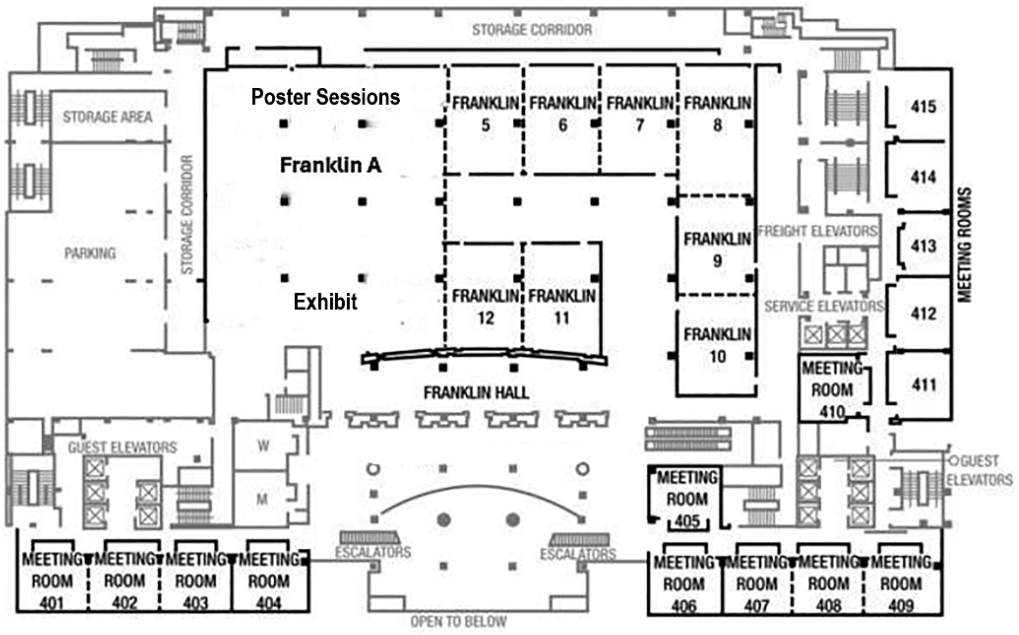
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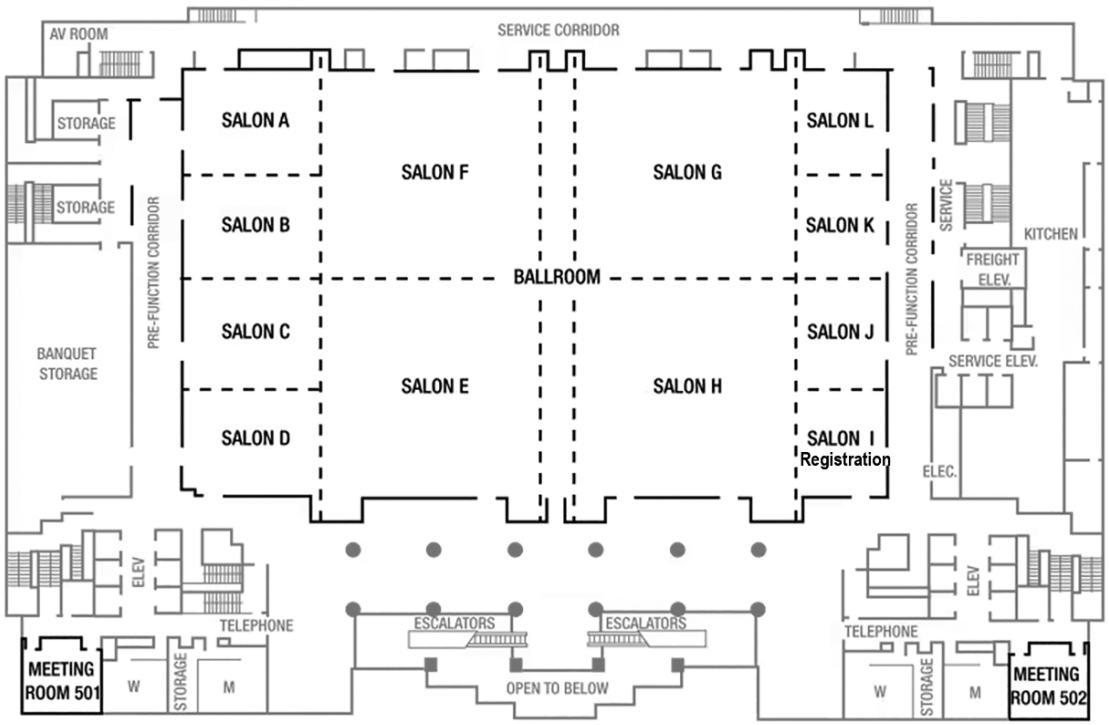
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LEVEL 4 FRANKLIN HALL



