonomous underwater acoustic vehicles, and transceivers are interconnected to the bottom of the ocean with battery deployed modems. Orthogonal frequency division multiplexing (OFDM) has become the most popular modulation technique in underwater acoustic communication due to its high data transmission and robustness over other modulation techniques. To maintain the operability of underwater acoustic communication networks, the power consumption of battery-operated transceivers becomes a vital necessity to be minimized. The OFDM technology has a major lack of peak to average power ratio (PAPR), resulting in more power consumption, creating non-linear distortion, and increasing the bit error rate (BER). To overcome this situation, we have contributed our research into three dimensions, firstly, we propose a machine learning-based underwater acoustic communication system through long short-term memory neural network (LSTM-NN). Secondly, the proposed LSTM-NN reduces the PAPR and makes the system reliable and efficient, turning into a better BER performance. Finally, the simulation and water tank experimental data results are executed which proves that the LSTM-NN is the best solution for mitigating the PAPR with non-linear distortion and complexity in the overall communication system

4p THU. PM

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will hold its meeting starting at 4:45 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 are as follows:

Engineering Acoustics (4:45 p.m.)	402
Acoustical Oceanography	Quinault
Animal Bioacoustics	302/303
Architectural Acoustics	Elwha A
Physical Acoustics	402/403
Psychological and Physiological Acoustics	Elwha B
Signal Processing in Acoustics	502
Structural Acoustics and Vibration	301/304

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Elwha A

Committees meeting on Thursday are as follows:

Computational Acoustics	306/307
Musical Acoustics	302
Noise	Elwha A
Speech Communication	Quinault
Underwater Acoustics	Elwha B

Session 5aAA

Architectural Acoustics, Computational Acoustics, Noise, Physical Acoustics, and Education in Acoustics: International Year of Sound Potpourri

Michael Vorläender, Cochair
ITA, RWTH Aachen University, Kopernikusstr. 5, Aachen 52056, Germany

Mark F. Hamilton, Cochair
Walker Department of Mechanical Engineering, The University of Texas at Austin, Austin, TX 78712-1591

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300

Chair's Introduction—8:00

Invited Papers

8:05

5aAA1. International Year of Sound 2020–2021: History and objectives. Marion Burgess (Univ. of New South Wales, Sydney, New South Wales, Australia) and Michael Taroudakis (Mathematics and Appl. Mathematics & IACM, Univ. of Crete, Voutes University Campus, Heraklion 70013, Greece, taroud@uoc.gr)

The International Commission for Acoustics considered that the proclamation of an International Year of Sound (IYS) would create a worldwide awareness of the importance of sound in our world. Sound is an integral part of culture and society from the basic requirements for communication, awareness of our environment, and expression of our culture through to sophisticated scientific and technological research and applications. After a decade of preparation, the ICA decided to organize the International Year of Sound 2020. The moto of the IYS 2020 was "Importance of Sound for Society and the World" and the IYS was associated with the UNESCO Charter of Sound and resolution 39C/59 on the ''Importance of sound in today's world—Promoting best practices." The French Organization "La Semaine du Son" became partner in this initiative. The ICA planned a number of central activities and the majority of the ICA member organisations embraced the opportunity to organize local and regional events during 2020. Of course the arrival of the Covid 19 pandemic curtailed the majority of the planned events but the enthusiasm remained to adapt to virtual activities and develop more resources. In response to the continued enthusiasm the ICA Board agreed the IYS should become a two years celebration. This resulted to the new reference as International *Year of Sound 2020–2021 (IYS 2020–21)*

8:25

5aAA2. International Year of Sound 2020–2021: Main events and achievements. Marion Burgess (Univ. of New South Wales, Sydney, New South Wales, Australia), Michael Taroudakis (Mathematics and Appl. Mathematics & IACM, Univ. of Crete, Voutes University Campus, Heraklion 70013, Greece, taroud@uoc.gr), Michael R. Stinson (none, Carlsbad Springs, ON, Canada), and Jean-Dominique Polack (Inst. D'Alembert, Sorbonne Univ., Paris, France)

The International Year of Sound 2020–2021 includes activities organized by the IYS Steering Committee, ICA member societies, international affiliates and individual organizations. The web page www.sound2020.org, is the primary contact for the IYS and receives more than 3500 visitors per month. The opening was held in Paris at the Grand Amphitheatre of the Sorbonne on 31 January 2020. A film "Sounds of our Life" was launched and is also available on Youtube. An international student competition involved over 15 000 students and attracted excellent contributions. The international scientific conferences included outreach and promotions for the IYS. Despite the restrictions due to COVID-19, ICA members maintained their enthusiasm and contributed an amazing range of innovative virtual activities and resources. Additional outreach included various media items and podcasts. The IYS has encouraged further collaboration with institutions including the WHO, the CHC and groups dealing with sound. This will eventually will strengthen the acoustics community into the future. Information on the activities and all the resources are freely available from the website and will remain as a legacy of the IYS. Despite the challenges of 2020 and 2021, the efforts of so many have shown the resilience and innovation of the acoustic community to enhance the understanding worldwide of the importance of sound.

8:45

5aAA3. Acoustical Society of America and the International Year of Sound. Keeta Jones (Acoust. Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11747-4300, kjones@acousticalsociety.org)

In this presentation, I will discuss how the Acoustical Society of America (ASA) celebrated the International Year of Sound. To stimulate the understanding and awareness of the important role that sound plays in all aspects of society, ASA produced short acoustics-themed videos, offered virtual acoustics demonstrations, presented online speakers, arranged a special issue of Acoustics Today, hosted a Twitter chat for the Journal of the Acoustical Society of America (JASA) and more. ASA education and outreach efforts will continue to improve and motivate the public to explore sound.

9:05

5aAA4. IYS Outeach through SPS Science Outreach Catalyst Kits (SOCKs). Noah Johnson (Society of Phys. Students, American Inst. of Phys., South Orange, NJ), Holly Fortener (Society of Phys. Students, American Inst. of Phys., Houston, TX), and Brad Conrad (Society of Phys. Students, American Inst. of Phys., 1 Phys. Ellipse, College Park, MD 20740, bconrad@aip.org)

Explore some of the outreach possibilities around the International Year of Sound through the 2021–2022 Society of Physics Student Science Outeach Catalyst Kit (SPS SOCK). SOCKs, are free to ASA and SPS chapters (while supplies last) and contain exploratory acoustic and physics activities that are specifically designed for undergraduate and graduate students to use in outreach presentations to elementary, middle and high school students. Through a grant from AIP, this year's SOCK kits come with a complete set of materials for students host an engaging and educationally effective outreach table aim at the general public and students alike. Explore the variety of ways we experience sound and music through several complete setups that are designed to expand a chapter's outreach capabilities and become tried-and-true outreach activities. This years kits are designed for chapters just starting out or those who are demo pros. This SOCK was designed and implemented by the SPS National interns, ASA Staff, ASA volunteers, and the SPS National office staff. Come learn about this kit and how it can help your group get everyone excited about sound: www.spsnational.org/programs/outreach/science-outreach-catalyst-kits

9:25

5aAA5. Celebrating sound in *Physics Today*. Christine Middleton (Phys. Today, American Inst. of Phys., 1 Phys. Ellipse, College Park, MD 20740, cmiddleton@aip.org)

To recognize the International Year of Sound, *Physics Today* devoted the feature articles section of its December 2020 issue to highlighting areas of acoustics research. My talk will cover how the special issue came together as well as the magazine's coverage of acoustics in general.

9:45

5aAA6. Strengthening the link between the ASA and the Spanish-speaking community. Ana M. Jaramillo (Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu) and Zachery O. L'Italien (McKay Conant Hoover, Inc., Westlake Village, CA)

This presentation will discuss the efforts of the Spanish Speaking Acousticians in the Americas regional chapter over the past two years of the International Year of Sound to foster stronger connections between the ASA and the Hispanic community. An overview of presentations, events, collaborations, and other activities will be shared, as well as commentary on current challenges for acousticians in Latin America, recent progress, and how the ASA and the SSA can serve as an invaluable resource for the region.

10:05–10:20 Break

10:20

5aAA7. Making waves in Elkhart for the international year of sound. Jay B. Brockman (College of Eng., Univ. of Notre Dame, 257 Fitzpatrick Hall, Notre Dame, IN 46556, jbb@nd.edu)

Making waves is a unique partnership between K–12 and college educators, Grammy-winning musicians, and a small woodworking business that offers educational programs that blur the lines between the STEM fields and music to bring out the best in students' abilities to experiment, create, analyze, and explore. Using topics drawn from grade-level standards in engineering, math, and science, as well as best practices in music education, students build and analyze custom musical instruments from kits and then compose and perform original works using those instruments. During the 2020–2021 school year, with COVID-19 restrictions in place, Making Waves moved its afterschool program with Elkhart, Indiana elementary and middle school students into a hybrid online learning mode, with instructors from across the country working remotely with Elkhart students in their classrooms. In celebration of the International Year of Sound, the team produced an arrangement of "The Sound of the World," with a video highlighting student learning and the musical heritage of the Elkhart community. This presentation will feature this video, along with audience participation activities.

Contributed Papers

10:40

5aAA8. A jumping-off point—Sound and listening as means of contemplating our world. Carl P. Giegold (Threshold Acoust. LLC, 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, cgiegold@thresholdacoustics.com) and Scott Pfeiffer (Threshold Acoust. LLC, Chicago, IL)

Our connection to the World through our ears is spatial, temporal, and emotive. In this way, it is no different from our sense of sight, but the parallels and capabilities are inadequately celebrated. The Year of Sound was an opportunity to do just that, and it caused us to ponder ourselves as part of the Universe's way of knowing itself. We thought in terms of boundaries, identity, protest, time and anachronism, uniqueness of place, the extension of ourselves to places and points of view we cannot reach in person. This talk presents some of those musings, for the simple pleasure of it.

10:55

5aAA9. The Detroit Orchestra Hall "Centennial Sound Lab" multi-location binaural acoustic measurement and audition project. Wade R. Bray (HEAD Acoust., Inc., 6964 Kensington Rd., Brighton, MI 48116, wbray@headacoustics.com)

Detroit Orchestra Hall, opened in October 1919, entered its centennial year in October 2019 with a variety of special events. A Board member, a recently retired automotive sound quality engineer, suggested starting the celebration with a comparative audition study of the Detroit Symphony Orchestra in Orchestra Hall with acquisition and analyses "using the tools of Sound Quality engineering." HEAD acoustics, Inc., was contacted by Symphony management and data were acquired during a dress rehearsal of the Brahms Fourth Symphony using five Aachen HEAD Measurement Systems (HMS): four at various seats in the Hall and one onstage behind the brass section. Analysis included conventional, psychoacoustic and advanced psychoacoustic tools including the Hearing Model of Sottek. One revealing analysis was an animatable display of left and right ear levels and loudnesses as XY plots, the left ear plotted on the vertical axis and right on the horizontal. In collaboration with Jaffe Holden Acoustics, Orchestra Hall acousticians, two presentations were made: one to the public as part of a roundtable discussion in the Hall, the other to Board members and donors including calibrated binaural playback in the Hall with ability to "jump" seamlessly during playback from any recorded seat to any other.

Invited Papers

11:10

5aAA10. UK Acoustics Network and International Year of Sound. Kirill V. Horoshenkov (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield S10 2TN, United Kingdom, k.horoshenkov@sheffield.ac.uk)

In 2017, the UK's Engineering and Physical Science Research Council (EPSRC) funded a project to create the new Acoustics Network of research active acousticians in the UK (UKAN). The main aim of UKAN was to bring together researchers working in different areas of acoustics to enhance communication between groups, provide a focus for collaboration and innovation, and to maximise the future impact of acoustics-based research in the UK. UKAN works closely with the UK Institute of Acoustics and other professional bodies in the UK and overseas to promote the wider importance of acoustics and to raise its profile nationally and internationally via a range of research exchanges, workshops and public engagement events. The duration of the UKAN grant overlapped well with the International Year of Sound (IYS), its representatives took active part in the IYS Launch Event in January 2020 and continued to support IYS activities throughout 2020/21 despite of the pandemics. The successful UKAN's operation paved the foundation for securing funding for UKAN + EPSRC grant that started in April 2021. This paper presents a summary of the accomplishments by UKAN/UKAN + and its wider role in planning and supporting of the activities related to the International Year of Sound.

11:30

5aAA11. The International Year of Sound (IYS2020+) in Spain. Antonio Pedrero (Sociedad Española de Acústica, Serrano 144, Madrid 28006, Spain, presidente@sea-acustica.es), Antonio Pérez-López, Antonio Calvo-Manzano, and Ana Delgado-Portela (Sociedad Española de Acústica, Madrid, Spain)

The International Year of Sound (IYS2020 +) is an international initiative aimed at raising awareness in society about the importance of sound in everyone's life. To this end, the International Commission for Acoustics has requested that activities be scheduled in all countries of the world. In addition, it has organized an international student contest in order to involve the youngest in this cause. Unfortunately, a few days after this initiative was officially inaugurated, the COVID-19 pandemic was declared in the world, forcing the reformulation or even cancellation of many of the scheduled activities. In this communication a general balance of the development of the IYS2020+ in Spain is made. The dissemination work of this initiative carried out by the Spanish Acoustics Society is described. In addition, the most notable activities carried out are discussed, as well as the participation of Spanish schools in the student contest.

11:50

5aAA12. The International Year of Sound: A worldwide student competition. Chiara Bartalucci (Vie en.ro.se. Ingegneria s.r.l., Viale Belfiore 36, Florence 50144, Italy, chiara.bartalucci@vienrose.it), Sergio Luzzi, Paola Pulella, Sara Delle Macchie, and Rossella Natale (Vie en.ro.se. Ingegneria s.r.l., Florence, Italy)

In the frame of the International Year of Sound (IYS), the global initiative recognized under the UNESCO Charter of Sound No. 39C/59, a Competition for students from around the world on the theme of "My World of Sounds" has been organized by the International Commission for Acoustics. Primary and middle school students produced and sent drawings, images, patchwork, collages and similar related to their world of sounds. High school students wrote and sent verses of the song entitled "The sound of the world." By 30 April 2021, respectively, 39 items for the first Competition category and 33 items for the second one have been sent to the Competition

office, as national finalists from overall 16 countries. An experts' jury expressed its preferences about the items answering to three questions concerning coherency, originality, inspiration and a web jury voted the preferred items on the official IYS facebook page. The IYS Competition office has then worked to combine votes from both juries and to declare the winners. As final output, a short movie and a long one will summarize the IYS story, its global initiatives and they will focus on the Student Competition.

FRIDAY MORNING, 3 DECEMBER 2021 QUINAULT, 9:00 A.M. TO 10:15 A.M.

Session 5aABa

Animal Bioacoustics: Animal Auditory Perception

Rolf Müller, Chair
Mechanical Engineering, Virginia Tech, ICTAS II, 1075 Life Science Cir, (Mail Code 0917), Blacksburg, VA 24061

Contributed Papers

9:00

5aABa1. Development of a tension-controlled soft-robotic actuation system for a biomimetic bat robot. Sanmeel Vijay Lagad (Mech. Eng., Virginia Tech, 445 Goodwin Hall, 635 Prices Fork Rd. - MC 0238, Blacksburg, VA 24061, sanmeelvijay@vt.edu), Ibrahim M. Eshera (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Sounak Chakrabarti, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

It has been observed that bat species such as the horseshoe bats (Rhinolophidae) that are capable of biosonar-based navigation in dense vegetation deform their noseleaves and pinnae during echolocation. To investigate the impact of these deformations on the encoding of sensory information in the biosonar echoes, a biomimetic robot has been developed to replicate this unique peripheral dynamics. Horseshoe bats have about twenty muscles on each pinna and demonstrate a considerable amount of variability in the motions of these structures. To replicate even a small portion of this flexibility and variability, an actuation mechanism with a small footprint on the deforming structure is necessary. In a prior version of the biomimetic robot, soft-robotic pneumatic actuators were used, but the size of these devices limited their number per pinna and relative orientation. To alleviate these issues, a tension-controlled actuation concept has been developed that consists a bank of servo motors that are connected to the pinna via tendons. At present, this system is designed for five degrees of freedom per pinna and three for the noseleaf. Ongoing work seeks to address how the tension-based actuation system can be controlled to recreate the variability in the noseleaf and pinna deformations of bats.

9:15

5aABa2. Abstract withdrawn.

9:30

5aABa3. *Achroia grisella*'s ear as inspiration for acoustic sensing. Lara Díaz-García (Ctr. for Ultrasonic Eng., Univ. of Strathclyde, 99 George St., Glasgow G1 1RD, United Kingdom, lara.diaz-garcia@strath.ac.uk), Andrew Reid, Joseph Jackson, and James Windmill (Ctr. for Ultrasonic Eng., Univ. of Strathclyde, Glasgow, United Kingdom)

Bio-inspiration, gathering insight from looking at nature, can be useful when trying to solve challenges innovatively. Insects are forced by their body size and the energy cost of a hearing organ to make many clever adaptations for hearing. *Achroia grisella* is a nocturnal moth with a hearing sense. It uses it both for avoiding predators and mating, the latter being an unconventional role among moths. The moth is capable of directionality that seems to be monoaural and dependent exclusively on morphology. We develop a model with the goal of 3D printing a structure whose behavior is similar to that of the moth ear, i.e., directionality at a low design cost. Starting from a simple circular plate model, we progressively incorporate more complex elements to improve the resemblance to the natural system. Equations, simulations, and 3D printed devices' frequency responses (measured through Laser Doppler Vibrometry) are compared. The directionality of the printed devices is analyzed.

9:45

5aABa4. The perception of ultrasonic signals in quiet and noise by mice exposed to noise. Payton Charlton (Psych., Univ. at Buffalo, Park Hall Rm. B80, 211 Mary Talbert Way, Buffalo, NY 14260, paytonch@buffalo.edu), Kali Burke (Psych., Univ. at Buffalo, Baltimore, MD), Anastasiya Kobrina (Psych., Univ. at Buffalo, Flagstaff, AZ), Amanda M. Lauer (Dept. of Otolaryngol. - Head and Neck Surgery, Johns Hopkins Medical Univ., Baltimore, MD), and Micheal Dent (Psych., Univ. at Buffalo, Buffalo, NY)

High-intensity noise exposures can lead to temporary or permanent increases in hearing thresholds for simple and complex stimuli. Mice produce complex signals known as ultrasonic vocalizations in a variety of behavioral contexts and it is thought these vocalizations serve a communicative function. The effects of a 2-h exposure to 8–16 kHz narrowband noise at 100 dB SPL on auditory sensitivity thresholds for two types of ultrasonic vocalizations (complex and downsweep vocalizations) in two different listening conditions (quiet and continuous 60 dB white noise) were measured. Male and female mice ranging from 100 to 1000 days old were trained to detect the vocalizations in a go/no-go operant conditioning paradigm with positive reinforcement. Each mouse participated in only one condition. Once consistent thresholds were obtained for a mouse, they were subjected to the noise exposure. Post-noise thresholds were obtained on a daily basis for up to 120 days. Similar to humans exposed to a traumatic noise, there was a lot of variability in the extent of the hearing loss following the noise exposures, with some mice experiencing large threshold shifts and other mice exhibiting only minor, temporary hearing loss. [Work supported by NIH DC016641.]

10:00

5aABa5. Autonomy, soft-robotics, deep learning, and bat biosonar. Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, rolf.mueller@vt.edu), Sounak Chakrabarti (Mech. Eng., Virginia Tech, Blacksburg, VA), Ibrahim M. Eshera (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Sanmeel Vijay Lagad, Ruihao Wang (Mech. Eng., Virginia Tech, Blacksburg, VA), and Liujun Zhang (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA)

Echolocating bat species that inhabit dense vegetation are promising model systems for achieving autonomy in complex natural environments. To replicate the skills of the bats in extracting information about complex environment that can support autonomous navigation in a man-made system three aspects deserve particular attention: (i) encoding of relevant sensory information; (ii) information extraction; and (iii) integration of information encoding and extraction with each other and the respective context. To facilitate progress towards understanding these issues, a biomimetic sonar system is being developed that is aimed at recreating the flexibility and variability that bats exhibit in the configuration of the emission and reception baffles of their biosonar systems (i.e., noseleaves and pinnae). Analyzing the significance of these dynamic effects on sensory information encoding requires an understanding of the stimulus ensemble of biosonar tasks in natural environment. To this end, the biomimetic sonar system has been used to collect large data sets with echoes that pertain to fundamental navigation tasks such as localization and passageway finding. Deep-learning methods have been shown to be a good match for analyzing this echo data. Finally, ongoing research is directed at data acquisition and inference with the specifics of a given habitat and biosonar task.

FRIDAY MORNING, 3 DECEMBER 2021 QUINAULT, 10:30 A.M. TO 11:45 A.M.

Session 5aABb

Animal Bioacoustics: Animal Bioacoustics in Noise

Elizabeth T. Küsel, Chair
Portland State University, 8630 Fenton Street, Suite 218, Silver Spring, MD 20910

Contributed Papers

10:30

5aABb1. Exposure radial distances. Elizabeth T. Küsel (JASCO Appl. Sci., 8630 Fenton St., Ste. 218, Silver Spring, MD 20910, Elizabeth.kusel@jasco.com), Michelle Weirathmueller, and David Zeddies (JASCO Appl. Sci., Silver Spring, MD)

During offshore construction activities that produce underwater sound, mitigation zones are established to define protective monitoring areas for fauna in the vicinity and to establish whether animals may be exposed to sound levels exceeding regulatory criteria. Traditionally, mitigation zones have been determined by estimating the range within which the sound levels remain above given thresholds. These statically defined "acoustic ranges" have drawbacks, especially for criteria based on cumulative metrics such as SEL for marine mammal injury. A threshold distance derived solely from the acoustic footprint of a source is inadequate in establishing whether an animal moving through a (possibly time-varying) sound field would exceed a time-dependent exposure criterion. An approach to defining mitigation zones based on an understanding of the animals' movement relative to the sound source overcomes the limitations of the static range calculation. The radial distance from a source that accounts for the exposures above threshold, or 'exposure range', can be estimated statistically from path-dependent sound exposure estimates generated through animal movement modeling. Exposure ranges can be calculated by this method also for moving and multiple sources. We present example scenarios comparing acoustic and exposure ranges in the context of seismic surveys and pile driving activities.

10:45

5aABb2. A tale of two singers: how do bats and bird mixed-flocks respond to petroleum industry noise in the Ecuadorian Amazon. José L. Rivera-Parra (Petroleum Dept., Escuela Politécnica Nacional, Ladrón de Guevara 253, Quito 170517, Ecuador, jose.riverap@epn.edu.ec), Pamela Rivera-Parra (Biology Dept., Escuela Politécnica Nacional, Quito, Ecuador), Christian Vasconez, Álvaro Dueñas-Vidal (Phys. Dept., Escuela Politécnica Nacional, Quito, Ecuador), and Luca Sorriso-Valvo (Istituto per la Scienza e Tecnologia dei Plasmi, Quito, Ecuador)

Industrial noise can have a significant impact on animal groups that rely on acoustic communication for fundamental survival activities. Our research focuses on two of these groups: insectivorous bats, which use ultrasound to navigate and find food; and mixed flocks of insectivorous birds, that rely on vocalizations to communicate about foraging direction and potential threats. We conducted *in situ* experiments in the Yasuni Biosphere Reserve in the Ecuadorian Amazon to understand the immediate and long term effects of industrial noise on behaviour, habitat use and local diversity. We characterized bats and birds vocalizations, and the industrial noise profile, both in ultrasound and audible frequencies. Experiments were performed using pure tone sounds representative of the industrial noise acoustic profile. Moreover, experiments included behavioral response characterization and acoustic transmission modeling. Our results suggest, in both groups, the immediate response is avoidance. In the long term, the effects included a change on habitat use, and modification on community composition. These effects are proportional to how similar is the noise to the frequencies commonly used by the animals and how much noise the habitat itself absorbs. Modeling and technology development are the next steps to further understand these effects and how to mitigate them.

11:00

5aABb3. Noise elicits the Lombard effect in midshipman mating vocalizations. Francis Juanes (Dept. of Biology, Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, juanes@uvic.ca), Nicholas Brown (Univ. of Victoria, Victoria, BC, Canada), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), and Sigal Balshine (McMaster Univ., Hamilton, ON, Canada)

Anthropogenic noise pollution is an emerging global threat to fish populations. Among a suite of deleterious effects, noise can potentially impede reproductive success in some fishes by masking their mate advertisement vocalizations. Using the plainfin midshipman fish (*Porichthys notatus*), a marine toadfish that produces a distinctive "hum" during courtship, we investigated how noise affects male vocalizations and spawning success in the wild. We recorded nesting males for three days and measured the frequency, amplitude, and duration of their vocalizations before, during, and after exposure to artificial noise (a c. 118-Hz tone). We also counted eggs in nests exposed to 10 days of artificial noise versus control nests that were not exposed to artificial noise. Males exposed to noise reduced the number of vocalizations they produced, reduced the frequency of their vocalizations, and increased the amplitude of their mating hum (Lombard effect). However, chronic noise exposure did not clearly affect spawning success, suggesting that the Lombard effect allowed males to sustain clear advertisement signals when competing with the relatively weak artificial noise source.

11:15

5aABb4. The impact of different forms of noise on female mate choice in the Australian black field cricket (*Teleogryllus commodus*). Jessica Briggs (Dept. of Biological Sci., Univ. of New Hampshire, 6 Stark Ave., Dover, NH 03820, jb1424@wildcats.unh.edu) and Daniel Howard (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

The increasing prevalence of anthropogenic noise presents a new selection pressure that is rapidly changing the acoustic landscape in which animal signals have adapted. Increasing noise levels can impact an animal's ability to use acoustic cues and signals to properly assess their environment or recognize, localize, and evaluate a potential mate, leading to decreases in fitness and survivorship. Anthropogenic noise outside the range of the human sensory system is often overlooked. Therefore, it is important to understand how noise in multiple modalities may disrupt informed decision making. Crickets can detect airborne noise in the low and high frequencies as well as substrate-borne vibrations and rely on detecting airborne calls to successfully assess potential mates. A systematic approach was used to test the behavioral response of Australian black field crickets (*Teleogryllus commodus*) under variable noise conditions, with the aim of understanding how female choice is influenced by multisensory noise. Playback trials in the laboratory were performed to examine how noise in all three forms influenced female mate choice. This study will expand our knowledge of the effects of anthropogenic noise on an understudied taxa and highlight why we need to consider the animal's sensory system sensitivity when studying noise.