LEVEL 5 BALLROOM



TECHNICAL PROGRAM CALENDAR

190th Meeting of the Acoustical Society of America
11-15 May 2026

Monday Morning

8:00 1aAAa **Architectural Acoustics:** Sustainability in Architectural and Building Acoustics. Salon C

10:30 1aAAb **Architectural Acoustics:** New Challenges in Stage Acoustics Design. Salon B

7:55 1aAB **Animal Bioacoustics and Psychological and Physiological Acoustics:** Effects of Hearing Loss and Abnormal Acoustic Experience on Cognitive and Affective Behavior in Animals and Humans. Franklin Hall 5

8:30 1aAOa **Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics:** Acoustic Observations from Uncrewed Platforms I. Salon A

11:00 1aAOb **Acoustical Oceanography:** Acoustical Oceanography Prize Lecture/A.B. Wood Medal Lecture. Salon A

8:00 1aBAa **Biomedical Acoustics, Physical Acoustics, and Signal Processing in Acoustics:** Advanced Ultrasound Vascular Imaging and Vessel Characterization I. Franklin Hall 9

10:15 1aBAb **Biomedical Acoustics:** General Topics in Biomedical Acoustics: Advanced Methods. Franklin Hall 7

10:00 1aCA **Computational Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Computation in Phononics and Metamaterials. Franklin Hall 11

8:45 1aEA **Engineering Acoustics:** Acoustic Transducers and Sensors. Franklin Hall 12

8:15 1aPAa **Physical Acoustics:** Acoustics at the Synchrotron and Neutron Beamlines. Franklin Hall 1

11:00 1aPAb **Physical Acoustics:** General Physical Acoustics: Manufacturing. Franklin Hall 10

8:00 1aSP **Signal Processing in Acoustics, Acoustical Oceanography, and Animal Bioacoustics:** Signal Processing and Modeling for Distributed Acoustic Sensing (DAS) I. Salon D

Monday Afternoon

1:00 1pAA **Architectural Acoustics, Engineering Acoustics, and Noise:** Memorial Session Honoring Robert Coffeen. Salon C

1:00 1pAB **Animal Bioacoustics:** Learning and Coordination of Vocal Signaling. Franklin Hall 5

1:00 1pAO **Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics:** Acoustic Observations from Uncrewed Platforms II. Salon A

1:10 1pBA **Biomedical Acoustics, Physical Acoustics, and Signal Processing in Acoustics:** Advanced Ultrasound Vascular Imaging and Vessel Characterization II. Franklin Hall 9

1:00 1pEAa **Engineering Acoustics and Physical Acoustics:** 3D Printing-Enabled Acoustic Transducers and Sensors. Franklin Hall 12

3:00 1pEAb **Engineering Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Piezoelectric Metamaterials. Franklin Hall 12

1:00 1pED **Education in Acoustics:** Sound Science in Informal Learning Settings. Franklin Hall 11

1:00 1pID **Interdisciplinary and Student Council:** Undergraduate and Graduate Program in Acoustics (Poster Session). Franklin Hall A

1:00 1pPA **Physical Acoustics and Noise:** Atmospheric Turbulence Effects on Sound Propagation. Franklin Hall 10

1:00 1pPP **Psychological and Physiological Acoustics:** Processing of Affective Sounds: Intersection of Audition and Emotion. Franklin Hall 8

1:15 1pSA **Structural Acoustics and Vibration:** Laser-Based Measurement and Wave Control. Salon B

1:00 1pSC **Speech Communication:** Production & Articulation (Poster Sesson). Franklin Hall A

1:00 1pSP **Signal Processing in Acoustics, Acoustical Oceanography, and Animal Bioacoustics:** Signal Processing and Modeling for Distributed Acoustic Sensing (DAS) II. Salon D

Tuesday Morning

8:00 2aAAa **Architectural Acoustics:** Student Design Competittion. Franklin A

8:15 2aAAb **Architectural Acoustics, Noise, and ASA Committee on Standards:** Acoustics in Public Accommodations Spaces Part I. Salon C

8:00 2aAB **Animal Bioacoustics:** Acoustics Ecology and Biological Soundscapes. Franklin Hall 5

8:30 2aAO **Acoustical Oceanography:** Geoacoustic Inversion and Sediment Characterization. Salon A

7:55 2aBAa **Biomedical Acoustics:** Biomedical Methods: Messy, Marvelous, and Meh I. Franklin Hall 9

7:55 2aBAb **Biomedical Acoustics and Physical Acoustics:** Ultrasound Beamforming and Its Applications I. Franklin Hall 7

9:00 2aCA **Computational Acoustics, Acoustical Oceanography, Underwater Acoustics, and Physical Acoustics:** 3D Acoustics. Franklin Hall 11

9:00 2aMU **Musical Acoustics and Speech Communication:** Cross-Cultural Musical Instruments and Instruments Across Culture. Franklin Hall 6

9:00 2aNS **Noise:** Community Noise Testing and Analysis I. Salon D

7:55 2aPA **Physical Acoustics Biomedical Acoustics, Computational Acoustics, Education in Acoustics, Noise, Structural Acoustics and Vibration, and Archives and History:** Celebration of Allan D. Pierce's Contributions to Acoustics I. Franklin Hall 10

8:00 2aPP **Psychological and Physiological Acoustics:** Individual Factors in Spatial Speech Unmasking. Franklin Hall 8

8:00 2aSA **Structural Acoustics and Vibration and Physical Acoustics:** Acoustic Metamaterials I. Salon B

9:00 2aSC **Speech Communication:** Acoustics Voice Quality and Prosody (Poster Session). Franklin Hall A

8:00 2aSP **Signal Processing in Acoustics and Acoustical Oceanography:** Bayesian Signal Processing I. Franklin Hall 12

9:30 2aUW **Underwater Acoustics:** Methods and Theory in Underwater Acoustics. Salon F

Tuesday Afternoon

1:00 2pAAa **Architectural Acoustics, Noise, and ASA Committee on Standards:** Acoustics in Public Accommodations Spaces Part II. Salon C

1:00 2pAAb **Architectural Acoustics:** Advances in Acoustics of Mass Timber Assemblies. Salon F

1:00 2pABa **Animal Bioacoustics:** Animal Vocal Communication and Physiology. Franklin Hall 5

2:45 2pABb **Animal Bioacoustics:** General Topics in Animal Bioacoustics. Franklin Hall 5

1:00 2pAO **Acoustical Oceanography:** Ocean Acoustic Propagation Modeling. Salon A

1:00 2pBAa **Biomedical Acoustics:** Biomedical Methods: Messy, Marvelous, and Meh II. Franklin Hall 9

1:45 2pBAb **Biomedical Acoustics and Physical Acoustics:** Ultrasound Beamforming and Its Applications II. Franklin Hall 7