11:30

5aABb5. Predictable variability in humpback whale (*Megaptera novaeangliae*) song acoustics and potential masking of songs by anthropogenic noise. Christina E. Perazio (Univ. at Buffalo, SUNY, Buffalo, NY 14260, cperazio@buffalo.edu), Madison Dolan, Ashley Johnston, Sidney Lyytikainen (School of Social and Behavioral Sci., Univ. of New England, Biddeford, ME), and Eduardo Mercado (Univ. at Buffalo, SUNY, Clarence, NY)

Past analyses of humpback whale songs emphasize the flexibility with which whales change their songs across years. Acoustic measurements of song provide a way to systematically assess changes in songs and to predict humpbacks' ability to adapt to changing soundscapes. Singers concentrate energy at predictable frequencies while continuously adjusting song acoustics. Analyses of songs from the Southeastern Pacific and Central North Pacific stocks (2013–2019) were used to characterize variations in peak frequencies. Preliminary results showed that humpbacks consistently concentrated energy in 3–4 frequency bands, suggesting that this is a universal feature of song production, and that singers may be less flexible in their ability to modify songs than has been assumed. Frequency use was also compared across different locations that vary in boat traffic. Initial findings revealed a negative association between peak frequency and the "noisiness" of a location. Examining patterns of variability in song acoustics may reveal an impact of anthropogenic noise on singing behavior. Predicting how increases in ocean noise may affect song is critical because whales are ecosystem engineers on a global scale. The extensive geographic and temporal coverage of these analyses adds a novel piece to this conservation puzzle.

Session 5aAO

Acoustical Oceanography and Underwater Acoustics: Topics in Acoustic Oceanography

Scott Loranger, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 86 Water Street, Woods Hole, MA 02543

Marina Yarina, Cochair
Marine Geoscience, University of Haifa, Abba Khoushy 199, Haifa 3498838, Israel

Contributed Papers

8:00

5aAO1. Environmental drivers of the low-frequency ambient noise on the Chukchi Shelf. Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Bazile Kinda (Service Hydrographique et Océanographique de la Marine, Brest, France), and Daniel P. Zitterbart (Woods Hole Oceanographic Inst., Woods Hole, MA)

This study presents an analysis of the low frequency (250–350 Hz) ambient noise (after individual transient signals are removed) recorded on the Chukchi shelf during the CANAPE experiment from October 2016 to July 2017. Ambient noise drivers are evaluated by correlating noise levels with external oceanographic data including local wind speed, local ice coverage, and distant ice drift magnitude. Also, the experimental area is known to be home of the Beaufort duct, a warm layer intrusion which creates an acoustic channel that favors long range propagation and impacts the ambient noise. The study shows that, as usual in polar areas, the ambient noise is quieter and wind force influence is reduced when the sea is ice-covered. More interestingly, the study demonstrates that the ambient noise drastically reduce (up to 10 dB/Hz) when the Beaufort duct disappears. It also shows that a large part of the noise variability is driven by distant cryogenic events. [Work supported by the Independent Research & Development Program at WHOI and by the Office of Naval Research.]

8:15

5aAO2. Estimation of gassy sediment parameters in shallow water using shipping noise signals in frequency domain near cut-off. Marina Yarina (Marine Geoscience, Univ. of Haifa, Abba Khoushy 199, Haifa 3498838, Israel, myarina@campus.haifa.ac.il), Andrey Lunkov (Prokhorov General Phys. Inst., Russian Acad. of Sci., Moscow, Russian Federation), and Boris Katsnelson (Marine Geosciences, Univ. of Haifa, Haifa, Israel)

Modal structure of shipping noise in a shallow water waveguide is studied experimentally and theoretically. Experiments were carried out in the Sea of Galilee (Lake Kinneret, Israel) having maximal depth of 40.5 m and remarkable concentration of methane bubbles in the upper sedimentary layer. Receiving system consisted of 27-m vertical line arrays (VLAs) with ten hydrophones at each. They spanned a lower part of the waveguide with interval ~3 m. The VLAs were fixed in the centre of the lake with the distance ~40 m between them. R/V "Hermona" was moving along a straight line connecting VLAs at the range of up to 1 km from the arrays. Modal filtration technique on two synchronized VLAs with a phase correction was proposed to extract complex eigenvalues of each mode, including modal attenuation, which strongly increase in the proximity of cut-off frequency for all normal modes. Based on frequency dependences of attenuation coefficients of lower modes, sound speed in gas-saturated bottom layer was estimated. For application of this technique, there is no need to know exact coordinate and speed of noise source, which seems to us to be an advantage of the method. [Work was supported by RFBR, Grant No. 20-05-00119.]

8:30

5aAO3. Acoustic scattering by elastic cylinders: Practical sonar effects. Miad Al Mursaline (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 70 pacific St., Cambridge, MA 02139, miad@mit.edu), Timothy K. Stanton, Andone C. Lavery, and Erin M. Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic scattering from elastic cylinders has been studied extensively over the last few decades. However, the vast majority of theoretical investigations have focused on idealized plane wave, point receiver solutions, ignoring realistic sonar effects. Acoustic transducers are characterized by spherically spreading waves and often have transmit and receive beams that are directional. Because of the spreading, even during broadside measurements, these sonars excite cylinders obliquely. This investigation involves acoustic scattering by spot-insonified infinitely long elastic cylinders, and studies effects caused by practical sonar constraints both theoretically and experimentally. Calibrated broadside measurements of backscattering are presented for various cylinder and source/receiver geometries including both monostatic and near-monostatic configurations. In all measurements, the ratio of the footprint length of the target relative to the first Fresnel zone diameter is chosen such that the entire first Fresnel zone dominates the scattering. This ensures that the scattering from the cylinder spreads cylindrically and hence the cylinder behaves like an infinite target. A theoretical formulation is developed to account for beam spreading and directivity of practical echo-sounders. Comparisons between theoretical predictions and experimental data are presented. [Work funded by ARPA-E.]

8:45

5aAO4. Spatial variation in acoustic field due to submarine melting in glacial bays. Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, Singapore), Grant B. Deane (Scripps Inst. of Oceanogr., Code 0238, UCSD, La Jolla, CA 92093-0238, gdeane@ucsd.edu), Mandar Chitre (Acoust. Res. Lab., National Univ. of Singapore, Singapore), Oskar Glowacki (Inst. of Geophys., Polish Acad. of Sci., Warsaw, Poland), M. Dale Stokes (Scripps Inst. of Oceanogr., La Jolla, CA), Mateusz Moskalik (Inst. of Geophys., Polish Acad. of Sci., Warsaw, Poland), and Hayden A. Johnson (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Climate-change induced melting is leading to accelerating ice loss at tidewater glaciers worldwide. A significant component of the freshwater flux from these glaciers arises from submarine melting at the glacier-ocean interface. This melting causes a distinct acoustic signature due to the release of pressurized bubbles underwater, opening up the possibility of monitoring this phenomenon on a large scale using passive acoustic systems. To evaluate the use of sound in monitoring submarine glacier melting, we made acoustic measurements using vertical hydrophone arrays in four glacial bays in Svalbard in 2019. As the recording array was moved away from the glacier, the variation in the recorded acoustic level due to melting ice did not follow a uniformly decreasing trend as one might expect. Moreover, the

acoustic intensities at different glaciers were clustered at different levels. These observations indicate that the geometry of the glacier-ocean interface, thermohaline structure of the underwater channel and presence of floating ice in the bay played a role in determining the acoustic field. Most of this variation can be explained through propagation modeling. Moving forward, this model-based interpretation of the field will play an integral part in inverting the sound to estimate the submarine melt rate.

9:00

5aAO5. Modeling Chirp SONAR returns due to random scatterers in sediment volum. Dajun Tang (Appl. Phys. Lab, Univ. of Washington, Seattle, WA, dtang@uw.edu)

The goal is to simulate time domain signals received by a subbottom profiling SONAR. While such SONAR systems are widely used and data from which are employed to interpret subbottom features and structures, there is a lack of modeling capability to simulate the SONAR returns scattered by random subbottom layers and sediment heterogeneity. Such a modeling capability is desirable to help make SONAR data interpretation from qualitative toward more quantitative. A two-way forward- and backward scatterer model is developed that is based on the mode coupling approach in a two-dimensional setting followed by Fourier synthesis to obtain time-domain signals. Numerical examples will be given to highlight the model capability, including cases with rough subbottom interfaces and random distribution of sound speed and density variations.

9:15

5aAO6. Joint estimation of water-column and seabed properties by inversion of modal-dispersion data. 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 joint estimation of water-column sound-speed profiles (SSP) and seabed geoacoustic models through Bayesian inversion of modal-dispersion data (arrival time as a function of frequency from warping time-frequency analysis). The inversion is formulated in terms of separate trans-dimensional models for the water column and seabed to intrinsically parametrize each according to the information content of the data. An efficient reversible-jump Markov-chain Monte Carlo algorithm is applied to estimate marginal posterior probability profiles, quantifying the resolution of SSP and seabed structure. Measured and simulated dispersion data, as well as varying levels of prior information, are considered to examine the ability to estimate water-column and seabed profiles, and to investigate relationships between these.

9:30–9:45 Break

9:45

5aAO7. Fisher information and Cramér-Rao bound analysis with stable gradients for ocean acoustic models. Michael C. Mortenson (Phys. and Astronomy, Brigham Young Univ., N 283 ESC, Provo, UT 84602, michaelcmort@byu.edu), Tracianne B. Neilsen, Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

The Fisher Information Matrix (FIM) and the Cramér-Rao Bound (CRB) are powerful tools for quantifying uncertainty in multi-parameter models. However, it is difficult to estimate the derivatives needed to calculate the FIM and CRB in ocean acoustic models. In this work, a methodology for estimating derivatives using physics-guided parameter preconditioning and Richardson extrapolation is presented. The method is validated on a case study of transmission loss from a range-independent normal mode model in ocean environments with a single sediment layer over a basalt basement. Five examples of sediment types ranging from mud to gravel are considered across frequencies from 50–400 Hz. Results demonstrate the utility of FIM and CRB analysis in quantifying both model sensitivities and parameter uncertainties, and in revealing parameter coupling in the model. This methodology is a general tool that can inform model selection and experimental design for inverse problem applications. [Work supported by ONR Contract No. N00014-19-C-2001.]

10:00

5aAO8. Estimating seafloor compressional sound speed using a long towed horizontal line array. Ernst Uzhansky (Marine Geosciences, Univ. of Haifa, Izhaq Greenboim Str., Apt. 12 (Koren Family), Haifa 3498793, Israel, ernstuzhansky@gmail.com), Yizhaq Makovsky, Boris Katsnelson (Marine Geosciences, Univ. of Haifa, Haifa, Israel), Shelly Copel, Tom Kazaz (The School of Phys. and Astronomy, Tel-Aviv Univ., Tel-Aviv, Israel), and Omri Gadol (Marine Geosciences, Univ. of Haifa, Haifa, Israel)

The methodology and results of estimations of seafloor compressional sound speed along with sound speed gradient in a compacting passive margin basin (offshore Israel) setting using standard multichannel seismic data (35 Hz peak frequency, 7200 m long array with 576 hydrophones) is presented. Geo-acoustic inversion technique with three matching parameters: (1) compressional sound speed at the water-sediment interface c_0; (2) constant positive velocity gradient K; and (3) thickness of the sub-bottom layer with a constant velocity gradient H_2 was used for estimations. The model considered consists of a water layer with a velocity that varies according to the season of data acquisition, lying above a layer in which velocity increases linearly with depth. Estimated c_0 and K assessed for bottom depths of 1000–1350 m, varied from 1500 to 1600 m s^{-1} and from 0.7 to 0.9 m s^{-1} per m, respectively, and tend to increase with bottom depth. It was also verified that there is no inter-dependence in the estimation of the different parameters. The proposed method allowed to make assessments of relatively low sound speed speeds at water-sediment interface, where conventional refraction-based methods are not applicable due to very high critical distance of the head waves.

10:15

5aAO9. Estimation of parameters of moving nonlinear internal waves using spectrogram of sound intensity fluctuations due to mode coupling. Yanyu Jiang (Marine Geosciences, Univ. of Haifa, Haifa, Israel), Valery Grigorev (Phys., Voronezh State Univ., Voronezh, Russian Federation), and Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il)

In this paper, results of experiment ASIAEX in the South China sea are considered. Episodes are analyzed when soliton-like nonlinear internal wave moves during ~6 h approximately along acoustic track ~30 km; from the source to receiver, where depths of the sea is ~350 and ~120 m correspondingly. The source, placed near the bottom, transmitted pulses (M-sequences) with the frequency 224 Hz. Spectrogram of intensity fluctuations calculated on the base of theory of authors [Grigorev *et al.*, *J. Ascout. Soc. Am.***140**(5), 3980–3994 (2016)] shows that peak frequencies in spectrogram correspond to the most strongly coupling pairs of modes: in given case 2-3, 3-4. Due to narrowing channel velocity V of moving soliton should decrease; note that the corresponding change of the predominating frequency as a function of position of soliton (or time) was observed. It is shown that temporal dependence of soliton's speed can be the fitting parameter to estimate this dependence using comparison of experimental data and results of modeling. The corresponding procedure gives values of V changing from 1.8 to 0.8 m/s, what is in a good agreement with satellite data, obtained for the same event. [Work was supported by RFBR, Grant 20-55-S52005.]

10:30

5aAO10. Comparison of measured and modeled speed of sound in the challenger deep. Scott Loranger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 86 Water St., Woods Hole, MA 02543, sloranger@whoi.edu) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Echo sounding and acoustic propagation in the hadal zone (depths below 6 km) rely on models of sound speed that suffer from uncertainty due to a paucity of measurements at those depths. In order to assess models of sound speed in deep water, and to provide additional measurements, the Deep Acoustic Lander, a passive acoustic profiler, was deployed into the Mariana Trench from the RV Pressure Drop in April 2021. The Deep Acoustic Lander houses an array of hydrophones along with an AML Scientific CTD and AML MinosX sound speed meter. Pressure, temperature and salinity data measured by the CTD were used to inform some of the most commonly used sound speed models. The modeled sound speed was then compared with direct measurements of the speed of sound in the hadal zone to depths beyond 10 000 m. The goodness of fit between each model and the direct measurements was then compared to assess the performance of each model in deep water.

10:45

5aAO11. Predicting the acoustic energy radiated by melting glacier ice. Hayden A. Johnson (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8622 Kennel Way, La Jolla, CA 92037, h3johnso@ucsd.edu), Grant B. Deane, M. Dale Stokes (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Oskar Glowacki, Mateusz Moskalik (Inst. of Geophys., Polish Acad. of Sci., Warsaw, Poland), Mandar Chitre, and Hari Vishnu (Acoust. Res. Lab., National Univ. of Singapore, Singapore, Singapore)

Ice sheets are expected to be major contributors to sea level rise in the coming decades and centuries. Ocean-forced melting at marine terminating glaciers has the potential to be an important driver of mass loss from these ice sheets, but is difficult to measure directly using conventional techniques, and thus remains poorly understood. The air pressure in bubbles trapped in glacier ice is typically greater than the hydrostatic pressure near the surface of the water. This pressure difference causes the bubbles to expand rapidly and radiate sound when they are released as the ice melts. This sound presents a promising signal which could be used to make passive acoustic measurements of submarine melting at glacier termini, if it can be characterized sufficiently. Prior work has demonstrated that there is considerable variability in the acoustic energy radiated by the release of individual bubbles. Here we show that the total acoustic energy radiated by small blocks of glacier ice as they melt is correlated with the total potential energy stored by the compressed air bubbles in the ice. We find that the conversion efficiency is highest for the most energetic blocks of ice, and present some modelling efforts to explain this trend.

FRIDAY MORNING, 3 DECEMBER 2021 — 401 (L)/404 (O), 8:30 A.M. TO 12:00 NOON

Session 5aBAa

Biomedical Acoustics: Acoustics for Kidney Stone Management I

Michael R. Bailey, Cochair
University of Washington, 1013 NE Boat St., Seattle, WA 98105

Adam Maxwell, Cochair
Department of Urology, University of Washington, Seattle, WA 98195

Invited Paper

8:30

5aBAa1. Diminishing comminution of urinary stones with subsequent laser pulses. Yuri A. Pishchalnikov (Applaud Medical, Inc., 953 Indiana St., San Francisco, CA 94107, yuri.pishchalnikov@applaudmedical.com), William M. Behnke-Parks (Applaud Medical, Inc., San Francisco, CA), and Marshall L. Stoller (Dept. of Urology, Univ. of California San Francisco, San Francisco, CA)

Ureteroscopic lithotripsy with Holmium:YAG lasers is a common treatment for urinary stones but the physical mechanisms of stone comminution are still under investigation. The dominant mechanisms were reported to be associated with the direct action of light on the stone surface causing the photothermal decomposition of stones and explosive vaporization of interstitial water. The objective of this study was to better understand stone fragmentation by the direct action of light and other potential mechanisms (cavitation) using ultra-high-speed video microscopy. Stone fragmentation was quantitatively assessed using an Open-source library of Computer Vision algorithms (OpenCV) with programs written in Python. Stone fragments were identified by detecting difference between the images and tracked with identification numbers considering cross-section areas and velocities of the fragments. Images of stone comminution were collected using a Shimadzu HPV-X2 camera at 200 000–500 000 fps with dry stones in air and those hydrated in water. Stone fragmentation by the direct action of light was observed to diminish with subsequent laser pulses. The diminishing fragmentation, however, was offset by the action of cavitation bubbles that continued to erode and fragment stone surface, augmenting stone breakage with subsequent laser pulses. [Work supported by the NIDDK of the NIH under Award R43DK129104.]

Contributed Papers

8:45

5aBAa2. Modelling laser-fluid coupling and cavitation in laser lithotripsy. Xuning Zhao (Virginia Tech, 460 Old Turner St., Randolph Hall Rm. 332, Blacksburg, VA 24060, xuningzhao@vt.edu), Wentao Ma, and Kevin Wang (Virginia Tech, Blacksburg, VA)

The absorption of laser by fluid and the resulting vapor bubbles plays a significant role in laser lithotripsy and many other industrial applications. In this talk, we present a coupled photo-thermal-mechanical model of laser interaction in compressible multiphase fluid flows. The model couples the three-dimensional compressible Navier–Stokes equation with a nonlocal laser absorption equation. The computational framework is an extension of the recently developed FIVER ("a Finite Volume method with Exact multi-material Riemann solver") framework, which features the use of level set and embedded boundary methods for tracking material (fluid–fluid and fluid–solid) interfaces, and the solution of local, one-dimensional multi-material Riemann problems for enforcing interface conditions. We will start with a brief review of the key model equations and numerical algorithms adopted by FIVER, as well as recent efforts on verification and validation. Next, physical models of laser absorption and laser-induced heating will be presented. Numerical models and methods for simulating phase change and tracking the vapor–liquid interface will be introduced and assessed. The performance of this computational framework will be demonstrated using several simplified model problems and a numerical experiment of cavitation in Holmium:YAG laser lithotripsy.

9:00

5aBAa3. Cavitation is the primary mechanism for stone dusting in holmium:YAG laser lithotripsy. Junqin Chen (MEMS, Duke Univ., 100 Sci. Dr., Durham, NC 27708, jc698@duke.edu), Derek Ho, Gaoming Xiang, Georgy Sankin (MEMS, Duke Univ., Durham, NC), Glenn Preminger (Div. of Urology, Duke Univ., Durham, NC), Michael Lipkin (Div. of Urology, Duke Univ., Durham, NC), and Pei Zhong (MEMS, Duke Univ., Durham, NC)

Ureteroscopy with holmium(Ho):YAG laser lithotripsy (LL) is the primary treatment modality for kidney stone patients. Despite this, the mechanism of stone dusting in LL is unclear. In this work, we treated 6×6 mm cylindrical BegoStone phantoms using 0.2–0.4 J and 20 Hz at various fiber-to-stone standoff distances (SD) either in water or air with perpendicular fiber orientation; the latter eliminates the contribution of cavitation. The laser fiber was advanced at different offset distances (OSD) from a flexible ureteroscope tip to alter LL-induced bubble dynamics. Furthermore, stones were treated in parallel fiber orientation to minimize thermal ablation while isolating cavitation damage. Our results demonstrate that stone damage in water is 2.5- to 90-fold of those produced in air. At OSD = 0.25 mm when the bubble collapsed away from the stone surface due to the presence of the scope tip, stone damage was greatly reduced to the level in air. Moreover, the average crater volume produced by parallel fiber orientation after 100 pulses was comparable with those produced by perpendicular fiber within SD = 0.25–1 mm. Altogether, we conclude that cavitation is the primary mechanism over photothermal ablation for stone dusting in Ho:YAG LL.

9:15

5aBAa4. Three-dimensional cavitation mapping in shock wave and laser lithotripsy. Mucong Li, Georgy Sankin, Tri Vu (Duke Univ., Durham, NC), Yun Jing (Acoust., Penn State Univ., State College, PA), Pei Zhong (Duke Univ., Durham, NC), and 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 non-invasive treatment of kidney stones. Cavitation plays an important role in stone fragmentation, yet it may also contribute to renal injury during SWL and LL. It is therefore crucial to determine the spatio-temporal distributions of cavitation activities to maximize stone fragmentation while minimizing tissue injury. Traditional cavitation detection methods include high-speed optical imaging, active cavitation mapping (ACM), and passive cavitation mapping (PCM), among which PCM has most practical applications in biological tissues. To image the dynamics of cavitation bubble collapse, we have developed tri-modality cavitation imaging that includes 3D high-speed optical imaging, ACM, and PCM seamlessly integrated in a single imaging system. 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 tri-modality system, we imaged and analyzed laser-induced single cavitation bubbles in both free field and constricted space and shock wave-induced cavitation clusters. Collectively, our results have demonstrated the high reliability and spatial-temporal accuracy of the 3D PCM approach, which paves the way for future *in vivo* applications in large animals and human studies during SWL and LL.

9:30

5aBAa5. Dual mode piezoelectric lithotripter for efficient breakup of urinary stones. Timothy L. Hall (Univ. of Michigan, 2200 Bonisteel Blvd., 1107 Gerstacker Bldg, Ann Arbor, MI 48109, hallt@umich.edu) and William W. Roberts (Univ. of Michigan, Ann Arbor, MI)

The coalescence and dispersion of cavitation bubbles through Bjerknes forces created by supplemental low amplitude acoustic fields has been shown to greatly improve the efficiency of shockwave lithotripsy for breaking urinary stones. Unfortunately, the large size of extracorporeal lithotripters typically leaves little acoustic window for additional transducers to create these fields. In this study, we explored the feasibility of using a piezoelectric lithotripter transducer in a dual mode generating both shockwaves for stone comminution and low amplitude tone bursts for bubble coalescence and dispersion. A piezoelectric lithotripter head and driver unit (piezolith 3000) was provided by the manufacturer (Richard Wolf GmbH). A custom class-D driver circuit was constructed from very high voltage blocking IGBTs and wired in parallel with the manufacturer's excitation circuit for shockwave generation. Acoustic output was measured with a fiber optic probe hydrophone at the focus in a tank of degassed water. The driver produced acceptable 200 kHz tone bursts of 5–100 cycles up to at least 1 MPa amplitude. Lithotripter power settings 1–3 could be safely used with the combined driver and were sufficient to break model begostones *in vitro*. Future studies will compare efficiency of stone comminution using the dual mode system.

9:45

5aBAa6. Surface acoustic waves and Mach stem formation produced by lithotripter shock wave-stone interaction. Gaoming Xiang (Duke Univ., 229 Hudson Hall, Durham, NC 27708, gaoming.xiang@duke.edu), Georgy Sankin (Duke Univ., Durham, NC), Shunxiang Cao (Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Kevin Wang (Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), and Pei Zhong (Duke Univ., Durham, NC)

Lithotripter shock wave (LSW)-stone interaction may vary significantly due to respiratory motion of the patient. In this work, we have investigated the variations of the LSW-stone interaction and associated stone fracture pattern using photoelastic imaging, phantom experiments, and three-dimensional fluid-solid interaction modeling at different lateral locations in a lithotripter field. In contrast to the T-shaped fracture pattern observed in a disk-shaped stone under symmetric loading, we have observed a gradual transition of the fracture pattern to a tilted L-shape under asymmetric loading. Importantly, our model simulations reveal the generation of surface acoustic waves (SAWs)—a leaky Rayleigh wave on the anterior and a Scholte wave on the posterior surface of the stone. The propagation of SAWs on the stone boundary is accompanied by a progressive transition of the LSW reflection pattern from regular to von Neumann and to weak von Neumann reflection near the glancing incidence and, concomitantly, the development of a Mach stem, swirling around the stone boundary. The maximum tensile stress and stress integral are produced by SAWs on the stone boundary under asymmetric loading, which drive the initiation and extension of surface cracks into the bulk of the stone that has been confirmed by microCT analysis.

10:00–10:30 Break

10:30

5aBAa7. Particle swarm optimization of the manipulation of acoustic waves through nonhomogeneous, anisotropic mediums for application in shock wave lithotripsy. Joseph J. Roemhildt (Dept. of Eng. Sci., Bethany Lutheran College, 700 Luther Dr., Mankato, MN 56001, jroemhildt@blc.edu), Jiadi Fan (Dept. of Aerosp. Eng. and Mech., Univ. of Minnesota, Minneapolis, MN), Ziping Wang (Faculty of Civil Eng. and Mech., Jiangsu Univ., Zhenjiang, Jiangsu, China), and Sheng Sang (Dept. of Eng. Sci., Bethany Lutheran College, Mankato, MN)

Shock Wave Lithotripsy is the most widely used treatment for kidney stones, a fragmentation process using shock waves generated outside the body that are focused onto the stone. However, the nonhomogenous, anisotropic nature of the human body causes the waves to be refracted, which can lead to less efficient fracturing of the kidney stone, requiring longer or more frequent treatment sessions, and increased risks of side effects from treatment. In this paper, a Particle Swarm Optimization (PSO) algorithm was utilized to program wave generators using a swarm of individual particles representing element configurations, such as amplitude, phase difference, and stimulation parameters. A Finite Element Model of the area between the shockwave generator and the kidneys used to demonstrate how the PSO algorithm can efficiently optimize a multi-objective function that maximizes predictions of activation in the kidney stone (Region of Interest, ROI) and minimizes predictions of activation in the surrounding sensitive tissues (Regions of Avoidance, ROA).