1:15 2pCA **Computational Acoustics:** Verification, Validation, and Uncertainty Quantification. Franklin Hall 11

1:00 2pEA **Engineering Acoustics, Archives and History, and Public Relations:** The Philadelphia Acoustics Story. Franklin Hall 12

1:00 2pED **Education in Acoustics:** General Topics in Education in Acoustics. Salon E

1:30 2pMU **Musical Acoustics:** General Topics in Musical Acoustics. Franklin Hall 6

1:15 2pNSa **Noise:** Community Noise Testing and Analysis II. Salon D

3:15 2pNSb **Noise, Architectural Acoustics, and Structural Acoustics and Vibration:** Building Base Isolation. Salon D

1:00 2pPA **Physical Acoustics Biomedical Acoustics, Computational Acoustics, Education in Acoustics, Noise, Structural Acoustics and Vibration, and Archives and History:** Celebration of Allan D. Pierce's Contributions to Acoustics II. Franklin Hall 10

1:00 2pPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics: Best Student Award Poster Session. Franklin Hall A

1:00 2pSA **Structural Acoustics and Vibration and Physical Acoustics:** Acoustic Metamaterials II. Salon B

1:00 2pSC **Speech Communication:** Technology, Methods & Computational Modeling (Poster Session). Franklin Hall A

1:00 2pSP **Signal Processing in Acoustics:** Signal Processing Potpourri I. Franklin Hall 8

Wednesday Morning

8:35 3aAA **Architectural Acoustics and Noise:** Interesting Measurement Techniques and Results Part I. Salon C

7:55 3aAB **Animal Bioacoustics, Computational Acoustics, and Acoustical Oceanography:** Machine Learning Approaches for Analyzing Large Acoustic Datasets I. Franklin Hall 5

8:00 3aBAa **Biomedical Acoustics and Physical Acoustics:** Vortex Ultrasound. Franklin Hall 9

8:00 3aBAb **Biomedical Acoustics:** General Topics in Biomedical Acoustics: Quantitative Ultrasound and Doppler. Franklin Hall 7

8:55 3aEA **Engineering Acoustics and Physical Acoustics:** Artificial Intelligence (AI) Meets Electroacoustic Transducers: Emerging Methods and Applications. Franklin Hall 12

7:55 3aID **Interdisciplinary, Education in Acoustics, and Archives and History:** Student Paper Competition - Seminal Acoustics Papers. Franklin Hall 6

9:00 3aNS **Noise:** Sounds of Freedom. Salon D

8:30 3aPA **Physical Acoustics and Computational Acoustics:** Software for Infrasound Research and Applications. Franklin Hall 10

8:00 3aPP **Psychological and Physiological Acoustics:** Blossoming Science and Cultivating Researchers in Psychological and Physiological Acoustics. Franklin Hall 8

8:00 3aSP **Signal Processing in Acoustics and Acoustical Oceanography:** Bayesian Signal Processing II. Franklin Hall 11

8:30 3aUW **Underwater Acoustics:** Acoustic Scattering. Salon B

Wednesday Afternoon

1:00 3pAA **Architectural Acoustics:** Interesting Measurement Techniques and Results Part II. Salon C

1:00 3pAB **Animal Bioacoustics, Computational Acoustics, and Acoustical Oceanography:** Machine Learning Approaches for Analyzing Large Acoustic Datasets II. Franklin Hall 5

1:00 3pAO **Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, Signal Processing in Acoustics, and Physical Acoustics:** New Results and Future Directions for Ocean Sensing Using Seismic Reflection Platforms. Franklin Hall 9

1:00 3pBA **Biomedical Acoustics:** Biomedical Acoustics Best Student Paper Award Poster Session. Franklin Hall A

1:30 3pCA **Computational Acoustics:** Showcases of High Performance Computing in Acoustics (Poster Session). Franklin Hall A

2:15 3pED **Education in Acoustics:** Acoustics Education Prize Lecture. Franklin Hall 12