10:45

5aBAa8. Ultrasound imaging metrics to characterize cavitation activity during burst wave lithotripsy. Wayne Kreider (CIMU/APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Christopher Hunter, Ga Won Kim, Bryan W. Cunitz, Yak-Nam Wang, Jeff Thiel, Michael R. Bailey (CIMU/APL, Univ. of Washington, Seattle, WA), and Adam Maxwell (Urology, Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a noninvasive approach for comminuting kidney stones with short bursts of focused ultrasound. An important practical aspect of this approach is the ability to deliver treatments with a handheld therapy probe integrated with an ultrasound imaging probe to facilitate ultrasound guidance and monitoring. Because cavitation can influence both injury and efficacy for stone comminution, we seek to develop cavitation-related imaging metrics that correlate with stone fragmentation and tissue injury. Toward this end, a Verasonics system was programmed to synchronize therapy and imaging pulses and record B-mode and Doppler data using a P4-2 probe. Imaging data were recorded during BWL exposures at 350 kHz both for *in vivo* injury experiments in a porcine model and *in vitro* breaking experiments for stones in a kidney phantom. For *in vivo* exposures, cavitation was detected more sensitively with Doppler than B-mode, and the detection of sustained cavitation in the kidney parenchyma appears to precede the onset of hemorrhagic injury. For *in vitro* exposures using different pulse lengths and pulse repetition frequencies to break stones, the axial position of a bubble cloud proximal to the stone appears to correlate with efficacy. [Work supported by NIH NIDDK, Grants P01 DK043881 and K01 DK104854.]

11:00

5aBAa9. Progression of hemorrhagic kidney injury from burst wave lithotripsy exposures. Yak-Nam Wang (CIMU/APL, School of Oceanogr., Univ. of Washington, 1122 NE Boat St., Seattle, WA 98105-6770, ynwang@uw.edu), Adam Maxwell, Tony T. Chen (Urology, Univ. of Washington, Seattle, WA), Bryan W. Cunitz, Jeff Thiel, Christopher Hunter, Daniel Leotta (CIMU/APL, Univ. of Washington, Seattle, WA), James C. Williams, Ziyue Liu (Indiana Univ. School of Medicine, Indianapolis, IN), Akshit Arora, Ravneet Vohra, Donghoon Lee (Radiology, Univ. of Washington, Seattle, WA), and Wayne Kreider (CIMU/APL, Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a new non-invasive treatment for comminuting kidney stones. Although exposure parameters used in current clinical trials have produced negligible renal injury in preclinical pig studies, it remains of interest to understand how quickly injury accumulates, should it occur. BWL treatments lasting $\leq$30 min were delivered to 36 sites in 20 kidneys of 11 pigs. Treatments at 350 kHz utilized pulses with supra-clinical 8–9 MPa peak negative pressure, 20 cycles/pulse, and a pulse repetition frequency (PRF) of 10 or 40 Hz. Real-time monitoring was performed with B-mode and Doppler ultrasound to identify the presence and duration of cavitation activity, which has been correlated with the onset of injury. After treatment, kidneys were excised for magnetic resonance imaging (MRI) within 3 h of euthanasia. MRI was used to quantify the volume of any hemorrhagic injury. For sites with injury at 40 Hz PRF, the average ($n = 5$) injury volume increased with the number of BWL pulses (72, 720, 7200, or 72 000 pulses) delivered with cavitation. Notably, the rate of lesion growth slowed above 7200 pulses while all lesion volumes were smaller for equivalent cavitation exposures at 10 Hz PRF. [Funding support by NIH P01-DK043881, K01-DK104854.]

11:15

5aBAa10. Single-framework simulations of acoustic-wave–bubble cloud–stone interactions. Jean-Sebastien A. Spratt (Mech. and Civil Eng., California Inst. of Technol., 1200 E California Blvd, MC 104-44, Pasadena, CA 91125, jspratt@caltech.edu), Mauro Rodriguez (Brown, Providence, RI), Spencer H. Bryngelson (Georgia Inst. of Technol., Pasadena, CA), Shunxiang Cao, and Tim Colonius (Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA)

Understanding the interaction of acoustic waves and stones is of principal importance to the development and optimization of shock wave and ultrasound-based therapies, such as shock wave and burst wave lithotripsy (SWL and BWL). Simulating these interactions is useful in furthering our understanding, but the development of simulation tools is complex. These tools often use separate numerical frameworks for compressible fluids and solids. Introducing cavitating bubble clouds which can occur during these therapies further complicates the problem, introducing a broad range of relevant spatio-temporal scales. To tackle these challenges, we have developed a single-framework code capable of simulating all relevant physics. Cavitating bubble clouds are introduced via a sub-grid model, and a hypoelastic model simulates solid materials, all in the same Eulerian framework. These are implemented in our open-source Multi-component Flow Code (MFC) [Bryngelson *et al*., *Comp. Phys. Commun*. (2020)]. We demonstrate the solver's capabilities for 3D SWL and BWL simulations, where we can obtain the stress state in the stone for various configurations. GPU-accelerated simulations also elucidate the computational advantages of the single-framework approach. [This research was supported by the National Institutes of Health (NIH Grant No. 2P01-DK043881) and the Office of Naval Research (ONR Grant No. N0014-18-1-2625).]

11:30

5aBAa11. Input-output analysis of fluid–structure interaction to optimize stone breakage in burst-wave lithotripsy. Shunxiang Cao (Gate-Thomas Lab., Mech. and Civil Eng., California Inst. of Technol., RM-112, 1200 E. California Blvd, MC 104-44, Pasadena, CA 91125, csxtocit@caltech.edu), Tim Colonius (Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Adam Maxwell, Ga Won Kim (Urology, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Univ. of Washington, Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Burst-wave lithotripsy is a non-invasive treatment of kidney stones using short pulses of focused ultrasound. In this talk, we present the use of input-output analysis to optimize the design and control of a multi-element array transducer to achieve optimal stone breakage. The analysis is based on a linear fluid-structure coupled system that maps the acoustic forcing from individual elements to the stress in a stone of assumed shape, size, and composition. We determine the optimal frequency and optimal distribution of the phase and amplitude across the aperture by maximizing a cost function that represents the strain energy in stone. The optimal parameters are obtained by applying a state-of-the-art, randomized singular value decomposition (SVD) to the discrete linear operator. The results show that carefully controlling the relative phase and amplitude between elements can increase the strain energy (by 2-3 times in certain cases) compared to a uniform distribution under the same input energy. This suggests that the stone

fragmentation can be accelerated or performed with lower energy for safer treatment. The improvement is interpreted and validated by combining high-fidelity simulations with high-speed camera images of crack formation in model stones from *in vitro*- experiments. [Work supported by NIH, Grant No. P01-DK043881.]

11:45

5aBAa12. Effect of pulse duration and repetition rate on burst wave lithotripsy stone fragmentation *in vitro*. Ga Won Kim (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of WashingtonHenderson Hall (HND), 1013 NE 40th St., Seattle, WA98105, gawonkim@uw.edu), Christopher Hunter (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Adam Maxwell (Urology, Univ. of Washington, Seattle, WA), Bryan W. Cunitz, Michael R. Bailey, and Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a noninvasive technology to fragment urinary stones using short pulses of focused ultrasound. The aim of this study was to assess how different combinations of pulse length and pulse repetition frequency (PRF) impact stone fragmentation rate and cavitation activity in an *in vitro*-model. Human calcium oxalate stones between 4–7 mm were exposed in a tissue phantom with a 350-kHz BWL transducer with a beamwidth of 6 mm. Treatments were delivered in a 45–50% degassed water bath. Ultrasound imaging was used to target the stone and monitor cavitation activity around the stone. Stones were exposed to ten different parameter sets at peak negative pressure = 6.2 MPa in 5-min intervals up to 30 min total. After each interval, the remaining mass of fragments >2 mm was measured to determine the rate of comminution. Longer pulse duration generally produced improved fragmentation. Increasing the pulse repetition rate with set pulse duration did not always produce faster comminution. The results suggest that more effective fragmentation may be achieved in BWL by long pulse durations and low PRF. [Work funded by NIDDK, Grants P01 DK043881 and K01 DK104854. Maxwell, Cunitz, and Bailey have consulting agreements with and equity in SonoMotion, Inc.]

FRIDAY MORNING, 3 DECEMBER 2021 | 405 (L)/406 (O), 8:00 A.M. TO 11:50 A.M.

Session 5aBAb

Biomedical Acoustics and Physical Acoustics: An Educational Session to Explore the History of Cavitation Bioeffects and How Lessons Learned are Shaping the Future of Cavitation in Medical Ultrasound

Hong Chen, Cochair
Washington University in St. Louis, 4511 Forest Park Ave., St. Louis, MO 63108

Julianna Simon, Cochair
Penn State University, 201E Applied Science Bldg, University Park, PA 16802

Chair's Introduction—8:00

Invited Papers

8:05

5aBAb1. The importance of acoustic cavitation in the evolving technologies of therapeutic ultrasound. Lawrence A. Crum (Appl. Phys. Lab, Univ. of Washington, 4662 175th Ave. SE, Bellievue, WA 98006, lacuw@uw.edu)

Cavitation has played an active and important role in the development of therapeutic ultrasound. From its vital role in the effectiveness of lithotripsy to its essential role in the application of histotripsy, the awareness of cavitation must be considered in practically every clinical implementation of High Intensity Focused Ultrasound. This lecture will present several examples of how cavitation has been an essential ingredient of many current and promising therapeutic ultrasound methodologies and technologies.

8:35

5aBAb2. Know when something happens: Some thoughts on Apfel's three laws of cavitation research. Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, ronald.roy@hmc.ox.ac.uk)

When it comes to experimental research on acoustic cavitation, connecting the dots can be challenging. It requires that the researcher possesses good command of the essential facts associated with the medium, the sound field, and the relevant observable phenomena. This maxim is embodied in Robert Apfel's "Three Laws" of cavitation research: (1) Know Thy Fluid, (2) Know Thy Sound Field, and (3) Know When Something Happens. To honor the late Professor Apfel, I will discuss when and how these laws are applicable in a variety of scenarios across application areas (biomedical, physical, industrial) and share some reminiscences on what it was like for an aspiring young scholar to rub shoulders with the greats during a golden age of acoustic cavitation, physical acoustics, and biomedical ultrasound.

9:05

5aBAb3. Sonoluminescence as a probe of conditions during cavitation. Kenneth S. Suslick (Dept. of Chemistry, Univ. of Illinois at Urbana-Champaign, 600 S. Mathews Ave., Urbana, IL 61822, ksuslick@uiuc.edu)

Acoustic cavitation occurs in all liquids irradiated with sufficient intensity of sound or ultrasound. The collapse of such bubbles creates local heating and provides a unique source of energy for driving chemical reactions. In addition to sonochemical bond scission and formation, cavitation also induces light emission in many liquids. This phenomenon of sonoluminescence (SL) and has captured the imagination of many researchers since it was first observed 85 years ago. SL provides a direct probe of cavitation events and has provided most of our understanding of the conditions created inside collapsing bubbles. Spectroscopic analyses of SL from single acoustically levitated bubbles as well as from clouds of bubbles have revealed molecular, atomic, and ionic line and band emission riding atop an underlying continuum arising from radiative plasma processes. These studies permit quantitative measurement of the intracavity conditions: relative peak intensities for temperature measurements, peak shifts and broadening for pressures, and peak asymmetries for plasma electron densities. Extraordinary conditions are generated inside the collapsing bubbles in ordinary room-temperature liquids – observable temperatures exceeding 15 000 K, pressures $>$1000 bar, and heating and cooling rates in excess of 10^{12} K/s. For clouds of cavitating bubbles, multibubble sonoluminescence (MBSL) reveal slightly more moderate local conditions inside bubble clouds of $\sim$5000 K, $\sim$300 atm. Biomedical applications of SL will be discussed.

9:35

5aBAb4. Can we harness cavitation for therapeutic benefit? 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)

Therapeutic ultrasound and cavitation nucleation agents are currently being developed for a range of clinical applications such as enhanced cardiovascular drug delivery, blood brain barrier disruption, immunotherapy, sonothrombolysis, sonobactericide, and bioactive gas delivery. Microbubbles oscillate when exposed to ultrasound and create stresses directly on nearby tissue that elicit a biological response, thereby enhancing drug transport into vascular tissue, biofilm, thrombi, or tumors. Catheter-directed therapeutics in combination with an ultrasound contrast agent nucleate sustained bubble activity and enhanced drug penetration and efficacy. Passive cavitation imaging aids in mapping the spatial and temporal extent as well as type of bubble activity. The conditions for the onset of inertial and stable cavitation will be reviewed in the context of the Mechanical and Cavitation Indices. Successful implementation of ultrasound and cavitation nuclei-mediated drug and bioactive gas delivery has the potential to change the way drugs are administered systemically, resulting in more effective therapeutics and less-invasive treatments.

10:05–10:20 Break

10:20

5aBAb5. Cavitation from fundamental research to clinical applications. Kullervo Hynynen (Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca)

The therapeutic use of focused ultrasound has a long history exploring both the use of inertial cavitation and thermal effects to induce tissue effects noninvasively. The thermal effects have been better understood and easier to control in clinical practice and thus they were first explored for example for cancer treatments. The utilizations of cavitation has been slower although it offers a wide spectrum of useful bio-effects. One of the first observations was that ultrasound generated gas bubbles will enhance energy absorption and can thus reduce the required energy for ablative exposures. Similar enhancement can be achieved with preformed microbubbles injected in the blood stream. These intra-vascular bubbles when excited with low pressure amplitude waves can also modulate the permeability of the blood vessels. This is most useful in organs such as brain where the endothelial cells are tightly connected forming a barrier for molecular diffusion into the tissue. The local modulation of the Blood-Brain barrier has been explored for targeted drug delivery into the brain in animal models and is now in clinical testing. In this talk the history, current status and future potential of ultrasound excited gas bodies will be discussed.

10:50

5aBAb6. Cavitation and microbubble modeling. Charles C. Church (NCPA, Univ. of MS, NCPA, 145 Hill Dr., University, MS 38677, cchurch@olemiss.edu)

Bubbles in tissue may either help or hinder biomedical ultrasound. Exogenously introduced ultrasound contrast agents may comprise suspensions of micrometer-sized gas bubbles of sulfur hexafluoride, perfluorocarbon, or other low-solubility gas that are stabilized against dissolution by shells of lipids, proteins or synthetic, or alternatively, quasi-stable micro-droplets of certain liquids that nucleate

gas bubbles in response to an impinging acoustic beam. However, tiny vapor bubbles may form spontaneously in tissues and act as potentially damaging cavitation nuclei under the influence of an acoustic wave. Microbubbles respond to the acoustic field by contracting and expanding to a much greater extent than does the surrounding tissue. This 'acoustic cavitation' may be violent, producing highly reactive chemical species, emission of ultraviolet and soft x-ray photons, and radiated shock waves, or less powerful, inducing, e.g., continuous heat production, radiation forces on neighboring particles, and microstreaming. Much early work assumed cavitation bubbles were surrounded by simple liquids, but biological tissues are viscoelastic, which can significantly affect bubble responses and thus must be considered. Mathematical models for these and related processes will be presented and compared, and details of their various physical mechanisms and the effects they may produce in biological tissue will be discussed.

11:20

5aBAb7. A review of modeling and simulation of cavitation with application to ultrasound and shock-wave therapies. Tim Colonius (Mech. and Civil Eng., California Inst. of Technol., 1200 E California Blvd, Pasadena, CA 91125, colonius@caltech.edu)

High-intensity ultrasound and shock waves are used for intra- and extra-corporeal manipulation of cells, tissue, and urinary calculi. In many applications, these waves induce the expansion and collapse of preexisting or newly cavitating bubbles whose presence can either mediate the generation of localized stresses or lead to collateral damage, depending on how effectively they can be controlled. We review efforts aimed at simulating wave propagation, nucleation, and bubble dynamics. We discuss both interface capturing approaches that can be used to look at micro-scale bubble collapses as well as stochastic models that treat the bubbles as an unresolved disperse phase that can be used to simulate how bubble clouds alter macroscopic wave focusing, scattering, and stress fields. The translation of the fundamental fluid dynamics into improvements in the design and clinical application of shockwave lithotripters will also be discussed.

FRIDAY MORNING, 3 DECEMBER 2021 — 402 (L)/403 (O), 8:40 A.M. TO 12:00 NOON

Session 5aPA

Physical Acoustics: Physical Acoustics Potpourri

Christopher M. Kube, Chair
Engineering Science and Mechanics, The Pennsylvania State University, 212 Earth and Engineering Sciences Bldg, University Park, PA 16802

Chair's Introduction—8:40

Contributed Papers

8:45

5aPA1. Comparison of dry or wetted sand column vibrating over a clamped elastic plate: A nonlinear sand-plate-oscillator. Emily V. Santos (Psych. and Visual Arts, Univ. of Maryland Baltimore County, 201 Church St., Baltimore, MD 21225, santosemily08@gmail.com) and Murray S. Korman (Phys., U.S. Naval Acad., Annapolis, MD)

The sand plate oscillator apparatus uses two aluminum circular flanges sandwiching and clamping a thin circular elastic acrylic plate (density 1.21 g/cc). Dry sifted or wetted masonry sand is supported at the bottom by the clamped acrylic plate (4.5 in.) and stiff cylindrical sidewalls. Sifted masonry sand has a density of 1.55 g/cc. An accelerometer centered on the plate measures vibration response using a swept tone spectrum analyzer between 50 and 850 Hz. The plate was driven from below by a small loudspeaker. Three experiments were performed. (1) Nonlinear tuning curve response required driving the loudspeaker and incrementing the signal amplitude between sweeps. A fixed column of 354 g of dry masonry sand was used, with comparisons when 0, 11, 22, and 33 g of water were carefully sprayed on. (2) At constant drive amplitude, tuning curve response with 354 g of dry sifted sand was compared with results when 2 g of water were sprayed every 104 s. (3) Fixed drive amplitude tuning curves for incremental dry sand loading with 47 g for each layer were compared to incremental layers of 47 g dry sand sprayed with 12 g water. The wetted sand had increased stiffness.

9:00

5aPA2. Nonlinear slow dynamic elasticity in metal systems. 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 Richard Weaver (Phys., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Nonlinear "slow dynamic elasticity" is found to be universal amongst geological materials and other materials with complex microstructures, such as cement and cracked glass blocks. The slow dynamic behavior is characterized by a log(time) recovery, after an initial drop of stiffness, induced by strain events as small as a microstrain. Slow dynamic behavior is also reported in unconsolidated aggregates of glass beads under pressure and in single glass beads confined between glass plates, systems that are probed by diffuse ultrasound and monitored for stiffness changes using coda-wave

interferometry. At present, there is no consensus as to the microphysical basis for the behavior, though hypotheses include glassy dynamics at grain-grain interfaces, microcracking, thermal activation of a distribution of atomic-scale barriers, and moisture. Here we extend the materials known to exhibit slow dynamics to include metal structures: an aluminum bead pack under pressure, and individual aluminum and steel beads confined between aluminum and steel plates, respectively. We find that the strength of the slow dynamics is largely the same as it is for glass beads, suggesting that neither cracking nor glassy micro-structures are essential.

9:15

5aPA3. Acoustoelastic wave propagation in initially stressed solids using a pure stress formulation. Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu) and Andrew N. Norris (Rutgers Univ., Piscataway, NJ)

Wave propagation in homogeneous elastic solids is governed by Cauchy's first law of motion, which relates the acceleration of material points to the divergence of stress. For Hookean solids, the stress is typically written in terms of displacements to form an equation of motion with displacement as the dependent variable, i.e., Navier's equation of elastodynamics. Previously, the authors considered wave propagation in anisotropic elastic solids by formulating the equation of motion using only stress as the dependent variable. Eigenvalue solutions provided the (stress) wave phase velocity in terms of elastic compliance constants. In this presentation, the stress formalism is extended and applied to the nonlinear problem of stress wave propagation in an initially stressed material (also known as acoustoelasticity). The stress formulation is viewed as a more natural fit over historical derivations, which were based on wave displacements superimposed on an initially strained material state. The stress formulation of elastodynamics is firstly reviewed. Then, the extension to stress wave propagation in an initially stressed material is given. The phase velocity of the stress wave is related to the initial stress through second- and third-order compliance constants (instead of elastic stiffnesses in traditional acoustoelasticity).

9:30

5aPA4. Influence of residual stress on the free vibrations of an elastic solid; application to resonant ultrasound spectroscopy (RUS). Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu) and Jared Gillespie (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Resonant ultrasound spectroscopy (RUS) is a standard technique for experimentally measuring the elastic constants of anisotropic elastic materials. Traditionally, the influence of internal or residual stresses on the measurement of elastic constants is not taken into consideration. However, it is well-known that residual stress can impart a small variation in the phase velocity of ultrasonic bulk waves, which is known as the acoustoelastic effect. Thus, it is plausible that residual stress will influence the resonances of samples measured during RUS. In this presentation, the elastodynamic theory for describing free vibrations is reformulated by using a stress-dependent constitutive relationship. The model culminates in an eigenvalue problem with a stiffness matrix being dependent on the spatially varying residual stress field within the domain of the sample. The model is evaluated by considering a realistic residual stress distribution. Results show the degree at which frequencies of resonance depend on the residual stress. While accounting for residual stress is expected to improve the estimates of elastic constants, there is also potential to use RUS as a tool to assess and characterize the residual stress present in samples.

9:45

5aPA5. Improved understanding of natural weathering on the seismic-to-acoustic coupling ratio over buried containers. James Sabatier (Univ. of MS, NCPA, University of MS, Oxford, MS 38677, sabatier@olemiss.edu) and Brandon Hubbard (SOAIR, Oxford, MS)

The long term effects of natural weathering on the acoustic-to-seismic coupling ratio over buried containers is being investigated at a test site near the campus of the University of Mississippi. The weathering experiment has been carried out over five years. Two containers were buried at a 30 cm depth in 2014 and the coupling ratio monitored during 2014 and 2017. During this time period the soil matrix was not disturbed. Four different containers, buried at 15 cm in 2021, are exposed to the natural weather and the coupling ratio is being monitored. The soil water potential or pressure is measured continuously over the measurement periods. This talk will present the long-term effects of natural weathering on the ratio of normal velocity component of a surface geophone normalized by a surface microphone in response to normally incident airborne sound. As the soil matrix dries it becomes more consolidated increasing the soil acoustic stiffness, the S/A ratio becomes less sensitive to the hydraulic capillary pore pressure. During the rainy season, the soil consolidation relaxes resulting in an increased coupling ratio. The phenomenon is cyclical with the local annual soil wetting and drying cycles.

10:00

5aPA6. Acoustic, thermal, and viscous modes in MEMS perforated back plates. Richard Raspet (NCPA, Univ. of MS, 145 Hill Dr., University, MS 38677, raspet@olemiss.edu), Craig J. Hickey (NCPA, Univ. of MS, University, MS), and Vahid Naderyan (Knowles Electronics, Itasca, IL)

The back plates of MEMS microphones are commonly perforated with holes with radii on the order of a few micrometers. The plates themselves are only a few micrometers thick. Acoustic propagation in small pores is usually described in terms of the fundamental acoustic mode whose solution can be derived using the Low Reduced Frequency method of Tijdeman. Higher order acoustic modes and the viscous and thermal modes are assumed to be highly attenuated and negligible in pores of longer tubes. Since the holes in MEMS are relatively short, it is possible that additional modes may be necessary to accurately calculate the acoustic losses in the back plate. To investigate this possibility, Rayleigh's determinant is solved numerically and the behavior of the additional modes beyond the fundamental as the pore size is reduced determined. For larger tubes, all modes behave as expected, but as the tube radius is reduced, the higher order acoustic modes become non-physical, leaving only the viscous and thermal modes as possible contributors to the attenuation.

10:15–10:30 Break

10:30

5aPA7. Resonant ultrasound spectroscopy of free vibrating oscillators for massive particles detection. Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, 108 Lewis Hall Phys., University, MS 38677, iostrov@phy.olemiss.edu)

The half-inch diameter quartz oscillators are set for free vibrations. The crystal admittance (Y) versus frequency of ultrasound (F) is measured near fundamental harmonic of 9-10 MHz. The quartz crystallographic axes X, Y, interplain and rotated are oriented in space toward the sky locations of the Sun, center of the Milky Way galaxy or another star. It is demonstrated that Y(F) spectra are sensitive to an alignment of the crystallographic axes with space direction toward the stars. The two general results include: (1) changing Y(F) from a few to tens of percents under aligning crystallographic axes toward the stars and (2) occurring periodic minima in Y(F)-dependencies taken during some seconds with a tracking generator synchronized with a spectrum analyzer. The deepest minima in Y(F) appear when the azimuths of crystal axes coincide with those of the stars. These changes may be explained by electrically neutral massive particles (mp) emitted from a star and interacting with the crystals. The mass of a single mp-particle is estimated around $\sim 5 \times 10^{-21}$ kg. RUS measurements correlate with the data from a crystal pendulum having a quartz bob and starts swinging when mp-particle hits it. RUS of free vibrating oscillators is perspective for detecting stellar massive particles.

10:45

5aPA8. Temporal coherence of sound propagated along vertical paths through the atmosphere. Matthew J. Kamrath (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, matthew.j.kamrath@erdc.dren.mil), Vladimir Ostashev, D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Michael J. White (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), and Anthony Finn (Defence and Systems Inst., Univ. of South Australia, Mawson Lakes, South Australia, Australia)

Reductions in signal coherence caused by atmospheric turbulence constrain the performance of signal processing methods. This presentation compares theoretical predictions and measurements of acoustic temporal coherence, which describes the similarity of a signal at two times. Combining sound propagation theory with turbulence models yields the theoretical coherence. To be applicable to vertical and slanted propagation, the turbulence models use height-dependent variances and length scales for the fluctuations in temperature, shear-produced velocity, and buoyancy-produced velocity. Meteorological instruments on a 135-m tower measured the required model input data. The coherence measurements used a ground-based source emitting tones between 0.6 and 3.5 kHz and nine microphones on the same tower at heights 39, 80, and 130 m. In most cases, the predicted coherences accurately approximate the measured temporal coherences. In the two early-morning trials, the measured coherence times were much larger than predicted because the atmospheric turbulence was not fully developed. Excluding these cases, the measured coherence times were in the range 0.1–100 s. For each trial, the coherence times decreased slightly more than an order of magnitude from the smallest to largest frequencies. From the shortest to the longest propagation distances, the coherence times decrease slightly less than an order of magnitude.

11:00

5aPA9. Far-field acoustic subwavelength imaging with blind structured illumination. Jinuan Lin (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1415 Eng. Dr., Rm. 3533, Madison, WI 53706, jlin328@wisc.edu) and Chu Ma (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, Madison, WI)

The blind structured illumination microscopy (blind-SIM) is an optical method to surpass the diffraction limit, a fundamental limitation to all wave-based imaging techniques. By illuminating the object with multiple structured illumination patterns, the spatial frequency mixing between the object and the illuminations converts evanescent waves to propagating waves that can be detected by sensors in the far field. In this work, we develop blind-SIM in the acoustic domain. The blind acoustic structured illuminations (ASI) are generated by randomly located scatterers with subwavelength spatial distributions. The image of the object can be reconstructed with a subwavelength resolution from multiple far-field measurements (each produced by an ASI pattern) utilizing a compressed sensing algorithm. We demonstrate subwavelength acoustic imaging resolution in both kilohertz airborne and megahertz underwater acoustic imaging systems. The developed blind-SIM method has great potential in medical imaging, non-destructive testing, and underwater acoustic imaging for improving the imaging resolution of objects far away from the sensors.

11:15

5aPA10. Unbounded full-aperture acoustic wave experimentation using multidimensional deconvolution. Xun Li (Inst. of Geophys., ETH Zurich, No. H 41.2, Sonneggstrasse 5, Zurich CH-8045, Switzerland, lixunjack@outlook.com), Theodor Becker (Inst. of Geophys., ETH Zurich, Zurich, Switzerland), Matteo Ravasi (Earth Sci. and Eng., KAUST, Thuwal, Saudi Arabia), Johan Robertsson (I), and Dirk-Jan van Manen (Institutite of Geophys., ETH Zurich, Zurich, Switzerland)

Physical experiments are set up to study acoustic wave propagation in a laboratory. However, wave propagation experiments carried out in a size-limited volume often suffer from unwanted reflections at the boundaries of the experimental domain. We propose a method of multidimensional deconvolution (MDD) to process recorded data such that the undesired imprint of the laboratory boundary is completely removed from the data. The MDD results consist of Green's functions between any pair of points on a mathematically closed receiver surface, sampling the scattering wavefield related to an object of interest with a full aperture. We apply the MDD algorithm to process data recorded in a 2D acoustic waveguide in which a rigid steel block is placed inside of a circular recording array. The scattering imprint of the waveguide boundary is removed from the data, and the scattering Green's functions only related to this steel block are obtained. The MDD methodology used for the physical experiment is also validated using synthetic simulations; this further corroborates the effectiveness of MDD in obtaining the desired Green's functions associated with arbitrary scatterers in unbounded domains.

11:30

5aPA11. Physical parameters influence on the transmitted signal through a rigid porous medium in the low-frequency range of ultrasound. Abdelmadjid Mahiou (Acoust. and Civil Eng. Lab., Khemis-Miliana Univ., Khemis-Miliana 44000, Algeria, mahioumadjid2020@gmail.com), Ilhem Sellami, and Mustapha Sadouki (Dept. of Material Sci., Acoust. and Civil Eng. Lab., Khemis Miliana Univ., Khemis-Miliana, Algeria)

The acoustic characterization of porous materials with rigid structure plays an important role in the prediction of the acoustic behavior of these media. In order to obtain an accurate prediction of its response as a function of frequency, several parametric models have been proposed. However, semi-phenomenological models are the most used to describe the acoustic behavior of porous materials with rigid or flexible structures according to a certain number of macroscopic physical properties related to the internal structure of the material and essential for their valuation. Among these models, the one proposed by Johnson *et al*. to describes the complex density of an acoustic porous material with an immobile skeleton having arbitrary pore shapes. The objective of this work is to study the effect of these parameters involved in Johnson's model as well as other new parameters, recently introduced by Sadouki [*Phys. Fluids* **33**, (2021)), on the transmitted signal in the low-frequency regime of ultrasound.

11:45

5aPA12. Effect of physical and mechanical parameters on the reflected signal from the first interface of human cancellous bone—application of Biot's theory. Mustapha Sadouki (Acoust. and Civil Eng. Lab., Khemis-Miliana Univ., Rte. Thenia el Had, Khemis-Miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz) and Fatima Zahra Hamadouche (Acoust. and Civil Eng. Lab., Khemis-Miliana Univ., Khemis-Miliana, Algeria)

In this work, a simulated study on the effects of physical and mechanical parameters describing human cancellous bone on reflected ultrasound waves is proposed. The visco-inertial exchanges between the structure and the saturating fluid are described by the Biot theory modified by the Johnson model. A simplified expression of the reflection coefficient at the first interface is obtained in the frequency domain. This expression depends on three physical parameters which are porosity, tortuosity, and viscous characteristic length as well as the mechanical parameters; Young's modulus and Poisson's ratio. The reflected signal is calculated in the frequency domain by the product of the spectrum of the incident signal with the reflection coefficient. In the time domain, the reflected wave is obtained by taking the inverse Fourier transform of the reflected frequency signal. The effect of each parameter on the reflected signal is studied. The results obtained are discussed and compared with those given in the literature.

Session 5aPP

Psychological and Physiological Acoustics Development and Aging; Clinical Populations (Poster Session)

Ellen Peng, Chair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68131

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and authors of even-numbered papers will be at their posters from 10:30 to 12:00 noon.

Contributed Papers

5aPP1. Subcortical measures of individual tolerance to background noise and sensitivity to speech distortions induced by noise-reduction processing. Subong Kim (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, kim3578@purdue.edu) and Hari Bharadwaj (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Listeners with sensorineural hearing loss have significant difficulty understanding speech in noisy environments. Thus, current digital hearing aids employ noise-reduction (NR) algorithms to suppress background noise. Unfortunately, the NR algorithms distort some speech cues. Though considerable variability exists in hearing-aid users' reactions to these conflicting effects, little is known about the neural underpinnings contributing to the variance in perceptual outcomes with the addition of noise and the use of NR processing. Recently, we documented that neuroimaging-based measures of cortical responses can predict individual speech-in-noise perception benefits (or lack thereof) from NR processing. Here, we describe novel objective subcortical measures for quantifying individual noise tolerance and individual sensitivity to speech-cue distortions. Preliminary results show that the effect of adding noise and introducing NR processing on brainstem responses varies considerably across individuals. Because the brainstem responses are relatively unaffected by cognitive processes like selective attention, these preliminary results suggest an independent "bottom-up" contribution to individual variations in NR benefits. We will also discuss how the neurophysiological measures of subcortical encoding of speech sounds relate to individual profiles of peripheral (cochlear) pathophysiology and to individual perceptual outcomes.

5aPP2. Sound lateralization ability of elderly individuals in three onset conditions. Kazumoto Morita (Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, mtkkojiro@gmail.com), Moeko Shiroki, and Takeshi Toi (Chuo Univ., Bunkyo-ku, Tokyo, Japan)

Authors had previously reported that elderly individuals cannot sufficiently grasp the onset portion of sound. To further this research, the current study varied the sound conditions in the onset portion to three—(1) condition in which the onset portion on the delayed ear side was silent; (2) condition in which sound was presented to the onset portion of the delayed ear side as well; and (3) condition in which silence was inserted only during onset portion on the delayed ear, with no phase delay in the ongoing portion. Two sound frequencies of 1 kHz and 500 Hz were presented to 23 elderly participants and 22 young adults. Interaural Time Differences (ITDs) of 0.2, 0.4, 0.6, and 0.8 ms were relayed separately to each ear, in addition to the condition of zero ITD to both ears. In the experiments, a test stimulus sound was presented after the reference sound, and participants answered the direction of the sound from among three responses of "Right," "Left," or "Same" compared with the reference sound. The results of condition (3) obviously revealed that the elderly participants could not grasp the onset portion of sound compared to the younger participants.

5aPP3. Children's age matters, but not for audiovisual speech enhancement. Liesbeth Gijbels (Speech and Hearing, Univ. of Washington, 1715 NE Columbia Rd., Seattle, WA 98195, lgijbels@uw.edu), Jason D. Yeatman (School of Medicine, Div. of Developmental-Behavioral Pediatrics, Stanford Univ., Stanford, CA), Kaylah Lalonde (Ctr. for Hearing Res., 5. Boys Town National Res. Hospital, Omaha, NE), and Adrian KC Lee (Speech & Hearing Sci., Univ. of Washington, Seattle, WA)

Articulation movements help us identify speech in noisy environments. While this has been observed at almost all ages, the size of the perceived benefit and its relationship to development in children is less understood. Here, we focus on exploring audiovisual speech benefit in typically developing children ($N = 160$) across a wide age range (4–15 years) by measuring performance via an online audiovisual speech performance task that is low in cognitive and linguistic demands. Specifically, we investigated how audiovisual speech benefit develops over age and the impact of some potentially important intrinsic (e.g., gender, phonological skills) and extrinsic (e.g., choice of stimuli) experimental factors. Our results show an increased performance of individual modalities (audio-only, audiovisual, visual-only) as a function of age, but no difference in the size of audiovisual speech enhancement. Furthermore, older children showed a significant impact of visually distracting stimuli (e.g., mismatched video), where this had no additional impact on performance of the youngest children. No phonological or gender differences were found given the low cognitive and linguistic demands of this task.

5aPP4. Development of temporal cues perception in noise and reverberation. Ellen Peng (Boys Town National Res. Hospital, 1500 Highland Ave., Madison, WI 53705, ellen.peng@boystown.org) and Viji Easwar (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

When compared to adults, children's speech perception is particularly vulnerable to reverberation from indoor environments such as classrooms. Temporal acoustic cues, such as amplitude modulation (AM) and voice onset time (VOT), provide important information for perceiving speech in noise, and are susceptible to temporal smearing from reverberation. How children develop the ability to perceive distorted temporal cues is not well understood. Here, we compared school-age children between 9–14 years old with adults on their perception of temporal cues in quiet, in speech-shaped noise (SSN), in moderate reverberation, and in combined SSN with reverberation. In experiment 1, we measured children's just-noticeable-difference threshold of detecting AM depth at the rate of the fundamental frequency (F0; $\sim$100 Hz). In experiment 2, we measured children's categorical boundary of VOT using a voiced-voiceless /b/-/p/ consonant contrast. Preliminary

findings suggest that the elevation of AM thresholds at F0 is more strongly affected by noise rather than reverberation. On the contrary, the salience of VOT is harder to maintain in reverberation than in noise, resulting in a more biased percept toward voiced consonant. Further, our preliminary analysis suggests that adverse acoustics lead to more severe performance deterioration in children than adults for both temporal cues of interest.

5aPP5. Modeling the time course of cue weighting angle calculations. Sara Ng (Linguist., Univ. of Washington, Box 352425, Seattle, WA 98195-2425, sbng@uw.edu), Gregory M. Ellis (Commun. Sci. and Disord., Northwestern Univ., Chicago, IL), Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR), and Richard A. Wright (Linguist., Univ. of Washington, Seattle, WA)

A series of previous studies (Souza *et al*., 2015; 2018; 2020) developed a "cue profile" test based on synthetic speech sounds used to assess a hearing-impaired listener's use of speech cues. The resulting "weighting angle" quantifies how well individual listeners can utilize higher-precision spectro-temporal information (such as formant transitions), or whether they rely on lower-precision temporal (amplitude envelope) cues in consonant identification. The fact that the amount of hearing loss was not associated with the cue profile underscores the need to characterize individual abilities in a more nuanced way than can be captured by the pure-tone audiogram (Souza *et al*., 2020). One drawback to the current test is that it is time-consuming, making it impractical to deploy in clinical settings. This study employed an inter-trial stability metric based on deviations from a moving average to explore an early stopping point of fewer than 200 trials, rather than the full test of 375 trials, a significant reduction of testing time. A 3-way weighting angle classifier, trained via Long Short-Term Memory Network, was piloted for feasibility as an alternative time reduction method. The results, while preliminary, are encouraging. [Work supported by NIH, Grant No. R01 DC006014 (PI: Souza).]

5aPP6. Perceptual weighting of acoustic cues for word stress in normal hearing listeners and cochlear implant users. Justin T. Fleming (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, jtf@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Accurate perception of word stress is key to segmenting and understanding speech. We examined perception of four acoustic cues to word stress in English—vowel quality (VQ), pitch, syllable duration, and intensity—and how perception of these cues is affected by specific kinds of signal degradation. Cochlear implants (CIs) severely degrade VQ and pitch, which could demand different relative reliance on these cues. We tested NH listeners in unprocessed, vocoded, and simulated bimodal (unprocessed low frequencies, vocoded high frequencies) conditions as well as CI users. Stimuli included stress-contrastive word pairs (DESert versus dessERT, SUBject versus subJECT) with orthogonal manipulations of each cue. For both NH and CI listeners, there were strikingly different patterns of cue weighting between these word pairs, highlighting how stress cue weighting is flexible across words rather than being absolute. Vocoding the stimuli led to a down-weighting of VQ and pitch cues and a compensatory upweighting of duration and intensity; perception of pitch and VQ were largely restored in the bimodal conditions with 250 and 800 Hz bands preserved, respectively. CI users relied on duration more than the NH controls in any condition, while also showing a surprising ability to use VQ and pitch cues.

5aPP7. Singing proficiency of prelingually deaf children with cochlear implants as evaluated by a musician. Lisa S. Liu (Athens High School, Athens, OH 45701, lisaliu041@gmail.com), Helen S. Liu (Athens High School, Athens, OH), and Li Xu (CSD, Ohio Univ., Athens, OH)

Cochlear implant (CI) has become the standard clinical rehabilitation device for children born with hearing impairments. While CI has been successful in helping those children with speech and language development, many studies have found that children with CIs have trouble perceiving tones and pitches. Poor pitch perception would lead to poor vocal singing. In a previous study, samples of singing by normal-hearing children ($N = 43$) and deaf children with CIs ($N = 81$) were analyzed acoustically. The present study analyzed those 124 samples with human judgement (i.e., by a musician). The purpose of the present study was to evaluate the correlation between objective measures using acoustic analysis and subjective measures using an adult musician listener. The musician provided scores for each singing sample on several categories of pitch and tempo. The results showed that the children with CIs did well with tempo, but they had trouble singing the correct pitches and the overall listening pleasure was low. Furthermore, the scores of human judgements on pitch direction, pitch consistency, and pitch interval consistency were significantly correlated with the interval deviation computed from the acoustic analyses of the singing samples (with correlation coefficients ranging from 0.541 to 0.624, all $p < 0.05$).

5aPP8. Influence of hearing aid compression speed and channels on measures of psychophysical tuning curves. Marc Brennan (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, 4075 East Campus Loop South, Lincoln, NE 68583, Marc.Brennan@unl.edu) and Sarah Garvey (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, Lincoln, NE)

While much work has been completed on spectral resolution in listeners with SNHL, little attention has been paid to the impact of hearing-aid signal processing on access to spectral information. In this study, we hypothesized that increasing the hearing aid compression speed and number of compression channels would improve access to spectral information by providing better audibility across frequency. However, limited outer hair cell function or inhibition could curtail the benefit of improved audibility on spectral resolution. In this experiment, psychophysical tuning curves were measured for 25 participants with hearing loss. Both an excitation and phenomenological model of the auditory system were used to examine potential mechanisms that contributed to the observed results. The provision of amplification improved the measured sharpness of tuning for the low-frequency side. While decreasing the compression speed did not have a systematic effect on the measured psychophysical tuning curves, the low frequency side improved as the number of compression channels was increased. Threshold estimates from the excitation and phenomenological models were able to account for some of these observed trends in the psychophysical tuning curves, but the threshold estimates from the phenomenological model were more consistent with the measured thresholds.

5aPP9. ITD sensitivity provided to bilateral cochlear implant listeners by a mixed-rate, real-time capable processing strategy. Stephen R. Dennison (Elec. and Comput. Eng., Univ. of Wisconsin-Madison, 1500 Highland Ave., Waisman Ctr., Madison, WI 53705, srdennison@wisc.edu), Tanvi Thakkar (Psych. Dept., Univ. of Wisconsin-La Crosse, Lacrosse, WI), Alan Kan (School of Eng., Macquarie Univ., Sydney, New South Wales, Australia), Mahan Azadpour, Mario A. Svirsky (Dept. of Otolaryngology-Head and Neck Surgery at NYU Grossman School of Medicine, New York Univ., New York, NY), and Ruth Y. Litovsky (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

Bilateral cochlear implant (CI) listeners have limited access to interaural time differences (ITDs) at low frequencies in part because clinical processors do not coordinate the timing of stimulation across the ears. Further, clinical strategies, which are optimized for good speech reception, operate at stimulation rates that are too high [$\sim$1000 pulses per second (pps)] to enable ITD sensitivity from the timing of pulses. Our newly developed, real-time capable sound coding strategy, delivered using a bilaterally synchronized research processor, enables binaural coordination and pulse-timed ITD encoding. We employ a "mixed-rate" stimulation approach by maintaining the capacity to provide speech or envelope ITDs on high-rate channels while simultaneously providing ITDs in the pulse timing on low-rate (100 pps) channels. We hypothesized that a mixed-rate strategy encoding both high-rate envelope ITDs and pulse ITDs would yield better ITD sensitivity than a strategy encoding pulse ITDs or high-rate envelope ITDs only. This hypothesis was tested by measuring the perceived range of lateralized auditory images in bilateral CI listeners. Preliminary results indicate listeners may have ITD sensitivity with all conditions, though the mixed-rate strategy can provide better ITD sensitivity in some listeners. This demonstrates the

potential of the mixed-rate strategy for improving binaural hearing outcomes.

5aPP10. Individual perceptual profiles derived from automated psychophysical measures. Dana Cherri (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, dcherri@mail.usf.edu), David A. Eddins, and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Older adults with normal pure-tone thresholds or with age-related hearing loss face difficulties understanding speech in noisy environments. Despite this common challenge, the potential root deficits are likely heterogeneous across listeners and vary in the extent of severity beyond pure tone sensitivity. To better serve patients with listening challenges, it is believed that more precision in diagnostics and treatment are currently needed, and to that end, the purpose of the present study was to achieve more robust auditory perceptual profiles from a diverse population. Specifically, the present study tested three groups of listeners (young, normal hearing; older, normal hearing, and older, hearing-impaired; n = 20 per group) on a large battery of psychophysical measures using the Portable Automated Rapid Testing (PART) platform, which includes automated, unsupervised measures of: spectral resolution, temporal resolution, informational masking, binaural processing, working memory, and visual-spatial processing. Data show significant differences between the younger and older listeners, and broad variations in perceptual measures among the older listeners with normal hearing and hearing loss. These multi-dimensional variations were reduced using confirmatory factor analysis to reveal unique perceptual profiles that one day may inform targeted intervention options. [Work supported by NIHR21DC017832 and NIHR01DC01505.]

5aPP11. Identifying subclinical hearing problems. Ward R. Drennan (Otolaryngol., Univ. of Washington, VM Blodel Hearing Res. Ctr., UW Box 357923, Seattle, WA 98195, drennan@uw.edu), Lauren Langley (Otolaryngol. and Speech and Hearing Sci., Univ. of Washington, Seattle, WA), and Zeyu Wei (Statistics, Univ. of Washington, Seattle, WA)

Noise exposure and aging can cause subclinical hearing problems that are not identified with conventional audiometry. Potential clinical assessments were studied that could identify subclinical hearing deterioration: (1) Electrocochleography (ECoG); (2) Middle Ear Muscle Reflex (MEMR); (3) Extended High Frequency (EHF) thresholds (10–16 kHz); and (4) word recognition in noise. Sixty-one young, *normal-hearing* people were recruited. Each completed the four auditory assessments and the Life-Time Exposure to Noise Solvents Questionnaire (LENS-Q). ECoG and MEMR measurements were measured as a function of sound level using clinical equipment. Word recognition in noise was measured with a diotic, closed-set, 12-alternative spondees-in-noise test. It was hypothesized that normal-hearing individuals reporting more noise exposure would have slower growth of the ECoG measures and MEMR magnitudes with increasing levels. Individuals reporting higher noise exposure with the LENS-Q had significantly reduced MEMR magnitudes and smaller slopes, supporting the hypothesis. Gender effects were observed. No significant effect of self-report noise-exposure was seen with ECoG AP growth functions or SP/AP ratios, though modest age effects were observed. EHF thresholds were the only significant predictor of word recognition in noise. The results suggest that word recognition in noise, EHF audiometry, and MEMR could indicate subclinical hearing problems.

5aPP12. Auditory selective attention and speech-in-noise in cochlear implants. Jae Hee Lee (Commun. Sci. & Disord., Univ. of Iowa, 1105 Lake Shore Dr., Iowa City, IA 52246, jlee168@uiowa.edu), Mallory Orr, Hwan Shim, and Inyong Choi (Commun. Sci. & Disord., Univ. of Iowa, Iowa City, IA)

Auditory selective attention is a crucial mechanism for comprehending speech in noise. We investigated if cochlear implant (CI) users exhibit auditory selective attention that involves modulation of neural responses to target speech and if such attentional ability predicts their speech-in-noise performance. In our experiment, designed to assess the strength of attentional modulation, participants with normal hearing and with CI were given a pre-stimulus visual cue that directed their attention to either of two sequences in stationary background noise and asked to select a deviant syllable. We hypothesized that the amplitude of event-related-potential (ERP) would be greater when the syllable is attended if either group is capable of employing auditory selective attention and that the difference of ERP amplitude between attended and unattended trials predicts the performance in a speech-in-noise task. Our analysis showed that the amplitude of ERPs for the attended syllable was greater than that for the unattended syllable with the CI subjects, exhibiting that attention modulates CI users' cortical responses to sounds. Moreover, the strength of attentional modulation showed a significant correlation with the same CI users' speech-in-noise performance. These results show that attentional modulation provides a valuable neural marker for predicting CI users' success in real-world communications.

5aPP13. Lateralization biases for narrowband stimuli in listeners with typical and symmetrical hearing thresholds. Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., College Park, MD 20742, goupell@umd.edu), Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), and H. S. Colburn (Dept. of Biomedical Eng., Boston Univ., Boston, MA)

In listeners with normal audiometric thresholds, it is assumed that stimuli presented over headphones with zero interaural differences are perceived to be centered in the head. In this study, intracranial lateralization was measured by having ten young adults with normal and interaurally symmetric thresholds (within 5 dB) point to perceived sound locations along a visual reference scale. The stimuli were either tones or 50-Hz-bandwidth narrowband noises, at frequencies of 300, 400, 500, 600, and 700 Hz. A range of interaural time differences (ITDs) was applied while the interaural level difference was always zero. To diminish any transducer-based effects, the headphones were reset and reversed between each of ten blocks of trials. Lateralization curves for five of the ten listeners showed a substantial bias to one ear at a subset or all frequencies tested; the worst individual case demonstrated a perceived 60% lateralization bias (from the midline to the left ear) for a zero ITD. This result challenges a long-standing assumption held in binaural hearing and, while similar results have been observed previously, the basis of the phenomenon remains largely unexplored.

5aPP14. Development of an auditory fitness-for-duty standard that predicts performance in military hearing tasks from auditory thresholds and performance on the 80-word modified rhyme test. Douglas S. Brungart (Walter Reed NMMC, 4401 Holly Ridge Rd., Rockville, MD 20853, dsbrungart@gmail.com), Benjamin Sheffield (US Army Public Health Ctr., Bethesda, MD), Matthew J. Makashay (US Army Public Health Ctr., Aberdeen Proving Ground, MD), and Hector Galloza (dosGaRi Solutions, Aguada,)

Military operations often require Service Members (SMs) to make life-and-death decisions based on information they can only obtain auditorily. However, the wide variety of critical sounds, essential spoken messages, and masking environments encountered while performing military missions makes it difficult to use clinical tests to predict the operational performance of hearing-impaired SMs. In this study, $\approx$2400 SMs performed listening tasks derived from binaural recordings made during realistic battlefield exercises at the US Army Joint Readiness Training Center. The listening tasks included standardized speech recognition tasks based on the Oldenburg Matrix Test and Modified Rhyme Test (MRT), as well as scenario-based multiple-choice tasks where questions were asked about unmodified verbal exchanges that occurred between SMs engaged in the battlefield simulation. The scores SMs achieved on these operational tasks were then compared to their hearing thresholds and to their scores on the MRT_{80}. The results show that an auditory fitness-for-duty standard that requires individuals with thresholds that exceed the values defined in the new US Army H3 Profile to obtain a minimum score on the MRT_{80} is generally able to identify individuals who are most at risk of performing abnormally poorly on militarily relevant listening tasks. [The views expressed in this abstract are those of the authors and do not reflect the official policy of the Department of the Army, Department of Defense, or U.S. Government.]

5aPP15. Speech Intelligibility evaluation in the presence of speech masker of cochlear implant in a reverberant room. Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), Muhammad A. Asyraf, Ria R. Amelia, and Latifatul Fajriyah (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya, Indonesia)

In this report, we investigated the performance of a cochlear implant (CI) in a reverberant room with the presence of a competing speech masker. We compared Short-Time Objective Intelligibility (STOI) and Perceptual Speech Quality (PESQ) to represent the objective, to percent correct word (PCW) and Mean Opinion Score (MOS) as subjective evaluations, respectively. The target and the masker speech were convoluted with 0.4- and 0.7-s reverberation time, respectively, then processed with a 14-channel vocoder. We set the target to the masker 5dB and 0 dB SNR. We simulated two well-known techniques, namely CIS and SPEAK processors. Ten normal-hearing paid subjects were participated to evaluate the performance in clean, reverberant with no masker and reverberant with masker. The results showed that performance degraded significantly as the reverberation duration increased. Furthermore, the presence of a masker to a target speech in a reverberant room worsened all the evaluated performances compared to the reverberant alone. The results suggested that further investigation is required to reduce the masker effect to improve CI intelligibility.

5aPP16. Individual band-importance functions for children with normal hearing and children with hearing loss. Ryan McCreery (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE, Ryan.McCreery@boystown.org), Adam K. Bosen, Meredith Spratford (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Elizabeth Walker (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA, IA)

Band-importance functions represent the contributions of frequency regions to speech recognition. Average band-importance functions for adults with normal hearing are used to estimate speech intelligibility and in hearing-aid signal processing. Children and listeners with hearing loss show considerable individual variability in band-importance, but the factors that contribute to differences in band-importance are not well-understood. Two experiments were conducted to examine auditory and developmental factors that predicted individual band-importance for children with normal hearing and children with hearing loss. In the first experiment, the effects of receptive vocabulary and selective attention on band-importance for 5–12 year-old children with normal hearing was measured for words and nonwords. Our hypothesis was that children with larger vocabularies and better selective attention skills would have larger importance weights for mid-frequencies, a pattern consistent with adult band-importance functions than children with poorer skills in these domains. In the second experiment, the effects of audibility on band-importance functions for a group of children with mild-to-moderate sensorineural hearing loss was measured for words. Our hypothesis was that band-importance would be higher for bands where children had better audibility through their hearing aids. The results confirmed these hypotheses. Implications for future research are discussed.

5aPP17. Binaural filterbank hearing aid simulation V5. Ramesh Kumar Muralimanohar (Dept. of Speech, Lang., and Hearing Sci., Univ. of Colorado, 409 USB, 2501 Kittredge Loop Dr., Boulder, CO 80309, muralima@colorado.edu), James M. Kates, and Kathryn Arehart (Dept. of Speech, Lang., and Hearing Sci., Univ. of Colorado, Boulder, CO)

In this presentation, we describe the Binaural Filterbank Hearing Aid Simulation (BFHAS) which is a MATLAB-based package that simulates commonly available hearing aid signal processing. This package allows the user to process custom audio signals (e.g., noisy speech; music) with a suite of hearing aid algorithms. Linear or non-linear fitting prescription fitting formulae can be used for gain compensation. The hearing-aid algorithms include spectral-subtraction noise suppression, wide dynamic-range compression (WDRC), and frequency compression based on sinusoidal modeling of the high frequencies of the signal. Users can vary signal processing settings like number of bands; add/modify spectral subtraction; gains, knee points, attack and release times for WDRC; compression ratios and cutoffs for frequency compression etc. The simulation provides the processed hearing aid output that includes the compensation and hearing aid quality and intelligibility metrics for the input stimuli. Future updates will include the room-simulations and built-in noise generators to simulate different levels of signal degradation. We also provide a user guide and examples illustrating the use of different package modules. This package will be made available for download. [Work supported by NIH R01 DC012289 and by a grant from GN ReSound to University of Colorado.]

5aPP18. The timing of deploying and withholding listening effort, in listeners with normal hearing or with cochlear implants. Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall Rm. 115, Minneapolis, MN 55455, mwinn@umn.edu) and Katherine Teece (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Speech perception is more effortful when the listener needs to fill in a missing word that was masked or misperceived. However, that effort could come at the cost of hearing the next utterance, as conversational speech is continuous. Speech perception is typically tested one utterance at a time, critically overlooking this problem. The current study measured how effort is deployed in the moments of processing one sentence while balancing the need to continue attending or not attending to later speech. Stimuli included sentences where occasionally noise masked a single word that was recoverable from later context. Sentences were followed by silence or three random digits; participants were tested in blocks where they either ignored the digits or repeated them along with the full sentence. Effort was tracked via pupillometry. Listeners with normal hearing front-loaded effort when anticipating subsequent inattention, and spread effort more shallowly across time when anticipating later attended information. Listeners with cochlear implants did not show this local control over the timing of effort. Both groups showed greatest increase in effort in recovering missing words when listening to a single utterance, suggesting that capacity to exert effort is mediated by (and spread across) local demands of the task.

5aPP19. Predicting the effect of hearing-protection devices on horizontal-plane sound localization. Nathaniel J. Spencer (Air Force Res. Lab., 2610 Seventh St., Wright-Patterson AFB, OH 45433, nathaniel.spencer.1.ctr@us.af.mil), Zachariah N. Ennis, Natalie Jackson, Brian D. Simpson, and Eric R. Thompson (Air Force Res. Lab., Wright-Patterson AFB, OH)

Hearing-protection devices (HPD) are critical to hearing health in hazardous noise environments, but can be detrimental to job effectiveness and safety when degrading a wearer's sound localization accuracy. Understanding specifically how localization is disrupted by HPDs is critical for mitigating the deleterious effects of HPDs on situation awareness. This study attempts to better quantify the effects of the degradations by predicting horizontal sound localization. A 360 deg horizontal array with 10 deg between-speaker spacing is used to collect head-related transfer functions (HRTFs) and to measure localization [ANSI/ASA S3.71] for the different conditions (open-ear, passive plugs, and both passive and active muffs). Our model attempts to select the response location that was most likely to have elicited the observed stimulus for the trial, by comparing narrow-band statistics between prior knowledge ("templates") and trial observations. Bone conduction related noise is added, and the degree to which the priors are "device-related" (i.e., based on HRTFs captured for the device) or "open ear" related is varied. Results are discussed in terms of two potential mechanisms: (1) the wrong location is chosen due to a prior/template mismatch in percent device-related and (2) multiple response locations are viable because cue variability is reduced between locations compared to within.

5aPP20. Simulated audiometry reveals multiple asymptotic thresholds for the same listener depending on the number of presentations at threshold. Eric C. Hoover (Dept. of Hearing and Speech Sci., Univ. of Maryland, 0100 LeFrak Hall, 7251 Preinkert Dr., Columbia, MD 20742, ehoover@umd.edu) and Katherine N. Palandrani (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Changes in audiometric thresholds are used to detect auditory insult resulting from exposure to hazardous noise, blasts, and ototoxicity. Stimuli presented during audiometry reflect a stochastic process driven by listener

responses, and thus a threshold can be reported after a different number of presentations, e.g., two hits and zero, one, or two misses. We interpret thresholds as equivalent despite large differences in the proportion of hits observed. To test for systematic bias in thresholds obtained after a different number of presentations, a continuous distribution of audiometric threshold as a function of stimulus level was numerically approximated for a fixed listener model tested using 2 and 5 dB ascending steps. Results were sorted by the number of presentations at threshold and summary statistics were calculated directly from the proportion of Monte Carlo simulations reporting each stimulus level as threshold. Thresholds differed systematically as a function of the number of times the threshold stimulus level was presented, consistent with a model combining the probability of meeting the stopping criteria with the asymptotic distribution of ascending trials. The implication is that recording the number of presentations at threshold on the audiogram will improve our ability to detect changes in hearing. [R01DC015051.]

5aPP21. Electrophysiological responses to change in spectral ripple envelope phase in normal-hearing infants. David Horn (Otolaryngology-HNS, Univ. of Washington, Seattle, WA), Max Walter (Univ. of Washington, Seattle, WA), Jay T. Rubinstein (Otolaryngology-HNS, Univ. of Washington, Seattle, WA), and Bonnie K. Lau (Otolaryngology-HNS, Univ. of Washington, 1715 NE Columbia Rd., Box 357988, Seattle, WA 98195, blau@uw.edu)

Spectral resolution plays an important role in understanding speech in noisy real-world listening environments. Spectral ripple discrimination is one method used to measure spectral resolution that has been shown to correlate with open-set speech understanding in adults and children who use cochlear implants. In this study, we investigate the feasibility of obtaining an electrophysiological measure of spectral ripple discrimination in very young infants. Spectral ripple stimuli were constructed from two concatenated 1-s noise carriers consisting of 2555 random phase pure tone frequency components between 100–5000 Hz, with a spectral envelope sinusoidally modulated at 1-ripple-per-ocatave with peak-to-trough depth of 20 dB. The starting phase of the first half of the stimulus varied randomly while the starting phase of the second half was "inverted", shifted by 90-deg, relative to the first. Infants were presented with 150 repetitions of these stimuli at 80 or 65 dBA in sound field while undergoing 32-channel electroencephalography (EEG) recording. Our preliminary analyses show that an acoustic change response can be recorded in awake infants between 2 and 4 months of age in response to a spectral ripple inversion. These preliminary results suggest the potential utility of EEG in the measurement of spectral resolution in very young infants, warranting further investigation.

5aPP22. Comparison of speech, spatial, and qualities of hearing scale questionnaire and the abbreviated profile of hearing aid benefit in a large cohort of listeners. Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu) and Sadie O'Neill (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

The speech, spatial, and qualities of Hearing Scale (SSQ; Gatehouse and Noble, 2004) is a 49-item questionnaire developed to assess listeners' subjective sense of listening ability and listening experience in everyday complex situations that often involve spatial hearing. The Abbreviated Profile of Hearing Aid Benefit (APHAB; Cox and Alexander, 1995) is a 24-item self-assessment questionnaire in which listeners report the amount of trouble they have while communicating in various everyday listening situations. The APHAB assesses multiple listening environments that include ease of communication, reverberation, background noise, and aversiveness of sounds. Even though both the questionnaires measure the self-reported hearing ability/disability, there are very few studies in the literature that compares the outcome of these two questionnaires on same group of listeners. Both the questionnaires were administered to198 listeners (younger: n = 107, mean age = 22.8 years, range: 19–25 years; older: n = 91; mean age = 53.5 years, range: 40–70 years) through Qualtrics. The differences and relationships between the two groups for both the questionnaires will be discussed. These data are intended to aid in interpreting the SSQ and APHAB results in typical audiological clinics.

5aPP23. Central gain in aging, tinnitus, and temporary hearing loss. Kelsey Dougherty, Alexandra Hustedt-Mai, Anna Hagedorn (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN), and Hari Bharadwaj (Speech, Lang., and Hearing Sci., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, hbharadwaj@purdue.edu)

The nervous system adapts in many ways to changes in the statistics of the inputs it receives. An example of such plasticity observed in animal models is that central auditory neurons tend to retain their driven firing rate outputs despite reductions in cochlear input due to hearing loss or deafferentation. The perceptual consequences of such "central gain" are unknown; pathological versions of such gain are often hypothesized to underlie tinnitus and hyperacusis. To investigate central gain in humans, we designed an electroencephalogram (EEG)-based paradigm that concurrently elicits robust separable responses from different levels of the auditory pathway. Using this measure, we find that cortical responses are relatively invariant despite a large monotonic decrease in auditory nerve responses with age, and that this central gain is also associated with perceptual deficits in comodulation processing. We then applied the same measures to a cohort of individuals with persistent tinnitus and to a third cohort where a week-long monaural conductive hearing loss was induced using silicone earplugs. Overall, our results suggest that central gain is ubiquitous in response to reduced peripheral input and may affect auditory scene analysis, but does not in itself account for tinnitus perception.

5aPP24. The relationship between speech perception and production in 4-year-old children with hearing loss. Monika Maria Oster (Listen and Talk, 14305 24th Ave. NE, Seattle, WA 98125, mona.goeser@gmail.com)

The ability to produce speech is linked to accurately perceiving speech, which is challenging for children with hearing loss (HL). Additionally, the interactions through which children acquire speech frequently occur in difficult listening environments. This preliminary study aims to describe the relationship between perception of quiet speech/speech in noise and speech production skills in 4-year-old children with HL. The analysis used clinical assessment data. Only children with mild to profound bilateral HL using hearing aids (HAs) or cochlear implants (CIs) were included. Articulation was assessed through the GFTA (HA = 39, CIs = 43). Speech perception was assessed in 23 children through the NU-CHIPS, WIPI or PBK-50 in two conditions: 35dB in quiet and 50dB in 45dB multitalker babble. The results showed that speech production skills were similar for children with HAs (mean SS = 83.77) and CIs (mean SS = 79.88) and were not correlated with speech perception. However, results may have been influenced by the verbal responses required for the PBK-50. After its exclusion, results showed a moderate positive correlation between speech production and the perception of quiet speech ($r = 0.5$) and speech in noise ($r = 0.66$).

Session 5aSAa

Structural Acoustics and Vibration, Engineering Acoustics and Physical Acoustics: Novel Methods for Energy Dissipation in Structures

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Invited Papers

8:20

5aSAa1. Application of methods to determine time-dependent damping of flanged, bolted plates. Trevor W. Jerome (NSWCCD, GTWT Water Tunnel, State College, PA 16804, twj115@psu.edu)

The dynamic and steady-state response of flanged, bolted plates when excited with a hammer and acoustic methods exhibit nonlinear effects. Established Hilbert methods are used to estimate time-dependent and amplitude-dependent damping due to the joint, and the findings are presented here. Nonlinear damping of this system is sensitive to boundary conditions, and each of the first few modes behave nonlinearly in unique ways. Insight gained from knowledge about relationships between amplitude and damping may help to better understand friction joint dynamics.

8:40

5aSAa2. Broadband vibration attenuation via nonlocal acoustic black hole metastructures. Siddharth Nair (Mech., Ray W. Herrick Labs., Purdue Univ., West Lafayette, IN 47907, nair40@purdue.edu), Mehdi Jokar, and Fabio Semperlotti (Mech., Purdue Univ., West Lafayette, IN)

In recent years, the application of Acoustic Black Holes (ABH) to passive vibration attenuation, vibroacoustic control, and vibration-based energy harvesting of thin walled structures has seen a rapidly growing interest. It is well known that the dynamic operating range of an ABH is limited by the existence of a cut-on frequency below which the ABH is not effective. The specific value of this frequency bound is connected to the ABH characteristic dimension, typically its diameter. In order to expand the dynamic operating range and lower the cut-on frequency, ABH features require larger diameters. However, design and manufacturing constraints impose stringent limitations that are often incompatible with required ABH dimensions. This work introduces the concept of *nonlocal ABH metastructures* and explores the feasibility of using intentionally introduced nonlocality to extend the dynamic operating range of periodic ABH lattices towards the lower frequencies. The role of nonlocality is investigated via a dedicated semi-analytical model, and the dynamic performances of different nonlocal designs are studied numerically. Results show a remarkable ability of the nonlocal design to expand the operating range of the ABH metastructure and to achieve vibration attenuation behavior at frequencies significantly below the cut-on frequency of the individual ABH.

9:00

5aSAa3. Transmission loss of plates with embedded multi-scale and tuned acoustic black holes. Yu Xiong (Penn State, 606F Oakwood Ave., State College 92602, U.S. Minor Outlying Islands, xiongyu@outlook.com), Edward Smith, and Stephen C. Conlon (Penn State, State College, PA)

The Acoustic Black Hole (ABH) is an emerging method for developing lightweight, high-loss structures and has recently drawn extensive attention. Unlike many conventional methods, ABH implementations are typically able to reduce structure vibration and sound radiation without a net addition of mass. Transmission loss (TL) is a key criterion for panel structure in the noise and vibration industry. Many noise and control applications demand panel structures with high TL and low weight. Plates with embedded ABH cells have the potential to act as good candidates for TL applications. In this work, the transmission loss (TL) of an embedded multi-scale ABH plate was studied. The embedded large and small ABH cells were particularly designed to cut on below and above the critical frequency of the plate, respectively. The results were compared with a uniform plate and an embedded single-scale ABH plate. Discrete tuning masses were attached at the ABH cells' center to manipulate the ABH cut-on modes to increase the TL further. The results show that the damped multi-scale ABH plate achieved a 10 dB TL increase, flattened the TL curve, and nearly eliminated the plate coincidence dip. Manipulating the high loss ABH modes by adding tuning masses (20 g each) demonstrated a 2 dB increase at low frequencies within the mass-law range. Although damping material was applied, adding some mass and an overall weight advantage were still attained compared to the uniform plate. The damped multi-scale ABH plate is 7% lighter than the uniform plate.

9:20

5aSAa4. Damping of vibration-isolated structures: A review. Vyacheslav M. Ryaboy (Light and Motion, MKS, 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

Vibration presents a major challenge to advanced experiments in physics and life sciences, as well as to precise manufacturing processes. A vibration-isolated platform had become an indispensable element of precision experimental and manufacturing systems. Ideally, the platform should behave as an absolutely rigid body; unfortunately, absolutely rigid bodies do not exist: any structure deviates from the rigid behavior at its resonance frequencies. Resonance vibrations of the platform carrying precision optomechanical or acoustical equipment can be excited by acoustical inputs, residual vibrations coming from the floor through vibration isolators, and atmospheric turbulence. These vibrations must be mitigated by dissipation of mechanical energy, or damping. This paper summarizes current methods and achievements in active and passive damping of optical tables and breadboards, applicable also to other types of structures. Dynamic response of an isolated platform is analyzed. Physical principles of active damping systems, multiple tuned dynamic absorbers, and elastomeric constrained-layer damping links are presented; various designs of damping devices are considered, and their operation illustrated by experimental results; advantages and shortcomings of each method are discussed.

9:40

5aSAa5. Experimental evaluation of carbon/epoxy laminates with concentrated carbon nanotube interlayers for high damping. Avery Brown (Eng. Sci. and Mech., Penn State, 7 Res. West Bldg., State College, PA 16802, awb5709@psu.edu), Charles E. Bakis (Eng. Sci. and Mech., Penn State, State College, PA), and Edward Smith (Aerosp. Eng., Penn State, State College, PA)

Carbon nanotubes (CNTs) embedded in carbon/epoxy (c/ep) composites offer a lightweight, stiff solution with high damping for structural components. To manufacture CNT/c/ep hybrid composites with high concentrations of CNTs, a CNT yarn interlayer concept is used. Through a mechanism known as stick-slip, the interface between CNTs and c/ep laminates dissipate energy during dynamic cycling. It is predicted that altering the interfacial bond strength can enhance dynamic properties: *loss factor, loss modulus, and storage modulus*. In this research, the effect of surface treatments on CNT yarns prior to inserting the CNTs into the laminate was explored to determine if the dynamic properties of the composite were enhanced. It was determined that 2,3-dibromo-1,4-butanediol (23D14B), Triton X-100 (TX) and a solution of sulfuric and nitric (S/N Acid) were the best treatments of 12 tested based on the increased dynamic properties. For a composite with 5 vol% CNT yarn interlayers, 23D14B, TX, and S/N Acid increased *Loss Factor* by 230%, 130%, and 160%, respectively. Elevated temperature, moisture-saturated/elevated temperature, and cyclic loading frequency were investigated on 5 vol. % CNT interlayers to determine their effects on dynamic properties of the CNT hybrid composites.

Session 5aSAb

Structural Acoustics and Vibration: General Topics in Structural Acoustics

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Allison Kaminski, Cochair
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Contributed Papers

10:15

5aSAb1. Direct transmissibility and structural acoustics. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, anthony.bonomo@gmail.com) and Matthew Craun (Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD)

Direct transmissibility is a useful concept developed for transfer path analysis of linear systems. Defined loosely as the quotient between the responses at degrees-of-freedom i and j when i is excited and all other degrees of freedom are blocked, the direct transmissibility provides information on the connectivity of a system that more conventional transmissibility quantities cannot. Despite its apparent usefulness, the application of the direct transmissibility concept to structural acoustics has been somewhat limited. In this talk, the definition of direct transmissibility will be reviewed and the utility of this concept will be demonstrated through application to a few canonical structural acoustics systems. The extension of the direct transmissibility concept from spatial to modal coordinates will also be explored. [Work supported by ONR.]

10:30

5aSAb2. Analysis of modified structures by Rayleigh quotient. Allison Kaminski (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, allison9@bu.edu) and James McDaniel (Mech. Eng., Boston Univ., Boston, MA)

In structural vibrations, the Rayleigh quotient may be used to calculate the natural frequency (or eigenvalue) of a structure given the corresponding mode shape (or eigenvector). Previous works have shown that the eigenvalue may be calculated relatively accurately using the Rayleigh quotient as long as an appropriate guess is made for the eigenvector. Typically, the Rayleigh quotient is used to predict the eigenvalues when the structure does not change. However, in this work the structure will be modified and the change in the eigenvalue will be predicted using the Rayleigh quotient. In order to use the Rayleigh quotient a guess for the eigenvector must be made. Here, the displacement vector of the nominal structure forced near a resonance will be used as the guess. Analysis and computations will be done for an undamped beam that is harmonically forced. To modify the structure, the stiffness elements will be scaled. Using this approach, it will be demonstrated that changes in the natural frequencies can be predicted relatively quickly for a modified structure. In addition, this method provides insight into which elements should be scaled to get the greatest frequency change. [Work supported by ONR under Grant N00014-19-1-2100.]

10:45

5aSAb3. Experimental and numerical validation of vibration-based sound power measurements of arbitrarily curved panels. Trent P. Bates (Mech. Eng., Brigham Young Univ., Provo, UT 84602, tbateslefty24@gmail.com), Ian C. Bacon, Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Vibration-based sound power (VBSP) measurement methods are of interest due to their potential versatility in application compared to methods based on sound-pressure or sound-intensity. The VBSP method is based on the elementary radiator approach that relies on the acoustic radiation resistance matrix. Previous work has validated the form of the radiation resistance matrix for flat plates, cylindrical- and spherical-shells, and simple-curved plates. A form specific to arbitrarily curved structures has not been developed. Experimental VBSP measurements of two arbitrarily curved panels are shown to have excellent agreement with results from the ISO 3741 method. The VBSP method was applied by using the simple-curved plate form of the radiation resistance matrix and mapping each arbitrarily curved panel with a constant-radius curve fit. Numerical boundary element models, that inherently use the true form of the radiation resistance matrix, were also used to compute the sound power from the same curved panels. Surface velocity data from the numerical models were used to also compute sound power using the VBSP method with the simple-curved plate form of the radiation resistance matrix. Sound power results show excellent agreement between the numerical model and VBSP method. [Work supported by the National Science Foundation.]

11:00

5aSAb4. Understanding the vibration response and acoustic radiation of a transient excited thin structure. Allison M. King (Dept. of Mech. Eng., Univ. of Michigan, Univ. of Mich., 1231 Beal Ave., Ann Arbor, MI 48109, kingalli@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Remote sensing may be used to detect and locate acoustic sources on a structure as well as predict the geometry and material of the structure based on propagating vibration and acoustic waves. Use of remote sensing measurements to detect and locate impacts on a thin structure is desirable. For a transient excited thin structure, the coupling of the vibration waves in the structure and the acoustic waves radiating from the acoustic source can introduce difficulties with using traditional source localization techniques. To study this problem, the vibration of the structure must be modeled and

understood before source localization can occur. Repeatable axisymmetric transient impact experiments are conducted by dropping a stainless-steel ball bearing on a 3.2-mm-thick aluminum plate surrounded by air on all sides. Accelerometers are utilized to verify vibration models based on a frequency-domain Green's function as well as a finite element simulation. Wave speeds in the plate ranging between 420 and 1225 m/s are considered. A linear array of 13 microphones with 0.025 m spacing located just above the plate is used to study the resulting acoustic radiation. An acoustic-structure interaction model is developed based on these wave propagation measurements. [Sponsored by NAVSEA.]

11:15

5aSAb5. Rail vibration and sensitive equipment—A study of mitigation opportunities in building concept design. Aamna Arshad (Acoust. and Vib., SLR Consulting, 150 Res. Ln., Guelph, ON N1G 4T2, Canada, aarshad@slrconsulting.com) and Brad Pridham (Acoust. and Vib., SLR Consulting, Guelph, ON, Canada)

In the early stages of design for a building housing sensitive equipment there is a general level of uncertainty about how rail vibration will impact the development site. As vibration propagates from the rail line, through the ground, and within the building to the equipment, spectral analysis will show significant changes as certain frequencies are attenuated or amplified from one medium to another. Therefore, it is important to consider the entire system (soil, foundation, and floor framing) when making judgements about site feasibility. This also means that mitigation can be approached at several different points in the system, with considerable variation in cost-effectiveness. The US FTA detailed vibration analysis method provides some general guidance on foundation attenuation and floor resonances. This paper provides a detailed case study examining measured vibration data adjacent to a freight rail line. The investigations illustrate the potential for modifications to a building design in the concept/site feasibility stage to mitigate vibration impacts on sensitive equipment and reduce or in some cases eliminate the need for isolation systems.

11:30

5aSAb6. Active noise and vibration control of systems with primary path nonlinearities using FxLMS, Neural Networks and Recursive Neural Networks. Xander Pike (ISVR, Univ. of Southampton, ISVR, University Rd., Highfield, Southampton SO17 1BJ, United Kingdom, ap1u20@soton.ac.uk) and Jordan Cheer (ISVR, Univ. of Southampton, Southampton, United Kingdom)

Active control systems are often used to surmount the challenges associated with using passive control measures to control low frequencies, since they achieve control without the application of large or heavy control treatments. Historically, linear active control strategies have been used in feedforward control systems to drive the control source to minimise the signal measured at the error sensor. However, when the response from noise source to error sensor becomes nonlinear, either in the primary or secondary path, the performance of such controllers can suffer. To overcome this limitation, it has previously been shown that Neural Networks (NNs) can outperform linear controllers. Furthermore, Recursive Neural Networks (RNNs) have been shown to outperform classical feed-forward networks in some cases. This is usually explained by the RNNs ability to form a rudimentary "memory." This paper compares the behaviour of the linear FxLMS algorithm, a NN and an RNN through their application to the control of a simulated system with variable levels of saturation and hysteretic nonlinearities in the primary path. It is demonstrated that the NN is capable of greater control of saturation nonlinearities than FxLMS. Similarly, the RNN is capable of greater control of hysteretic nonlinearities than the NN.

11:45

5aSAb7. Flutter prediction on combined EPS and carbon sandwich structure for light aircraft wing. Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id)

Flutter prediction is an important step before conducting a flight test. In this study, we performed flutter prediction of a half-wing structure without control surfaces. The half-wing structure is made to resemble the scaled-down wing of a Boeing 737 NG at a scale of 1:39.34. The airfoil profile used is the wing profile of the Boeing 737 NG obtained from airfoiltools. The structure is constructed using a combination of carbon sandwich and EPS. The advantages of choosing this material are its low-cost and easy manufacture. We used the p-k method in the FEMAP software for flutter prediction. From the prediction results, the calculated flutter speed is ~14.5 m/s. The flutter mode shape is a combination of lateral bending and twist. Dimensional analysis was also carried out to predict the maximum speed on the scaled model and predicted at 27.88 m/s. Based on calculated flutter speed, the maximum operating speed of a constructed structure should be far less than the flutter speed. Thus, the structure's maximum speed is below the predicted value. Based on carried out flutter prediction, the wing structure, constructed using a combination of carbon sandwich and EPS, can fly safely at a maximum cruise speed of 10 m/s.

Session 5aSC

Speech Communication, Psychological and Physiological Acoustics, ASA Committee on Standards, and Signal Processing in Acoustics: Speech and Machines

Richard A. Wright, Cochair
Department of Linguistics, University of Washington, Box 352425, Seattle, WA 98195-2425

Gina-Anne Levow, Cochair
University of Washington, Seattle, WA 98195

Chair's Introduction—9:00

Invited Papers

9:05

5aSC1. Inclusive machine intelligence and its promise for speech-centered societal application. Shrikanth Narayanan (Univ. of Southern California, 3740 McClintock Ave. EEB400B, Los Angeles, CA 90089-2564, shri@sipi.usc.edu) and Dani Byrd (Linguist., Univ. of Southern California, Los Angeles, CA)

Developments across the machine intelligence ecosystem, from sensing and computing to data sciences, are enabling new possibilities in advancing speech science and the creation of speech-centric societal technologies. A critical aspect of this endeavor requires addressing two intertwined challenges: illuminating the rich diversity in the speech across people and contexts and creating inclusive, trustworthy technologies that work for everyone. This talk will highlight three specific areas of advancement. The first is the capturing and modeling the human vocal instrument during speaking and selected related technological and clinical applications that leverage this technology (Hagedorn *et al*., 2019). The second relates to speech-based informatics tools for supporting screening, diagnosis and treatment of behavioral and mental health with broad access and scale. For example, remote multimodal sensing of speech cues can enable new ways for screening and tracking behaviors (e.g., stress) that can progress to treatment (e.g., for depression) and offer just-in-time support (Bone *et al*., 2017). The final domain highlights the use of speech machine intelligence tools to analyze media, including film, television shows, news and advertisements. These tools provide insights about representation and portrayals of individuals along dimensions of inclusion such as gender, age, ability, and other attributes (Somandepalli *et al*., 2021).

9:25

5aSC2. The non-native canonical accent—And how to use it. Michael Tjalve (Linguist., Univ. of Washington, Guggenheim Hall 4th Fl., Seattle, WA 98195, mtjalve@uw.edu)

One of the key benefits of modern globalization is cross-pollination of cultures. This global community encourages diversity of thinking and communication across language boundaries. However, learning to speak a foreign language is predictably challenging and non-native speakers offer a quasi-infinite source of pronunciation variation. This variation presents a challenge to AI-based technologies where we make assumptions about user behavior and expected runtime input to the models. Inconsistent usage therefore is difficult to model successfully. A study of pronunciation patterns from a set of non-native speakers uncovered some of the underlying dynamics of different levels of non-nativeness. Anchored in key differences in phoneme inventory and phonotactic distance between L1 and L2, the notion of a non-native canonical accent is introduced. This accent captures a significant portion of the speakers in the data and allows us to describe the variation with the precision needed to model it. If we can more consistently anticipate the characteristics of the variation, we're better positioned to handle it. This means that by explicitly focusing on the pronunciation patterns that are likely to be challenging to non-native speakers, we can adapt the technology to build more robust speech recognizers and more personalized foreign language learning experiences.

9:45

5aSC3. What does parity mean? A detailed comparison of ASR and human transcription errors. Courtney Mansfield, Sara Ng (Linguist., Univ. of Washington, Seattle, WA), Gina-Anne Levow (Dept. of Linguist., Univ. of Washington, Box 352425, Seattle, WA 98195-2425, levow@uw.edu), Mari Ostendorf (Elec. and Comput. Eng., Univ. of Washington, Seattle, WA), and Richard A. Wright (Linguist., Univ. of Washington, Seattle, WA)

Automatic speech recognition (ASR) has seen dramatic improvements as a result of advances in deep learning. This has led several recent studies to conclude that ASR is approaching parity with human performance, at least in certain speech contexts. These studies use average word error rate (WER) as their primary evaluation metric, dividing the total insertions + substitutions + deletions by the reference length ($\mathrm{WER} = (\mathrm{I} + \mathrm{D}+\mathrm{S})/\mathrm{N_r}$), to compare ASR systems to human transcribers. Averaging combined error types obscures important differences between human and ASR error patterns which impact human comprehension of the output. Human transcribers tend to

delete pragmatic and discourse markers (such as fillers and backchannels) and disfluencies, whereas ASR tends to make more substitution errors on words. In conversational settings listeners are able to recover from missing discourse markers, function words, or backchannels, but word substitutions are harder to recover from because they interrupt the information flow. WER is a reasonable first pass metric of ASR performance, but when it comes to communicative parity, it averages out important ways in which it differs from human transcription.

10:05–10:20 Break

10:20

5aSC4. Why ASR + NLP isn't enough for commercial language technology. Rachael Tatman (Developer Relations, Rasa Technologies, Guggenheim Hall, 3940-2425 Benton Ln., Seattle, WA 98195, rctatman@uw.edu)

With an increasing commercial demand for speech interfaces to be integrated into language technology, many technologists have made an unfortunate discovery: combining existing automatic speech recognition (ASR) and natural language processing (NLP) systems often leads to disappointing results. This talk will discuss two factors that contribute to this disparity and make some general suggestions for language technologists and researchers looking to work with tem. The first is the greater degree of variation in speech than text (at least in languages like English) which can lead to higher error rates overall. The second is a mismatch in domain. Modern machine learning approaches to language technology are very sensitive to differences between datasets and (due in part to the disciplinary division between researchers working on language technology for speech and text) most NLP applications have not been trained on speech data.

10:40

5aSC5. APhL aligner: A neural network forced-alignment system. Matthew C. Kelley (Linguist., University of AB, 3-24 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, mckelley@ualberta.ca), Scott J. Perry (Linguist., Univ. of AB, Edmonton, ON, Canada), and Benjamin V. Tucker (Linguist., Univ. of AB, Edmonton, AB, Canada)

Forced alignment is increasingly used in phonetics to automatically produce boundaries between words and phones. These boundaries can have significant errors and are often only placed at some predetermined time interval, like every 10 ms. We discuss some potential remedies to these difficulties and test them in a new neural network-based forced alignment system called the APhL Aligner, trained on the TIMIT and Buckeye speech corpora. In part, errors incurred during forced alignment can be attributed to the acoustic models that attempt to separate phones from each other. Even state-of-the-art neural network models struggle to acoustically separate phones. We examine the effect of relaxing the requirement to separate phones by instead training separate detectors for each phone class. Resolving the 10 ms interval difficulty requires a different approach. As with most aligners, we perform a Viterbi-style alignment to align windows of audio spaced at 10 ms to the phone string given by a pronunciation dictionary. We add an additional step, however, and use linear interpolation to determine an intermediate point after the 10 ms interval to place the boundary. We compare the results of these manipulations to the results of the Montreal Forced Aligner, custom-trained on the same data.

11:00

5aSC6. Speech and hearing for the next billion users. Malcolm Slaney (Machine Hearing Res., Google, 1600 Amphitheatre Ave., Mountain View, CA 94043, malcolm@ieee.org) and Richard F. Lyon (Machine Hearing Res., Google, Mountain View, CA)

Speech recognition has not withered in the years since John Pierce's 1969 editorial. Far from it, speech recognition is everywhere, and now works very reliably, even when speaking to a speech assistant in another room with music playing in the background. If the current speech technology is widely available to billions of users, then the next frontier is to provide the same technology to the next billion users (NBUs). In addition, all of us have limited capabilities to speak and understand the acoustic world around us, whether it rises to the level to be labeled a real disability, or arises due to fatigue or being distracted. The next frontier for speech and hearing is to accommodate all of our needs, disabled or not as they arise in the real world, for not just the first billion users, but the next billion, and the next billion after that. The available technologies include real-time speech recognition (and translation) for the hard of hearing, real-time speech enhancement, and auditory attention decoding from EEG signals.

11:20–11:40
Panel Discussion

Session 5aSP

Signal Processing in Acoustics: General Topics in Signal Processing II

Michael J. Bianco, Chair
Marine Physical Laboratory, Scripps Institution of Oceanography, University of California San Diego, 9500 Gilman Dr., MC 0238, La Jolla, CA 92037

Contributed Papers

8:00

5aSP1. Bayesian inference for boundary admittance estimation using a multipole model for room-acoustic simulation. Ziqi Chen (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu), Kirill V. Horoshenkov (Univ. of Sheffield, Sheffield, United Kingdom), and Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Acoustic surface admittance/impedance at room boundaries is essential for wave-based room-acoustic simulations. In this work, two levels of Bayesian inference are applied to estimate the surface admittance based on a multipole admittance model. This work estimates the order of the multipole admittance model through the high level of inference, Bayesian model selection. The first (low) level of inference, Bayesian parameter estimation, is applied to estimate the parameter values of the surface admittance model once model order is selected. This work approximates the frequency-dependent admittance from experimentally measured a set of acoustic surface admittance data. Analysis results demonstrate that multipole model-based Bayesian inference is well suited in estimating the frequency-dependent boundary condition within wave-based simulation framework. Numerical simulations verify the estimation results of Bayesian inference.

8:15

5aSP2. Real-time joint dereverberation and speech enhancement for hearing aid applications using edge devices. Gautam Shreedhar Bhat (The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, gxs160730@utdallas.edu), Nikhil Shankar, and Issa Panahi (The Univ. of Texas at Dallas, Richardson, TX)

In this work, we propose a joint dereverberation and speech enhancement technique for real-time applications. Speech dereverberation is based on the coherent-to-diffuse energy ratio (CDR) measured from the direction-of-arrival (DOA) dependent complex spatial coherence function. A personalized speech enhancement (SE) technique based on joint maximum a posteriori probability (JMAP) estimation that inherently accounts for the effects of changes in acoustic path and reverberation is used. The combination of these masking functions is used to enhance the speech corrupted by reverberation and noise in a frame-by-frame process. The proposed method is implemented on the smartphone (edge device), to illustrate real-time usability. The efficiency of the proposed method is evaluated using speech quality and intelligibility measures and compared with that of other benchmark techniques.

8:30

5aSP3. A low-cost digital delay line for bat biosonar research. Michael Roan (ME, Penn State, 1430 Linn St., State College, PA 16803, mjr110@psu.edu)

Bat biosonar research and bat neuroscience as a whole relies heavily on experiments to determine bat responses to various stimuli. Typically, these are presented to the bat using real world targets such prey insects, but also via artificial stimuli using an ultrasonic loudspeaker. One key component of artificial target creation is the use of delay lines. Analog delay lines have been used for decades to generate artificial targets at programmable ranges. More recently, digital delay lines have become the norm. This talk will present the details of the development of a very low-cost digital delay line that can produce bat echoes with up to three glints. The main delay between the bat and the target as well as the inter-glint spacing are programmable with delays as small as 5 μs possible. The platform used is a commonly available SAMD51 microcontroller with a total system cost under $50. The talk will present experimental results, code development using commonly available programming tools, and schematics for an add-on board that uses microphone or general differential inputs. Finally, the talk will introduce an approach implement wideband Doppler shift (time-scaling) of digitally delayed echoes using direct memory access tools native to the SAMD51 microcontroller.

8:45

5aSP4. Employing deep learning method to predict global head-related transfer functions from scanned head geometry. Yuxiang Wang (Elec. and Comput. Eng., Univ. of Rochester, 250 e Squire Dr., Apt. 7, Rochester, NY 14623, ywang310@ur.rochester.edu), You Zhang, Zhiyao Duan, and Mark Bocko (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

We propose an HRTF personalization method in which a Convolutional Neural Network (CNN) is employed to learn subjects' HRTFs from the scanned geometry of their heads. The trained model can then be employed to predict the global HRTF set (for all directions) from the subject's head scan data alone. In our trial, the HUTUBS HRTF database was used as the training set. A truncated spherical harmonic expansion of head scan data was used to preserve the boundary shape features that are important in the acoustic scattering process. Each HRTF for a given subject also was represented by a truncated spherical harmonic expansion. The SHT coefficients of the scanned head geometry and the HRTFs serve as training data for a CNN that subsequently can be used to predict HRTFs from geometric scan data. A leave-one-out validation with log-spectral distortion (LSD) metric was used for evaluation. The results show a decent LSD level at both spatial & temporal dimensions compared to the ground truth, and have lower LSD than the finite element acoustic simulation of HRTFs that the database provides. In continuing work, we are validating the prediction results in listener tests.

9:00

5aSP5. Room impulse response inference from the coherence properties of speech and music signals. 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)

Analysis of the coherence properties of the harmonic partials of speech and music signals recorded in an acoustic space provides information about the impulse response of the space. When an acoustic signal is filtered by a space, the autocorrelation of individual partials and the cross-correlation between pairs of partials provides information about the frequency-dependent reverberation time of the space. The relationship between the coherence time of the original, dry, signal and the reverberation time of the acoustic space determines the information that is retrievable in this "blind" approach. Recovering information about the acoustic space can be done within bounds, given only minimal assumptions or knowledge of the signal. Specifically, the original signal is assumed to contain a set of harmonically related partials, with mutually correlated amplitude and phase modulations and that the coherence time of the signal is known. These modulations are modelled as a narrow band pseudo random process. This is a reasonable assumption based on the physics of sound generation in classes of realistic sources, such as wind musical instruments or the human vocal tract.

9:15

5aSP6. Analysis of partially coherent acoustic wavefields from multichannel sound reproduction systems. 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)

This work analyzes the performance of multichannel sound reproduction systems using the theory of partially coherent wavefields to model the acoustic velocity vector at the listener's location. In this framework, the original signal is modelled as a narrowband, locally stationary random process, defined by a coherence time. The system model incorporates attenuation and delays from both positioning of the listener relative to the loudspeakers and any processing used in the reproduction system (e.g., panning). Given the finite coherence time of the signal, the magnitude and direction of the resultant velocity vector fluctuates in time. The location and spread of the virtual source that a human listener infers is determined by the distribution of velocity vector directions and a psychoacoustic averaging time associated with the precedence effect. Using this framework, we demonstrate the relative importance of three time constants in forming a stable image of the source location: the signal coherence time, the acoustic decay time, and the perceptual averaging time. This framework can be extended to incorporate early reflections in the listening environment and generalized to encompass natural sounds, such as speech or music, which can be modelled as a sum of narrowband components.

9:30–9:45 Break

9:45

5aSP7. Abstract withdrawn.

10:00

5aSP8. Enabling multi-input, multi-output measurements with a fluidic transducer. Benjamin Bühling (Dept. of Non-Destructive Testing, Bundesanstalt für Materialforschung und -prüfung (BAM), Unter den Eichen 87, Berlin 12205, Germany, benjamin.buehling@bam.de), Stefan Maack, and Christoph Strangfeld (Dept. of Non-Destructive Testing, Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany)

Ultrasonic testing is a widely used method for detecting damage and material characteristics in fields ranging from medicine to aerospace applications. The use of air-coupled ultrasound (ACU) enables an increase in testing speed, since no coupling medium needs to be applied to the transducer-specimen interface at each measurement point. The time needed for testing can be further decreased by using multiple-input, multiple output (MIMO) measurement systems. These systems require the ultrasonic signals to be uncorrelated so the that the individual pulses can be discriminated. Recently, the fluidic transducer was introduced, which generates acoustic pulses in the low ultrasonic frequency range. Experimentally, it is shown that each pulse contains a unique phase signature, that can be extracted by removing the signal envelope information. This technique is used to uniquely discriminate between simultaneously triggered pulses from two fluidic transducers and measure their individual times of flight.

10:15

5aSP9. Classification of respiratory diseases based on lung and cough sound using convolutional neural network. Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), Zanjabila Zanjabila, Puspita Y. Putri, and Jamiatul Firda (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Accurate non-invasive methods for detecting respiratory diseases, including COVID-19, are needed to suppress the potential of infection deployment to medical personnel when assessing the patient. In this report, we proposed respiratory diseases detection based on the cough and lung sounds of the patient. The recorded cough and lung sounds are processed for obtaining the mel frequency cepstral coefficient (MFCC) which is used as classification features. The features are then used for data training on deep learning models using convolutional neural networks (CNN). The result showed that the model can classify respiratory diseases with high accuracy compared with the prior studies.

10:30

5aSP10. Audio feature optimization approach towards speaker authentication in banking biometric system. Szymon Zaporowski (Multimedia Systems, Gdansk Univ. of Technol., Gdańsk, Poland), Andrzej Czyzewski (Multimedia Systems, Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Gdansk, Poland)

Experiments were carried out using algorithms such as Principal Component Analysis, Feature Importance, and Recursive Parameter Elimination dentifying most meaningful Mel-Frequency Cepstral Coefficients representing speech excerpts prepared for their classification are presented and discussed. The parameterization was made using Mel Frequency Cepstral Coefficients, Delta MFCC and Delta MFCC. In the next stage, feature vectors were passed to the input of individual algorithms utilized to reduce the size of the vector by previously mentioned algorithms. The vectors prepared in this way have been used for classifying vocalic segments employing Artificial Neural Network (ANN) and Support Vector Machine (SVM). The classification results using both classifiers and methods applied for reducing the number of parameters were presented. The results of the reduction are also shown explicitly, by indicating parameters proven to be significant and those rejected by particular algorithms. Factors influencing the obtained results were considered, such as difficulties associated with obtaining the data set, and ts labeling. The broader context of banking biometrics research carried-out and the results obtained in this domain were also discussed. [Project No. POIR.01.01.01-0092/19 entitled: "BIOPUAP—A biometric cloud authentication system" is currently financed by the Polish National Centre for Research and Development (NCBR) from the European Regional Development Fund.]

Session 5pAO

Acoustical Oceanography: Acoustic Sensing of Bio-Physical Interactions in the Water Column and Seafloor

Wu-Jung Lee, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Elizabeth Weidner, Cochair
University of New Hampshire, 24 Colovos Road, Durham, NH 03824

Anthony P. Lyons, Cochair
Center for Coastal and Ocean Mapping, University of New Hampshire, Durham, NH 03824

Chair's Introduction—1:00

Contributed Papers

1:05

5pAO1. Relating tensile strength to acoustic and biological properties of sediments along a mud-sand gradient in Mobile Bay, AL. Kelly M. Dorgan (Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528, kdorgan@disl.org), Grant Lockridge, Madeline R. Frey, W. Cyrus Clemo (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, Durham, NH), Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs.,Univ. of Texas, Austin, TX), Nina Stark, Nick Brilli (Virginia Tech, Blacksburg, VA), and Joseph Calantoni (Naval Res. Lab., Stennis Space Ctr., MS)

Nearshore and estuarine sediments experience varying sediment inputs that create sharp gradients in sediment properties over fairly small vertical and horizontal distances. In this study, we explored the sand-mud transition in sediment cores collected from Mobile Bay, AL, with acoustic and geotechnical approaches. A custom built instrument was used to measure depth profiles of tensile strength at cm-scale resolution. These measurements were compared to acoustic sound speed and attenuation across a range of frequencies. Cores collected exhibited a horizontal gradient from sand to mud across the study area with a sandy layer on top of muddier sediment in most cores. This study explicitly examines the relationship between acoustic sound speed and attenuation and strength-related properties of sediments such as cohesion that can be measured *in situ* more easily than textural properties (e.g., porosity and grain size). We compare our data to *in situ* normal incidence measurements and portable free fall penetrometer data as well as infaunal community composition to better relate acoustic, geotechnical, and biological properties across a gradient of sediments.

1:20

5pAO2. Biogeoacoustic sediment properties along a horizontal sand-mud gradient. Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, g.venegas@unh.edu), Madeline R. Frey (Dauphin Island Sea Lab, Dauphin Island, AL), Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs.,Univ. of Texas, Austin, TX), W. Cyrus Clemo, and Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL)

Whether deposited from the water column or generated by benthic infauna, interstitial organic matter is a ubiquitous constituent of marine sediment and is particularly prevalent in sediments with significant silt/clay fractions. Interstitial organic matter suspends silt and clay particles in the sediment matrix, adsorbs onto mineral surfaces, and resides between mineral contacts, all of which are hypothesized to alter geoacoustic properties. However, the extent to which interstitial organics alter geoacoustic properties is understudied and warrants further investigation. To address this, diver cores were collected along a horizontal sand-mud gradient in Mobile Bay, Alabama. The sediment was subsequently processed in the laboratory, where compressional wave speed (10 kHz to 1 MHz), compressional wave attenuation (100 kHz to 1 MHz), density, porosity, grain-size distribution, and organic carbon and nitrogen content were measured. In addition, physics-based geoacoustic models, such as VGS(λ) and mCREB, were fit to the wide range of sediment types. Measured geoacoustic properties and fit parameters inherent to each sediment acoustics model are compared with organic sediment properties. The extent to which organics control sediment geoacoustic properties and considerations to include organics in future modeling efforts will be discussed. [Work sponsored by ONR.]

1:35

5pAO3. Normal-incidence bottom loss measurements over a sand-mud gradient and comparison to direct seabed measurements. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, Durham, NH), W. Cyrus Clemo, Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Nick Brilli, Nina Stark (Dept. of Civil and Environ. Eng., Virginia Tech, Blacksburg, VA), and Joseph Calantoni (Ocean Sci. Div., U. S. Naval Res. Lab., Stennis Space Ctr., MS)

Seabed variability introduced by physical or biogenic processes contributes to multi-scale horizontal and vertical variability in seabed acoustic and geotechnical properties. An experiment was conducted near the mouth of Mobile Bay to investigate such spatial variability over a mud-sand gradient. A normal-incidence bottom loss survey was conducted on a 750×1500 m^2 grid with 50-m track-line spacing, in which a broadband acoustic source transmitted frequency-modulated chirps (1–100 kHz) every second, resulting in dense sampling of the seabed reflection coefficient over a wide frequency band. Diver cores were collected in the survey area for direct analysis of near-surface seabed properties. Cores were acoustically logged to obtain vertical profiles of sound speed and attenuation (10–1000 kHz) and then analyzed for physical properties (porosity, grain size, carbon content), infauna community composition, and sediment strength properties. Additionally, *in situ* portable free fall penetrometer measurements were conducted within the survey area to characterize geotechnical properties within

the upper 50 cm of the sediment. Spatial distributions of properties estimated from bottom loss measurements (e.g. porosity and sound speed) will be compared with vertical profiles from the cores and from the penetrometer to examine potential connections between seabed acoustic, geotechnical, and biological parameters. [Sponsored by ONR.]

1:50

5pAO4. Examination of a humpback whale subsurface foraging on patchy prey in Juan de Fuca Strait. Rhonda Reidy (Biology, UVic, 3800 Finnerty Rd., Victoria, BC V8P 5C2, Canada, rreidy@gmail.com), Stephane Gauthier (Fisheries and Oceans Canada, Inst. of Ocean Sci., Victoria, BC, Canada), Laura Cowen (Mathematics and Statistics, UVic, Victoria, BC, Canada), and Francis Juanes (Uvic, Victoria, BC, Canada)

Contemporary data for humpback whale prey in British Columbia, Canada, are biased toward prey types that require surface feeding, with no information on the whales' feeding behaviour at depth on alternative prey sources. Here, we present evidence that a humpback whale in Juan de Fuca Strait was subsurface feeding on juvenile pollock. We combined data from a multisensor, animal-borne tag with a vessel-mounted, Acoustic Zooplankton and Fish Profiler to describe the foraging effort of the whale relative to its prey. Analysis of a fecal sample from the tagged whale also revealed juvenile pollock. This work suggests a comprehensive sampling framework for the deeper foraging humpback whales in B.C. that are difficult to access for setting up quantitative data collection programs.

2:05

5pAO5. Can a 38 and 200 kHz broadband echosounder differentiate copepods, other zooplankton, and hydrographic sources of backscatter? Joseph Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, joe.warren@stonybrook.edu), Hannah Blair, Brandyn M. Lucca, Toniann Keiling, Rachel Carlowicz, and Monique Escalante (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY)

Zooplankton are a diverse group of organisms representing multiple taxonomic categories with very different acoustic scattering characteristics. While net and optical sampling methods can provide species (or higher)-level organism identification, they can not match the high spatial resolution and large sampling volume of active acoustic echosounders. A challenge in interpreting backscatter data though is when different types of animals co-occur, and exacerbating this problem is when the animals are concentrated (either actively or passively) through turbulent processes or variations in density and soundspeed which also scatter sound. Broadband backscatter data and net tows were collected in three different parts of the Northwest Atlantic (New York Bight, Cape Cod Bay, and Gulf of Maine) where copepods (and other small zooplankton) were abundant in the upper portion of the water column. The ability of a dual-frequency (38 and 200 kHz miniWBT) broadband echosounder to measure copepod abundance are discussed relative to: the use of additional frequencies, zooplankton abundance and species diversity, and physical oceanographic data.

2:20

5pAO6. Impacts of infaunal activity and physical disturbance on acoustic properties of muddy coastal sediments. W. Cyrus Clemo (Dauphin Island Sea Lab, 101 Bienville Blvd, Dauphin Island, AL 36528, wclemo@disl.org) and Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL)

Storm disturbance of shallow coastal sediments can resuspend subtidal sediment and transport and deposit sand from beach erosion, creating sorted layers of differing grain sizes. These sediments provide habitat for diverse and abundant infaunal organisms that mix these layers (bioturbation), but how the reestablishment of disturbance-tolerant infauna affects recently disturbed sediment structure, including acoustic properties, is poorly understood. In this laboratory study, we compared the effects of physical disturbance (resuspension of surface muddy sediment and addition of sand, simulating a storm), infaunal activities (burrowing by brittle stars), and the combination of both on acoustic properties of sediments. We hypothesized that acoustic properties would reflect faster reconsolidation and more mixing of the initial mud layer with the deposited sand in sediment with infauna compared to defaunated sediment. At several timepoints following disturbance (1–14 days), we measured sound speed and attenuation at 400 kHz at multiple depths within and below the resuspended layer. After 14 days, we measured porosity, grain size, and bioturbation with tracer particles to relate to acoustic properties. Infaunal effects on post-storm sediment structure may be important to consider for acoustic mapping of reconsolidating sediment in shallow coastal areas.

2:35

5pAO7. Measuring and modeling time-dependent changes in seabed scatter caused by near-bottom hydrodynamics and biologic processes. Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, g.venegas@unh.edu) and Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Acoustic backscatter systems (ABS) can sense time-dependent changes in seabed surface roughness, internal structure, and composition caused by near-bottom hydrodynamics and biologic processes. However, a better understanding of how these processes affect the acoustic backscatter is required before data from ABS can be effectively used as a remote sensing tool. A Simrad EK80 Wideband Acoustic Transceiver (WBAT) was deployed on the seafloor at two sites off the coast of New Hampshire, and long-term backscatter measurements were acquired at 38, 70, and 200 kHz. The first site was located at the mouth of the Piscataqua River, near New Castle Island (NC), and the second at Star Island (SI) within the Isle of Shoals. The NC site exhibited strong tidal currents and mixed sediment with burrowing infauna, while the SI site was more susceptible to storm events and contained a coarser sediment with large quantities of shell fragments. Measurements of ping-to-ping decorrelation of scattered acoustic pulses were used to quantify timescales involved with near-bottom currents and storm events, while finite-element techniques were employed to study the backscatter from idealized burrows located at the water-sediment interface. An overview of these techniques and results will be discussed. [Work supported by ONR.]

2:50

5pAO8. Ecosystem monitoring of a seagrass meadow using wideband acoustic measurements. Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu), Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Ctr. for Coastal & Ocean Mapping, Univ. of New Hampshire, Durham, NH), Abdullah F. Rahman (Sch. of Earth, Env., and Marine Sci., The Univ. of Texas Rio Grande Valley, Brownsville, TX), Matthew C. Zeh, and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Seagrasses are sentinel species whose sensitivity to changing water conditions makes it an indicator for sea level rise and climate change. The biological processes and physical characteristics associated with seagrass are known to affect acoustic propagation due to gas bodies contained within the seagrass tissue as well as photosynthesis-driven bubble production that results in free gas in the water. In this work, acoustical methods are applied to assess the health of seagrass meadows using a ray-based model that includes losses due to the dispersion, absorption and scattering of sound related to the gas bodies in the seagrass tissue and free bubbles in the water. The approach is applied to data collected in the Lower Laguna Madre where the seabed was covered by a dense growth of Thalassia testudinum. During the experiment, a piezoelectric source transmitted frequency-modulated chirps (0.1 to 100 kHz) over a four-day period. At the peak of photosynthesis-driven bubble production in the late afternoon, an additional decrease in the received level of more than 10 dB was observed. The volume fraction of gas present in the environment is estimated through acoustic modeling and related to seagrass biomass and photosynthesis. [Work supported by ARL IR&D and ONR.]

3:05

5pAO9. Observations of the variability of the New England shelf break by shipboard echosounders. Scott Loranger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 86 Water St., Woods Hole, MA 02543, sloranger@whoi.edu), Zhaozhong Zhuang (MIT-WHOI, Woods Hole, MA), J. Michael Jech (NEFSC, Woods Hole, MA), Weifeng G. Zhang, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The New England Shelf Break is a dynamic region where warm salty slope water meets with colder less saline shelf water to form a distinct front. The position and shape of the front is modified by a range of processes on differing temporal and spatial scales. While seasonal changes are well documented, the influence of shorter time scale processes throughout the water column are less well understood. Of interest is the influence of mesoscale physical oceanographic processes such as meanders in the Gulf Stream and warm core rings and eddies. From March to June 2021 five cruises to the New England Shelf Break were conducted as part of the ONR Task Force Oceans New England Shelf Break Acoustics Experiments. The shipboard broadband echo sounders on the R/V Neil Armstrong were used to ensonify the front during each of these five cruises. Significant changes were observed, especially during the transit of a warm core eddy through the study area. Groups of scatterers in the study area were targeted by DeepSee (a broadband tow sled) and a broadband echo sounder equipped REMUS 600 to enable ensonification of individual targets to aide in acoustic identification of biology at the shelf break front.

FRIDAY AFTERNOON, 3 DECEMBER 2021 — 401 (L)/404 (O), 1:00 P.M. TO 4:00 P.M.

Session 5pBA

Biomedical Acoustics: Acoustics for Kidney Stone Management II

Michael R. Bailey, Cochair
University of Washington, 1013 NE Boat St., Seattle, WA 98105

Adam Maxwell, Cochair
Department of Urology, University of Washington, Seattle, WA 98195

Contributed Papers

1:00

5pBA1. Preliminary report on the functional changes associated with burst wave lithotripsy treated pig kidneys. Bret A. Connors (Anatomy, Indiana Univ. School of Medicine, 635 Barnhill Dr., Medical Sci. Bldg., Rm. 0051, Indianapolis, IN 46202, bconnors@iupui.edu), Tony Gardner (Anatomy, Indiana Univ. School of Medicine, Indianapolis, IN), Ziyue Liu (Biostatistics, Indiana Univ. School of Medicine, Indianapolis, IN), James E. Lingeman (Urology, Indiana Univ. School of Medicine, Indianapolis, IN), and James C. Williams (Anatomy, Indiana Univ. School of Medicine, Indianapolis, IN)

Burst Wave Lithotripsy (BWL) is a new stone treatment that utilizes ultrasound to fracture renal calculi. However, it is not known if BWL alters renal function, so we tested if a clinical dose of BWL impacts glomerular filtration rate (GFR) or kidney blood flow. Treatment consisted of placing each pig (32–42 kg) on its side, locating the lower pole of the right kidney, and treating that kidney with a BWL dose of 18,000 ultrasound pulses at 10 Hz, 20 cycles/pulse and a peak negative pressure of −7 MPa (n = 6). Six additional pigs were used as sham controls. Inulin and para-aminohippuric acid (PAH) were infused into the pigs with blood and urine samples being collected both before and 1-h after BWL treatment. The concentrations of inulin and PAH were determined in the samples and used to calculate GFR and effective renal plasma flow (eRPF). For the BWL treated group, GFR and eRPF did not change after BWL exposure ($-6.7 \pm 7.2\%$ for GFR, $-10.9 \pm 5.5\%$ for eRPF), and this response was like the sham control group ($P = 0.23$ for GFR, $P = 0.08$ for eRPF). A typical clinical dose of BWL caused no change in renal function. [Work supported by NIH, Grant P01 DK43881.]

1:15

5pBA2. Optimizing ultrasound frequency in burst wave lithotripsy. Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russia, olegs2@uw.edu) and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Unlike shock wave lithotripsy, burst wave lithotripsy (BWL) uses tone bursts, consisting of many periods of sine wave. The fragmentation threshold of a stone depends on the burst frequency or, for a given frequency, on the stone size. In this work, an analytical theoretical approach to modeling mechanical stresses in a spherical stone has been developed. The theory is based on the method previously used to study the acoustic radiation force acting on an elastic stone [Sapozhnikov and Bailey, *J. Acoust. Soc. Am.* (2013)]. It has been shown that at low frequencies, when the wavelength is much greater than the diameter of the stone, the maximum principal stress is approximately equal to the pressure amplitude of the incident wave. With increasing frequency, when the diameter of the stone begins to exceed about half the wavelength in a surrounding liquid (the exact condition depends on the material of the stone), the maximum stress increases and can reach five times the pressure amplitude or even more. This observation suggests adjusting the BWL frequency depending on the stone diameter in order to comminute it into passable fragments. [Work supported by NIH, Nos. P01-DK043881 and RBBR 20-02-00139.]

1:30

5pBA3. Breaking urinary stones to small size with burst wave lithotripsy. Shivani Ramesh (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, rameshsh@uw.edu), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Adam Maxwell (Urology, Univ. of Washington, Seattle, WA), Ga Won Kim (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), James C. Williams (Dept. of Anatomy, Cell Biology, and Physiol., Indiana Univ. School of Medicine, Indianapolis, IN), Ziyue Liu (Dept. of Biostatistics, Indiana Univ. School of Medicine, Indianapolis, IN), Tim Colonius, Shunxiang Cao (Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Ekaterina Kuznetsova, Jeff Thiel, Barbrina Dunmire, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Mathew D. Sorensen (Urology, Univ. of Washington, Seattle, WA), James E. Lingeman (Dept. of Urology, IU Health, Indianapolis, IN), and Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

It is hypothesized a 2.6-mm stone was too small to break with 390-kHz burst wave lithotripsy (BWL). In clinical trial NCT03873259, a 2.6-mm stone failed to break after 10 min of 390-kHz BWL and was removed intact. *Ex vivo*, the stone failed to break after another 30 min at 390 kHz but broke into four pieces in 4 min at 650 kHz. A linear elastic model was used to calculate the stress created inside stones of different sizes, shapes, and compositions by shock wave lithotripsy (SWL) and different BWL frequencies. The model predicts above a threshold frequency proportionate to wave speed over stone length, the maximum principal stress inside a stone increases to more than five times the acoustic pressure applied. Thus, smaller stones may fragment at higher but not lower frequency. Amplification remains with irregularly shaped stones but is not seen with an SWL waveform. *Ex vivo*, stones smaller than 3 mm broke fastest at 830 kHz while larger stones broke fastest with 390 kHz followed by 830 kHz. For small stones and fragments, increasing BWL frequency many produce, amplified stress in the stone causing the stone to break to smaller fragments to pass. [Work supported by NIH-P01-DK04331 and NIH-K01-DK104854.]

1:45

5pBA4. A burst wave lithotripsy system for urinary stones in pet cats. Adam Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA 98195, amax38@u.washington.edu), Ga Won Kim, Elizabeth Lynch, Brian MacConaghy (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Eva Furrow, Jody Lulich (Dept. of Veterinary Clinical Sci., Univ. of Minnesota, St. Paul, MN), Michael Borofsky (Dept. of Urology, Univ. of Minnesota, Minneapolis, MN), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Upper urinary tract stones are becoming more prevalent in cats. Surgery to address obstructing stones has poor outcomes and such stones can rarely be treated with medical therapy. Burst wave lithotripsy (BWL) is a non-invasive, ultrasound-guided, handheld focused ultrasound technology to disintegrate urinary stones, which is now in human clinical trials. In this project, we created a scaled version of the system for use on pet cats, adjusting geometric parameters and frequency of the transducer to account for differences in anatomic scale and stone size. A 650-kHz prototype transducer was fabricated, calibrated, and tested on 25 natural feline stones 2–5 mm applying 20-cycle pulses at 10 Hz pulse repetition rate with three different focal pressure amplitudes $\leq$8.9 MPa in an *in vitro* tissue phantom. The fragments were sieved in 10-min intervals to assess breakage. The results demonstrated that between 73%–96% of the stone mass was reduced to fragments i$<$1 mm in a 30-min exposure, with only 2/25 stones showing no visible fragmentation. These data suggest BWL can fragment feline ureteral stones and that a high-frequency system can produce small fragments more likely to pass through the small ureter of a cat. [Work supported by NIDDK P01 DK043881.]

2:00

5pBA5. First fragmentation of stones in humans by burst wave lithotripsy. Jonathan D. Harper (Urology, Univ. of Washington, Seattle, WA), James E. Lingeman (IU Health, Indianapolis, IN), Ian Metzler (Oregon Health and Sci. Univ., Portland, OR), Robert M. Sweet (Urology, Univ. of Washington, Seattle, WA), James C. Williams (Indiana Univ. School of Medicine, Indianapolis, IN), Adam Maxwell (Urology, Univ. of Washington, Seattle, WA), Jeff Thiel, Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Barbrina Dunmire (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA, mrbean@uw.edu), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Mathew D. Sorensen (Urology, Univ. of Washington, Seattle, WA)

We report stone comminution in the first 18 human subjects by burst wave lithotripsy (BWL). Subjects undergoing clinical ureteroscopy (URS) for at least one stone $\leq$12 mm on computed tomography (CT) are recruited. During the planned URS, either before or after ureteroscope insertion, BWL is administered with a handheld probe for 10 min, and any stone fragmentation and tissue injury are observed. The primary effectiveness outcome is the volume percent comminution of the stone into fragments $\leq$2 mm, where fragment volume is determined by μCT of basketed fragments or image processing of the URS video using the laser fiber as a size reference. The primary safety outcome is independent, blinded grading of tissue injury from the URS video. Overall, 10 of 22 stones (45%) fragmented completely in 10 min. 64% and 73% were projected to fragment completely in 15 and 40 min, respectively. For reference, shock wave lithotripsy success is about 65% to fragments $\leq$4 mm in 40 min. Of the other six stones, one was larger than the beamwidth, two were smaller than the wavelength, and the ureteroscope introduced air bubbles around another. Only mild reddening with some hematuria was observed ureteroscopically. [Work supported by NIH-P01-DK04331.]

2:15

5pBA6. Flexible ultrasound-based system for clinical trial of burst wave lithotripsy and pushing of kidney stones. Ekaterina Kuznetsova (Appl. Phys. Lab., Univ. of Washington, Seattle, WA, katykuz@uw.edu), Michael R. Bailey (Univ. of Washington, Seattle, WA), Bryan W. Cunitz, Barbrina Dunmire, Ga Won Kim, Elizabeth Lynch (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Adam Maxwell (Urology, Univ. of Washington, Seattle, WA)

Fragmentation of kidney stones by burst-wave lithotripsy (BWL) has shown promising results in preclinical and clinical trials. Based on initial findings, the system was upgraded to enhance imaging, target stones at deeper depths, fragment small stones, and potentially conduct dusting of stones. A 64-element, single-crystal, phased-array imaging probe is coaxially aligned with the therapy probe for image guidance and therapy feedback. Imaging is controlled through a Verasonics Vantage research ultrasound engine capable of harmonic imaging to enhance stone resolution and contrast. These features improve targeting and endpoint detection, particularly for small stones and fragments. New therapy probes were added to effectively target stones with greater skin-to-stone distance, including a higher (800 kHz) frequency transducer to effectively break $<$4 mm stones into sub-millimeter fragments to facilitate passage. The amplifier was upgraded to a custom class D/E design with increased power required by the therapy transducers and is capable of ultrasonic propulsion and real time monitoring of electrical power. This system will provide capabilities to treat a larger patient population as we begin trials breaking and expelling stones in the clinic setting. [Work supported by NIH-P01-DK043881.]

2:30–3:00 Break

3:00

5pBA7. Ultrasound to facilitate passage of distal ureteral stones. M. K. Hall (Dept. of Emergency Medicine, Univ. of Washington School of Medicine, Seattle, WA 95105, mkhall@uw.edu), Jeff Thiel (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Patrick C. Samson (Dept. of Urology, Weill Cornell Medical College, New York, NY), Ross Kessler (Dept. of Emergency Medicine, Univ. of Washington School of Medicine, Seattle, WA), Peter Sunaryo, Robert M. Sweet, Ian Metzler (Dept. of Urology, Oregon Health and Sci. Univ., Portland, OR), Layla Anderson (Dept. of Emergency Medicine, Univ. of Washington School of Medicine, Seattle, WA), Barbrina Dunmire, Christina Popchoi, Ravi Managuli, Bryan W. Cunitz, Barbara Burke (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Lisa Ding, Brianna L. Gutierrez (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Ziyue Liu (Dept. of Biostatistics, Indiana Univ. School of Medicine, Indianapolis, IN), Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Hunter Wessells, and Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Feasibility of ultrasonic propulsion and burst wave lithotripsy (BWL) to noninvasively reposition distal ureteral stones to facilitate passage and relieve pain was tested. Patients presenting to the clinic or emergency department were recruited. Thirteen subjects underwent ultrasonic propulsion (lower amplitude, longer duration pulses) alone, and 10 subjects also received intermittent BWL (higher amplitude, shorter duration pulses). All participants were awake and underwent a pain assessment pre- and post-procedure. For analysis, subjects were sub-categorized based on whether their stone was acute or chronic (present for $\leq$10 days or $>$10 days from their initial ED or symptomatic presentation, respectively). Seventeen subjects were enrolled in the acute study population and 6 subjects were enrolled in the chronic population. Overall, 94% of acute stones passed in an average of 3.4 days post-procedure relative to 54% in 7.5 days in the American Urological Association guidelines. 67% of chronic cases passed stones or fragments, and two surgeries were cancelled. Pain reduction was statistically significant ($p = 0.0215$). Adverse events were limited to hematuria on initial urination post-procedure (BWL only, $n = 3$) and a mild sensation, akin to a pinprick, associated with fewer than 10 of 620 propulsion bursts ($n = 3$). [Work supported by NIH-P01-DK04331.]

3:15

5pBA8. Acoustic radiation force on objects of arbitrary shape and composition. Blake E. Simon (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu) and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic radiation force exerted on an object of arbitrary shape and composition by an arbitrary incident sound beam can be calculated using spherical harmonic expansions of the pressure field. The coefficients in the expansions of the incident and scattered fields, when substituted into existing models, determine the radiation force on the object. Analytical expressions for the coefficients of both the incident and scattered fields are available for very few cases, such as a plane wave incident on a rigid sphere. In the present work, finite element modeling is used to calculate the coefficients, and resulting radiation force, for a variety of incident fields and object geometries. Different compositions of the objects are also considered. Validation of the approach is demonstrated by comparison with semi-analytical results available for the radiation force exerted by progressive and standing plane waves scattered by compressible spheroids in different orientations with respect to the incident field. Motivation for this approach is development of a model that can be used to compare with direct measurements of radiation force on irregularly shaped objects of different compositions. [BES is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

3:30

5pBA9. Acoustic manipulation of objects *in vivo*. Mohamed A. Ghanem (Appl. Phys. Lab., Univ. of Washington, 1013 NE Boat St., Seattle, WA 98105, mghanem@uw.edu), Adam Maxwell (Urology, Univ. of Washington, Seattle, WA), Yak-Nam Wang, Bryan W. Cunitz, Vera A. Khokhlova, Oleg A. Sapozhnikov, and Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Acoustic radiation forces can trap and manipulate objects in three-dimensional space (3D). The state of the art in acoustic trapping has demonstrated the ability to move small or lightweight objects using single or multiple transducers. For certain medical applications, acoustic manipulation can be utilized to control foreign objects in the body, such as kidney stones. However, difficulties from these applications include the ability to control large solid objects, transmitting robust acoustic beams through the skin, and the use of a single source due to limited acoustic window in the body. The goal of this work was to use a 1.5 MHz, focused, multi-element array to synthesize specific acoustic beams to trap, levitate, and steer kidney stone models (glass spheres) in a water bath and urinary bladders of live pigs. Stable acoustic traps were investigated numerically and experimentally in water bath to design a protocol for the 3D manipulation strategies used *in vivo*. Three programmed path were used to levitate and manipulate glass spheres in live pigs. The motion was recorded using a camera and a synchronized ultrasound imaging probe. Deviation from the intended paths was on average $<$10%. [Work supported by NIH P01-DK043881, K01-DK104854, R01EB7643, and RFBR 20-02-00139.]

3:45

5pBA10. Randomized control trial of ultrasonic propulsion to facilitate clearance of residual kidney stone fragments. Mathew D. Sorensen (Univ. of Washington and VA Puget Sound, Seattle, WA), Jessica C. Dai (UT Southwestern Medical Ctr., Dallas, TX), Tony T. Chen, Peter Sunaryo (Dept. of Urology, Univ. of Washington, Seattle, WA), Barbrina Dunmire, Jeff Thiel (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Michael Porter, Branda Levchak (Univ. of Washington and VA Puget Sound, Seattle, WA), Barbara Burke, Christina Popchoi, Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Ziyue Liu (Dept. of Biostatistics, Indiana Univ., Indianapolis, IN), Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), and Jonathan D. Harper (Dept. of Urology, Univ. of Washington, Seattle, WA)

The goal is to test effectiveness and safety of transcutaneous ultrasound pulses for facilitating clearance of residual urinary stone fragments in a randomized control trial (RCT). The study is conducted in a clinic setting with the control and treatment arms undergoing all the same activities with the exception of the ultrasonic propulsion procedure and associated pain questionnaire. Subjects are followed 90 days for assessment of adverse events and for up to 5 years for stone growth and symptomatic stone visits. Twenty-six of 33 subjects have been recruited in each arm The most recent treatment is reported. The subject received shock wave lithotripsy and did not pass any fragments for 17 months. The subject was randomized to the treatment arm. Fragments began moving with the third 3-s propulsion pulse at output 2 of 5 levels. Fifty-two percent of propulsion pulses (41 of 79) resulted in fragment movement. The subject reported no pain at the beginning or end of the procedure and telephoned within 2 h to report passing 9 fragments, no hematuria, and no adverse events. The treatment results of this RCT are compelling as an immediate and casual effect was measured. [Work supported by NIH-P01-DK043881 and VA Puget Sound resources.]

Session 5pSC

Speech Communication, Psychological and Physiological Acoustics and Signal Processing in Acoustics: Large-Scale and Remote-Platform Acoustic Analysis (Poster Session)

Laura Dilley, Chair
Communicative Sciences and Disorders, Michigan State University, 1026 Red Cedar Road, East Lansing, MI 48824

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

5pSC1. Processing methods for the detection of landmark acoustic cues. Belinda Shi (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, beeshi@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

Landmarks (Stevens, 2002) are acoustic cues that are correlated with certain changes in speech articulation, and can be used to infer some of the distinctive features useful for speech recognition, largely the manner features. This project identifies and organizes the processing steps involved in extracting eight types of Landmark acoustic cues for a feature-based hierarchical automatic speech recognition system, in which each module detects one landmark cue from samples of continuous speech, based on knowledge about human speech production and acoustic measurements such as formant frequency and energy. It maps out the processing steps necessary to accurately detect speech Landmarks, and describes a standardized system of modules implemented in Matlab that can accurately identify each of the Landmark cues in speech files quickly and efficiently. The processing methods laid out in this project provide a framework that can be later extended to find other acoustic cues related to place and voicing.

5pSC2. Multilingual speech annotation of landmarks and other acoustic cues to distinctive features. Antony Hernandez Mendoza (MIT, 97 Bay State Rd., Boston, MA 02215, tonyhm83@gmail.com), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

A speech annotation system developed for English that identifies landmarks and other acoustic cues to distinctive features (Stevens, 2002; Huilgol *et al.*, 2019) has been extended to Spanish and Korean. This process includes retrieving the (allo)phone sequence for the words of a target utterance from a standard lexicon, converting this sequence into an underlying phoneme sequence, and then generating citation-form landmark and other acoustic cue sequences from the phoneme sequence and its features. This standard set is presented to labelers who examine the waveform of the target utterance and mark which of the predicted cues have been realized, or deleted, and where unpredicted cues have been inserted. For each language, we draw from a superset of acoustic cues that function in all three languages, but each language also makes use of an additional set of acoustic cues specific to that language.

5pSC3. Using tone and break indices labelling to document patterns of prosodic prominence and their acoustic effects in a database of spoken Italian sentences. Alec DeCaprio (Radcliffe Inst. for Adv. Study at Harvard Univ., Cambridge, MA, alecdecaprio@college.harvard.edu), Javier Arango (Radcliffe Inst. for Adv. Study at Harvard Univ., Cambridge, MA), Ian Chan (Harvard Univ., Markham, ON, Canada), Maria-Gabriella Di Benedetto (DIET, Sapienza Univ. of Rome, Rome, Italy), Luca De Nardis (Sapienza Univ. of Rome, Rome, Italy), and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

This study examined how tonal prominence impacts the acoustic cues to features of Italian vowels. It was one aspect of an analysis of the LaMIT (Lexical Access Model for Italian) project [Di Benedetto *et al.*, "Speech recognition of spoken Italian based on detection of landmarks and other acoustic cues to distinctive features," in 179th ASA Meeting, Chicago, IL, 8–12 December (2020)]. We strove to understand and systematize the quantitative changes to vowel quality provoked by prosodic and lexical stress, using the LaMIT corpus (100 Italian sentences recited twice by four speakers). Prominent vowels were classified with the TOBI (Tones and Break Indices) labelling method. Accents deemed tonally prominent, known as "pitch accents," were starred and given a label that reflected their F0 contours. Labels were limited to those put forth by Avesani ("ToBIt: Un sistema di trascrizione per l'intonazione italiana," in Atti 5e Giornate di Studio del Gruppo di Fonetica Sperimentale, Povo, Italy, pp. 85–98) as constitutive of the prosody of standard Italian. Labels also marked silences, elisions, and spaces between words. Results will clarify if prominence in Italian induces enhanced vowel sonority, or instead a "hyperarticulation" of distinctive segments, contributing to an ongoing debate on the subject.

5pSC4. On the sensitivity of acoustic distance measures to different parameterizations of mel-frequency cepstral coefficients and temporal alignment algorithms. Charles H. Redmon (Univ. of Oxford, 427 Blake Hall, University of Kansas, Lawrence, KS 66045, redmon@ku.edu)

The increasing role of large speech databases in phonetic and psycholinguistic research has placed a premium on the efficiency and objectivity with which continuous acoustic signals are encoded in discrete parameters that may then be used in recognition models. One simple and common way speech has been encoded and compared in this work is by (1) converting the acoustic signal into a matrix of MFCCs, and (2) measuring the distance between two or more matrices following the application of a temporal alignment algorithm (e.g., dynamic time warping, cross-correlation) to account for differences in utterance rate. The specification of both the MFCCs and the alignment algorithm requires control over a wide array of variables, many of which have both theoretical and experimental implications, and yet most of these variables are set to default values specified in toolkits like the HTK and the Phonological CorpusTools software (Young *et al.*, 2002; Hall *et al.*, 2017). This project uses a large multispeaker database of nonword

syllable production and perception (Woods *et al*., 2010) to test the performance of this acoustic distance procedure over the full parameter space, and evaluate the degree of accordance of such measurements with phonological theory and perceptual data.

5pSC5. Automated detection of Glottal-related acoustic cues for feature-cue-based analysis. Sagnik Anupam (MIT, 351 Massachusetts Ave., Alpha Delta Phi Fraternity, Cambridge, MA 02139, sanupam@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

Detecting and interpreting individual acoustic cues to identify features that distinguish among speech sounds is thought to play a key role in automatic speech processing, modeling human speech perception, detecting and diagnosing speech disabilities (and tracking the effects of treatment for those disabilities), and studying cross-language differences. The individual acoustic cues form a bridge between transcription using phonemic symbols (phonemes) and raw acoustic measurements from the signal and allow the analysis to capture systematic language-related differences in how words are pronounced in different contexts–differences that are critical to speech recognition. Within this framework, analysis of glottal-related acoustic cues, such as aspiration and voicing, can contribute to the identification of phonemes that exhibit these acoustic cues in their respective productions. In this study, we propose an algorithm that uses time-series data processing methods and machine learning models to accurately predict the presence of the glottal cue labels from waveform analysis. We further highlight the accuracy of our model and cue-based labeling method in identifying speech modifications over phone and phoneme-based methods.

5pSC6. Speaker tracking across a massive naturalistic audio corpus: Apollo-11. Meena Chandra Shekar (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, 7220 McCallum Blvd, Apt. 1704, Dallas, TX 75252, Meena.ChandraShekar@utdallas.edu) and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

Apollo-11 was the first manned space mission to successfully bring astronauts to the moon. More than +400 mission specialists/support team members were involved whose voice communications were captured using the SoundScriber multi-channel analog system. To ensure mission success, it was necessary for teams to engage, communicate, learn, address and solve problems in a timely manner. Hence, in order to identify each speaker's role during Apollo missions and analyze group communication, we need to automatically tag and track speakers individually since manual annotation is costly and time consuming on a massive audio corpus. In this study, we focus on a subset of 100 h derived from the 10 000 h of the Fearless Steps Apollo-11 audio data. We use the concept of "Where's Waldo" to identify all instances of our speakers-of-interest: (i) Three Astronauts; (ii) Flight Director; and (iii) Capsule Communicator. Analyzing the handful of speakers present in the small audio dataset of 100 h can be extended to the complete Apollo mission. This analysis provides an opportunity to recognize team communications, group dynamics, and human engagement/psychology. Identifying these personnel can help pay tribute to the hundreds of notable engineers and scientists who made this scientific accomplishment possible. Sponsored by NSF #2016725

5pSC7. A strategy for correcting errors in automated formant extraction. Lisa M. Johnson (Linguist., Brigham Young Univ., 4064 JFSB, Provo, UT 84602, lisajohnson@outlook.com)

Sociophonetic vowel analysis relies heavily on measurements of resonant frequencies, particularly of the first and second formants. Automated formant estimation using linear predictive coding (LPC) algorithms in software like Praat greatly increases efficiency compared to hand measurements and allows researchers to analyze more data than was possible before this technological advancement. However, LPC analysis is prone to certain types of errors (e.g., Di Paolo *et al*., 2011; Harrison, 2013; Labov *et al*., 2006; Strelluf, 2019; Styler, 2017). In one common error, which I call "faulty low F2" (FLF2), LPC identifies a spectral peak between F1 and F2 as an F2 measurement. In automated extraction, the real F2 is then recorded as F3. Manually correcting these errors is time-consuming, potentially erasing the gains made by automated extraction. However, a systematic error suggests the possibility of a systematic solution. This presentation describes a method for identifying and correcting FLF2 errors using a script in R. The script identifies possible errors based on expected ranges and trajectories, determines whether the recorded F3 at the measurement point meets the criteria for a legitimate F2, and makes substitutions where appropriate. Results of tests with real data are presented. [Work supported by NSF #XXXXXXX.]

5pSC8. The effects of audio compression on voice quality measurements. Jailyn M. Pena (Linguist., New York Univ., 21-15 35th St. 1A, Astoria, NY 11105, jmp987@nyu.edu), Alicia Mason (Dept. of Linguist., New York Univ., New York, NY), and Lisa Davidson (Linguist., New York Univ., New York, NY)

This study examines audio compression effects on HNR <3500Hz, cepstral peak prominence (CPP), F0, H1*-H2*, H1*-A2*, and H4*-2K between uncompressed .wav files and the same files compressed to .mp3 and reconverted to .wav. Modal, breathy, and creaky vowels in Mazatec and !Xóõ were analyzed to examine compression effects both cross-linguistically and across phonation types. Statistical results showed no main effect of compression on HNR or CPP for Mazatec, but compression lowered both measures for !Xóõ. Interactions further revealed that compression lowered HNR for breathy vowels, but raised HNR for creaky vowels in both languages, while HNR and CPP for modal !Xóõ vowels decreased. F0 was lowered for breathy vowels, but raised for modal and creaky vowels in Mazatec. Conversely, compression raised F0 for breathy vowels and lowered it for creaky vowels in !Xóõ. For spectral measures, compression lowered H1*-H2* and H1*-A2* for breathy and modal !Xóõ vowels but raised these measures for creaky vowels. Compression also raised H4*-H2K* for breathy !Xóõ vowels and lowered it for creaky vowels. These results indicate that compression affects common voice quality measures unequally both cross-linguistically and between phonation types within languages, making the use of compressed and reconverted sound files risky for acoustic analysis.

5pSC9. High quality recordings and transcriptions of speech via remote platforms. Jonathan Wright (Linguistic Data Consortium, Univ. of Pennsylvania, 3600 Market St., Ste. 810, Philadelphia, PA 19104, jdwright@ldc.upenn.edu), Robert Parker, Jeremy Zehr, Neville Ryant, Mark Liberman, Christopher Cieri, and James Fiumara (Linguist. Data Consortium, Univ. of Pennsylvania, Philadelphia, PA)

High quality recordings and transcriptions of speech are important to a wide variety of disciplines from linguistics and human language technology to biomedical screening, diagnosis and tracking. The wide availability of internet connections and powerful mobile devices offers low cost opportunities for collecting speech data at scale. But even in the era of open source software, numerous challenges remain. For example, video call applications such as Zoom have become widespread and allow for recording, but only provide lossy codecs with low frame rates, and often contain missing, repeated, or interpolated frames, as well as freezes, longer dropouts and other audio artefacts. LDC has recently developed a suite of tools to allow high quality internet-based audio recordings and transcription, with a premium on portability and flexibility, using secure cloud computing services for storage and back-end processing. We present here the current design and capabilities of our software, as well as availability in terms of open source code and app distribution. We also discuss future plans; planned capabilities include 2+ sided conversational recordings, connecting participants via the internet, a modern extension of Conversational Telephone Speech (CTS) collections.

5pSC10. SpeechCollectR: An R package for web-based speech data collection. Abbey L. Thomas (The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, abbey.thomas@utdallas.edu) and Peter F. Assmann (The Univ. of Texas at Dallas, Richardson, TX)

To accommodate web-based speech production and perception data collection during the COVID-19 pandemic, we developed the R package, SpeechCollectR (https://github.com/abbey-thomas/speechcollectr), designed for speech scientists familiar with R [R Core Team, *R: A Language and Environment for Statistical Computing* (R Foundation for Statistical Computing, Vienna, 2021]. SpeechCollectR employs existing R functions for acoustic analysis, allowing researchers to collect, visualize, and analyze speech production and perception data with a single program. SpeechCollectR lets researchers construct interfaces that embed experimental tasks in games to sustain participant attention amid distractions in non-laboratory environments. The application for production experiments saves uncompressed audio data as WAV files, one recording per trial. Amplitude envelope features are used to isolate a speech token in each recording. This token is compared to the remainder of the recording to assess signal-to-noise ratio. Performing this analysis after each trial, we can supply the participant with feedback to improve recording quality. We tested the efficacy of using SpeechCollectR, collecting 1344 recordings of three types of affective prosody from 32 participants. 83% of these recordings were usable for analysis of prosodic variables. We present a bootstrap analysis demonstrating reliability of affective prosodic features despite variations in recording equipment and environment.

5pSC11. Performing forced alignment with Wav2vec 2.0. Jian Zhu (Linguist., Univ. of Michigan, 611 Tappan Ave., Ann Arbor, MI 48109, lingjzhu@umich.edu) and Cong Zhang (Radboud Univ., Nijmegen, The Netherlands)

Forced alignment, the task of aligning segmentation of audio speech files with an orthographic or phonetic transcript, is fundamental to many types of speech research. Yet the available toolkits for forced alignment are mostly based on the classic HMM/GMM systems, which are outperformed by neural network-based speech recognition models, especially the large-scale speech pre-trained models in recent years. We propose a method of forced alignment utilizing the pre-trained transformer-based model, Wav2vec 2.0. This model has been pre-trained on massive audio datasets, and is subsequently fine-tuned on 360 h of speech data guided by connectionist temporal classification (CTC) loss. The model has learned to jointly recognize phonemes and perform segmentation with or without orthographic transcriptions. During inference, the hidden states are converted to frame-level alignments through post-processing. Our preliminary analysis shows that the model performs competitively when segmenting the TIMIT benchmark, even without orthographic transcription provided.

5pSC12. Exploring the variable efficacy of Google speech-to-text with spontaneous bilingual speech in Cantonese and English. Nikolai Schwarz (Linguist., Univ. of BC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, schwarznikolai@yahoo.com), Khia A. Johnson, and Molly Babel (Linguist., Univ. of BC, Vancouver, BC, Canada)

With the growth of Automatic Speech Recognition (ASR) and voice user interface software, it is important to test for efficacy across different language varieties and identify sources of bias. Recent work assessing ASR efficacy and bias implicates factors like race, gender, dialect, and age as leading to different efficacy rates. Multilingualism presents another source of variation that ASR systems must grapple with, ranging from code-switching to phonetic variation both within and across speakers. Thus, variable ASR performance is likely exacerbated for multilingual communities. Using a spontaneous Cantonese-English bilingual speech corpus (Johnson, 2021), this study tests the efficacy of Google Speech-To-Text (STT) language models (Canadian English and Hong Kong Cantonese) with a heterogeneous bilingual speech community. Efficacy is assessed via fuzzy string matching between the STT and the manually corrected transcripts. STT performance is variable but overall better for English. Confidence ratings and matching scores are evaluated alongside listener ratings of perceived accentedness and various demographic groupings. The results of this study will provide practical guidance for using STT in the context of speech production research pipelines while highlighting its drawbacks concerning bias in a relatively understudied, multilingual group.

5pSC13. Automated accent rating using deep neural networks. Tyler T. Schnoor (Linguist., Univ. of AB, 4-32 Assiniboia Hall, University of AB, Edmonton, AB T6G2E7, Canada, tschnoor@ualberta.ca), Matthew C. Kelley, and Benjamin V. Tucker (Linguist., Univ. of AB, Edmonton, AB, Canada)

Automated accentedness rating has the potential to improve many human-computer interactions involving speech, including the adaptation of automatic speech recognition or other artificial intelligence models to the speaker's accent. Accent ratings may also be used as a metric by which language learners can quantify their progress. This study employs bidirectional long short-term memory layers in a neural network to predict human ratings of the accentedness of recorded speech. Speech data are extracted in 5-s segments from over 2000 first- and second-language English speakers from multiple corpora. Human ratings are obtained in an online experiment where participants rate the accentedness of a given speech recording on a 9-point Likert scale. Mel-frequency cepstral coefficients and mel-filterbank energy features are tested as speech input representations for the neural network. When inference is tested using 10-fold cross validation, the mean correlation between the model's predictions and human ratings is high ($r = 0.74$). While previous methods attained a similar correlation by automatically comparing speech that has been transcribed [Wieling *et al.*, *Lang. Dyn. Chang.* 4, 253–269 (2014)] or by making accent-specific Gaussian mixture models [Cheng *et al.*, *Interspeech 2013* (2013), pp. 2574–2578], the present model requires no transcription and can perform accent-general inference.

5pSC14. Automatic proficiency judgments: Accentedness, fluency, and comprehensibility. Seongjin Park (Dept. of Linguist., Univ. of Arizona, Tucson, AZ 85721, seongjinpark@email.arizona.edu) and John Culnan (Linguist., Univ. of Arizona, Tucson, AZ)

The goal of the present study is to investigate whether computational models can approximate human perceptual judgments of accentedness, fluency, and comprehensibility in non-native speech using low-level acoustic features and speech rhythm features. Previous studies have used the results of automatic speech recognition systems, such as word error rate, as features to automatically measure speakers' accentedness or fluency. However, in the present study, we aim to build automatic perceptual judgment model for lexically constrained speech using only audio vectors (wav2vec), acoustic features, and durational features. The results show that a neural network model could make predictions on speakers' proficiency using limited features, and that low-level acoustic features can take the place of suprasegmental features when these are not explicitly included in a given feature set. In addition, the results demonstrate that acoustic features are more useful when predicting accentedness of speech, while durational features are more useful when predicting fluency.

5pSC15. Audio-based spam call detection. Benjamin M. Elizalde (Res., Microsoft, Redmond, WA) and Dimitra Emmanouilidou (Res., Microsoft, 19201 226th Ave. NE, Woodinville, WA 98077, Dimitra.emmanouilidou@microsoft.com)

Spam communications are organized attempts of falsified claims with the purpose of marketing, spreading false information and deceiving the end recipient. Phone spam is an international nuisance, with the U. S. among the most spammed countries in the world in 2020. In addition to the agitating nature of these calls, criminal scams are defrauding subscribers of billions of dollars every year. Therefore, it is necessary to develop automated systems for the identification of spam calls to minimize fraud and reduce the displeasure of receiving them. The call origin, call duration and other Call Detail Records can be used to assess whether a call is fraudulent or not, but the actual audio content is overlooked. This work focuses on extracting acoustic features from voicemail recordings containing speech, which are used to train Machine Learning models that identify spam calls. Both local and global feature descriptors are used, including Mel-Frequency Cepstral Coefficients and Log-Mel Spectrum, and their efficacy for distinguishing spam from non-spam calls is explored. We demonstrate that a spam voice call can be detected while relying only on the acoustic information of the call. A further analysis of the temporal and spectral features that are most informative for the task is also presented.

5pSC16. Accuracy of the Language Environment Analysis (LENA) speech processing system in identifying communicative vocalizations of young children and adults. Laura Dilley (Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48823, elsiedee@alum.mit.edu) and Derek Houston (The Ohio State Univ., Columbus, OH)

The Language Environment Analysis (LENA) system is an automated audio processing system widely used for characterizing language behaviors of children and adults for clinical and basic research. While a number of studies have assessed LENA's reliability, its *accuracy* at identifying and counting speech communicative events is still not well-characterized under a range of naturalistic conditions. In two studies, we examined accuracy of LENA's speech vocalization classifications, relative to human gold standard coding for audio events, as well as word and speech vocalization counts for adults and child utterances, respectively. We found that the weighted average of accurate classification of 100-msec frames by LENA for child speech, adult female speech, and adult male speech was 57%, 61%, and 57%, respectively. Further, an analysis of LENA's ability to accurately discriminate frames of speech vocalizations from a "key child"—a child wearing the LENA device—from speech vocalizations of other child and adult talkers and sound sources showed that LENA correctly detected key child speech vocalization frames only 41% of the time. We are currently extending this research to examine the accuracy of LENA's child vocalization count (CVC) and conversational turn count (CTC) measures.

5pSC17. Gender classification from speech using convolutional networks augmented with synthetic spectrograms. Skyler Horn (Elec. and Comput. Eng., Univ. of Wisconsin-Platteville, Platteville, WI) and Hynek Boril (Elec. and Comput. Eng., Univ. of Wisconsin-Platteville, 320 Busby Hall, 1 University Plaza, Platteville, WI 53818, borilh@uwplatt.edu)

Automatic gender classification from speech is an integral component of human-computer interfaces. Gender information is utilized in user authentication, speech recognizers, or human-centered intelligent agents. This study focuses on gender classification from speech spectrograms using AlexNet-inspired 2D convolutional neural networks (CNN) trained on real samples augmented with synthetic spectrograms. A generative adversarial network (GAN) is trained to produce synthetic male/female-like speech spectrograms. In limited training data experiments on LibriSpeech, augmenting a training set of 200 real samples by 800 synthetic samples reduces equal error rate of the classifier from 23.7% to 1.0%. To further test the 'quality' of the generated samples, in a subsequent experiment, the real training samples are progressively replaced (rather than augmented) with synthetic samples at various ratios from 0 (all original samples preserved) to 1 (all original samples replaced by synthetic ones). Depending on the system setup, substituting between 50% to 90% of the original samples with the synthetic ones is found to have a minimal impact on the classifier performance. Finally, viewing the input CNN layers as filters that select salient spectrogram features, the learned convolutional kernels and filter outputs are studied to understand which spectrogram areas receive a prominent attention in the classifier.

5pSC18. Emotion recognition with speech articulatory coordination features. Yashish M. Siriwardena (Elec. and Comput. Eng., Univ. of Maryland College Park, 8223 Paint Branch Dr., College Park, MD 20742, yashish@terpmail.umd.edu), Nadee Seneviratne, and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland College Park, College Park, MD)

Mental health illnesses like Major Depressive Disorder and Schizophrenia affect the coordination between articulatory gestures in speech production. Coordination features derived from Vocal tract variables (TVs) predicted by a speech inversion system can quantify the changes in articulatory gestures and have proven to be effective in the classification of mental health disorders. In this study we use data from the IEMOCAP (acted emotions) and MSP Podcast (natural emotions) datasets to understand how coordination features extracted from TVs can be used to capture changes between different emotions for the first time. We compared the eigenspectra extracted from channel delay correlation matrices for Angry, Sad and Happy emotions with respect to the "Neutral" emotion. Across both the datasets, it was observed that the "Sad" emotion follows a pattern suggesting simpler articulatory coordination while the "Angry" emotion follows the opposite showing signs of complex articulatory coordination. For the majority of subjects, the 'Happy' emotion follows a complex articulatory coordination pattern, but has significant confusion with "Neutral" emotion. We trained a Convolutional Neural Network with the coordination features as inputs to perform emotion classification. A detailed interpretation of the differences in eigenspectra and the results of the classification experiments will be discussed.

5pSC19. Eye-tracking during remote collaborative communication under noisy, reverberant and delayed communication transmission conditions. Vishakha Rawool (Commun. Sci. & Disord., Univ. of MS, 1006 Briarwood Dr., Oxford, MS 38655, vishakharawool8@gmail.com)

This study, which is part of a larger study, was designed to monitor eye-tracking parameters during remote collaborative communication. Ninety (45 men and 45 women) participants were divided into 30 teams with three members in each team. Each team built different LEGO models under three communication transmission delay conditions in the following order: No delay, 750 ms delay and 5000 ms delay. One individual served as the LEGO builder, who did not have access to the instruction manual. The remaining two team members provided instructions for building the models. Participants were in separate rooms and communicated via earphones. Background white noise (50 dB SPL) was introduced in the ear canals and reverberation was applied to the transmitted communication. Participants were fitted with Tobii Pro Glasses 2 with a sampling rate of 100 fps for eye-tracking. ANOVA was performed on several eyetracking parameters including fixation rate/minute, total whole fixation duration/minute and saccades rate/minute. The results revealed that the cognitive load/or listening effort was lowest in the 750 ms condition compared to the no delay or 5000 ms delay conditions. Detailed results will be discussed during the presentation. Acknowledgement: Graduate students Sydney Osbarn and Melissa Reyes assisted in data collection.

5pSC20. Heart monitoring during remote collaborative communication under noisy, reverberant and delayed communication transmission conditions. Vishakha Rawool (Commun. Sci. & Disord., Univ. of MS, 1006 Briarwood Dr., Oxford, MS 38655, vishakharawool8@gmail.com)

This study, which is part of a larger study, was designed to monitor heart rate during remote collaborative communication. Ninety (45 men and 45 women) participants were divided into 30 teams with 3 members in each team. Each team built different LEGO models under three communication transmission delay conditions in the following order: No delay, 750 ms delay and 5000 ms delay. One individual served as the LEGO builder, who did not have the instruction manual. The remaining two team members provided instructions for building the models. Participants were in separate rooms and communicated via earphones. Background white noise (50 dB SPL) was introduced in the ear canals and reverberation was applied to the transmitted communication. Participants were fitted with Empatica E4 wristbands on their wrists. ANOVA was performed on the averaged heart rate and the RMSDD (the root mean square of successive differences between normal heartbeats). The results revealed lower heart rates and increased heart rate variability in the 750 and 5000 ms delay conditions compared to the no delay condition. Detailed results will be discussed during the presentation. Acknowledgement: Graduate students Sydney Osbarn and Melissa Reyes assisted in data collection. Sydney Osbarn also assisted in pre-processing the data.

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.

Plagiarism

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.

Publication Credit

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.

Proper acknowledgment of the work of others used in a research project must always be given. Information obtained privately, as in conversation, correspondence, or discussion with third parties, should not be used or reported without explicit permission from the investigator with whom the information originated. Information obtained in the course of confidential services, such as refereeing manuscripts or grant applications, cannot be used without permission of the author of the work being used.

Authors must obtain permission when reproducing or adapting any previously published materials from the original copyright holder. Proper credit lines for all previously published material must be included in the manuscript.

Reporting Research Results

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.

Reporting Errors in Publication

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.

Sustaining Members of the Acoustical Society of America

ASA
ACOUSTICAL SOCIETY
OF AMERICA

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 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

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R&D company providing custom advanced research based solutions

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Twinsburg, Ohio
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www.kineticsnoise.com
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Kinetics manufactures products to address vibration and noise control, room acoustics, and seismic restraint concerns for almost any building application

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Manufacture Loudspeakers and Acoustical Test Equipment

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www.ncac.com
Indianapolis, Indiana
An Association of Independent Firms Consulting in Acoustics

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www.nationalgypsum.com
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Manufacturer of acoustically enhanced gypsum board

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Offers a variety of insulation products ranging in density and dimension to meet any production requirements. Products are successfully used in numerous acoustical OEM applications providing solutions for a number of industries

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Somerset, United Kingdom
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Products for personal and environmental safety, featuring E·A·R and Peltor brand hearing protection and fit testing, Quest measurement instrumentation, audiological devices, materials for control of noise, vibration, and mechanical energy, and the E·A·RCALSM laboratory for research, development, and education, NVLAP-accredited since 1992.
Hearing conservation resource center
www.e-a-r.com/hearingconservation

Wenger Corporation
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The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

ACOUSTICAL · SOCIETY · OF · AMERICA

APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

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.

Name of Company ____________________

Address ____________________

Telephone: ____________________ Fax: ____________________

E-mail: ____________________ WWW: ____________________

Size of Business: ☐ Small business ☐ Small business—No Journal ☐ Large business

Type of Business ____________________

Please enclose a copy of your organization's brochure.

In listing of Sustaining Members in the Journal and on the ASA homepage we should like to indicate our products or services as follows:

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(please do not exceed fifty characters)

Name of company representative to whom journal should be sent:

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It is understood that a Sustaining Member will not use the membership for promotional purposes.

Signature of company representatives making application:

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Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360, asa@acousticalsociety.org

MEMBERSHIP INFORMATION AND APPLICATION INSTRUCTIONS

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

BENEFITS OF MEMBERSHIP	full Member	Associate	ce-Associate	Student
JASA Online–Vol. 1 (1929) to present	*	*	*	*
JASA tables of contents e-mail alerts	*	*	*	*
JASA, printed	*	*		
JASA Express Letters–online	*	*	*	*
Acoustics Today–the quarterly magazine	*	*	*	*
Proceedings of Meetings on Acoustics	*	*	*	*
Noise Control and *Sound*, *It's Uses and Control*–online archival magazines	*	*	*	*
Acoustics Research Letters Online (ARLO)–online archive	*	*	*	*
Programs for Meetings	Online	Online	Online	Online
Meeting Calls for Papers	Online	Online	Online	Online
Reduced Meeting Registration Fees	*	*		*
Society Membership Directory	Online	Online	Online	Online
Electronic Announcements	*	*	*	*
Physics Today	*	*	*	*
Eligibility to vote and hold office in ASA	*			
Eligibility to be elected Fellow	*	*		
Participation in ASA Committees	*	*	*	*

QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

Student: Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $50 per year.
Associate: Any individual interested in acoustics. Dues: $115 per year. After five years, the dues of an Associate increase to that of a full Member.
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.

JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
- PRINT JOURNAL. Twelve monthly issues of *The Journal of the Acoustical Society of America*. **Cost: $35 in addition to dues.**
- **EFFECTIVE DATE OF MEMBERSHIP**. If your application for membership and dues payment are received by 15 September, your membership and Journal subscription will begin during the current year and you will receive all back issues for the year. If you select the print journal option. If your application is received after 15 September, however, your dues payment will be applied to the following year and your Journal subscription will begin the following year.

OVERSEAS AIR DELIVERY OF JOURNALS

Members outside North, South, and Central America can choose to have print journals sent by air freight at a cost of $185 in addition to dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

ACOUSTICAL SOCIETY OF AMERICA

1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, asa@acousticalsociety.org

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APPLICATION FOR MEMBERSHIP

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

CHECK ONE BOX IN EACH COLUMN ON THE RIGHT	☐ NON-MEMBER APPLYING FOR: ☐ MEMBER REQUESTING TRANSFER TO:	☐ STUDENT MEMBERSHIP ☐ ASSOCIATE MEMBERSHIP ☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP ☐ FULL MEMBERSHIP	Note that your choice of journal option *may* increase or decrease the amount you must remit.

SELECT JOURNAL OPTION:

Student members will automatically receive access to The Journal of the Acoustical Society of America online at no charge in addition to dues. Remit $45.

Corresponding Electronic Associate Members will automatically receive access to The Journal of the Acoustical Society of America and Meeting Programs online at no charge in addition to dues. Remit $50.

Applicants for **Associate or full Membership** must select one Journal option from those listed below. Note that your selection of journal option determines the amount you must remit.

[ ] Online access only—$150
[ ] Online access plus print Journal $185

Applications received after 15 September: Membership and Journal subscriptions begin the following year.

OPTIONAL AIR DELIVERY: Applicants from outside North, South, and Central America may choose air freight delivery of print journals for an additional charge of $185. If you wish to receive journals by air, remit the additional amount owed with your dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

LAST NAME	FIRST NAME	MIDDLE INITIAL	MS/MR/MRS/DR/PROF
HOME ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
BUSINESS TELEPHONE: AREA CODE/NUMBER	FAX: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER	
E-MAIL ADDRESS: (PRINT CLEARLY)	MOBILE PHONE: AREA CODE/NUMBER		
DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status)	SEX: ☐ Female ☐ Male ☐ Non-Binary ☐ Transgender ☐ Prefer not to answer ☐		
HIGHEST ACADEMIC DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE
OTHER DEGREE	MONTH/YEAR	FIELD	INSTITUTION GRANTING DEGREE

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

Part I Continued ➛

PART I CONTINUED: ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

- ☐ ACOUSTICAL OCEANOGRAPHY **M**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ARCHITECTURAL ACOUSTICS **A**
- ☐ BIOMEDICAL ACOUSTICS **K**
- ☐ COMPUTATIONAL ACOUSTICS **O**
- ☐ ENGINEERING ACOUSTICS **B**
- ☐ MUSICAL ACOUSTICS **C**
- ☐ NOISE & NOISE CONTROL **D**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ UNDERWATER ACOUSTICS **J**

PART II: APPLICATION FOR STUDENT MEMBERSHIP

NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED		
DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME		
SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required.

SPONSORS AND REFERENCES: An application for full Membership requires the names, and email addresses of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

PRINT NAME OF REFERENCE (required for Full Member applications only)	PRINT NAME OF SECOND REFERENCE (required for Full Member applications only)
EMAIL ADDRESS OF REFERENCE	EMAIL ADDRESS OF SECOND REFERENCE
SIGNATURE OF APPLICANT	DATE

MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO: ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 300, MELVILLE, NY 11747-4300; FAX: 631-923-2875

METHOD OF PAYMENT

☐ Check or money order enclosed for $ ____________ (U.S. funds/drawn on U.S. bank)

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Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please return this form by Fax (631-923-2875) or by postal mail.

Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn Hoglund, Ohio State University, hoglund1@osu.edu and Sandra Guzman, Shure, Inc., guzman_sandra@shure.com

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AUTHOR INDEX

to papers presented at

181th Meeting of the Acoustical Society of America

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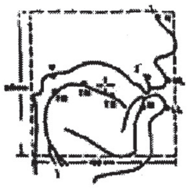

NOW AVAILABLE ON DVD

MEASURING SPEECH PRODUCTION:
VIDEO DEMONSTRATIONS OF SPEECH INSTRUMENTATION

This series of demonstrations, for use in teaching courses on speech acoustics, physiology and instrumentation are now available on DVD from the Acoustical Society of America. The DVD contains thirteen video demonstrations of equipment and techniques used in speech research. The demonstrations are categorized into three areas: (1) Respiration, phonation and aerodynamics; (2) Indirect articulatory measurements; (3) Direct articulatory measurements. A pdf file on the DVD describes the demonstrations and lists additional readings that are updated from the original videotape.

PART ONE - RESPIRATION, PHONATION AND AERODYNAMICS
1. The whole body plethysmograph in speech research. John J. Ohala
2. Aerodynamic end respiratory kinematic measures during speech. Elaine T. Stathopoulos
3. Physiologically based models of phonation. Ingo R. Titze
4. Use of the electroglottograph in the laboratory and clinic. James J. Mahshie
5. Endoscopy, stroboscopy, and transillumination in speech research. Anders Lofqvist, Kiyoshi Oshima

PART TWO - INDIRECT ARTICULATORY MEASUREMENTS
6. Magnetic resonance imaging (MRI) in speech research. Carol Gracco, Mark Tiede
7. Imaging the tongue with ultrasound. Maureen Stone
8. Estimating articulatory movement from acoustic data. Kenneth N. Stevens
9. Electromyography in speech research, Kiyoshi Oshima. Katherine S. Harris, Fredericka Bell-Berti

PART THREE - DIRECT ARTICULATORY MEASUREMENTS
10. The rise and fall of the soft palate: The Velotrace. Fredericka Bell-Berti, Rena A. Krakow, Dorothy Ross, Satoshi Horiguchi
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13. Optoelectronic measurement of orofacial motions during speech production. Eric Vatikiotis-Bateton, Kevin Munhall, David Ostry

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