1:00 3pID **Interdisciplinary and Student Council:** Hot Topics in Acoustics. Franklin Hall 7

1:00 3pNS **Noise:** Pickleball Noise. Salon D

1:00 3pPA **Physical Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Animal Bioacoustics:** Sensing Vector Acoustic Fields. Franklin Hall 10

1:00 3pPP **Psychological and Physiological Acoustics:** Auditory Neuroscience Prize Lecture. Franklin Hall 8

1:00 3pSA **Structural Acoustics and Vibration:** Structural Acoustics and Vibrations Tutorial. Salon A

1:00 3pSP **Signal Processing in Acoustics:** Convex Optimization and Sparsity in Signal Processing. Franklin Hall 6

1:00 3pUW **Underwater Acoustics:** Air-Sea Interaction Acoustics. Salon B

Thursday Morning

8:15 4aAA **Architectural Acoustics, Psychological and Physiological Acoustics, Signal Processing in Acoustics, and Speech Communication:** The Many Dimensions of Jont B. Allen's Scientific Legacy. Salon C

8:00 4aBA **Biomedical Acoustics:** Shocking Considerations in Therapeutic Ultrasound I. Franklin Hall 9

9:00 4aCA **Computational Acoustics, Biomedical Acoustics, Physical Acoustics, and Underwater Acoustics:** Ray Methods Across Acoustics. Franklin Hall 11

10:30 4aEA **Engineering Acoustics:** General Topics in Engineering Acoustics. Franklin Hall 12

9:00 4aED **Education in Acoustics and CIRDI:** SURIEA - Past and Future. Franklin Hall 7

9:00 4aMU **Musical Acoustics and Signal Processing in Acoustics:** Parameter Estimation for Musical Acoustics. Franklin Hall 6

8:30 4aNS **Noise:** Methods in Soundscape Assessment and Design. Salon D

8:00 4aPAa **Physical Acoustics:** Infrasound I. Franklin Hall 5

8:05 4aPAb **Physical Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics:** Ultrasonic Characterization Across Scales and Applications I. Franklin Hall 10

8:00 4aPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics: Poster Session. Franklin Hall A

9:00 4aSA **Structural Acoustics and Vibration:** Advanced Modeling and Computational Methods. Salon A

8:00 4aSC **Speech Communication:** Exploring Creaky Voice/Glottal Fry Across Disciplines. Franklin Hall 8

8:30 4aSP **Signal Processing in Acoustics:** Signal Processing Potpourri II. Salon F

8:30 4aUW **Underwater Acoustics and Physical Acoustics:** Session in Honor of Thomas G. Muir. Salon B

Thursday Afternoon

1:00 4pAA **Architectural Acoustics, Musical Acoustics, and Psychological and Physiological Acoustics:** Audio for Architectural Acoustics, Indoors and/or Outdoors. Salon C

1:00 4pBAa **Biomedical Acoustics:** Shocking Considerations in Therapeutic Ultrasound II. Franklin Hall 9

2:30 4pBAb **Biomedical Acoustics:** Wearable Ultrasound. Franklin Hall 9

3:45 4pBAc **Biomedical Acoustics:** General Topics in Biomedical Acoustics: Therapy, Microbubbles, and Beyond. Franklin Hall 7

1:30 4pCA **Computational Acoustics:** General Topics in Computational Acoustics. Franklin Hall 11

1:00 4pED **Education in Acoustics and Public Relations:** Sound Communication in Today's Mediascape. Franklin Hall 12

1:30 4pMU **Musical Acoustics:** Musical Acoustics Poster Session. Franklin Hall A

1:00 4pPAa **Physical Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics:** Ultrasonic Characterization Across Scales and Applications II. Franklin Hall 10

1:00 4pPAb **Physical Acoustics:** Infrasound II. Franklin Hall 5

3:30 4pPAc **Physical Acoustics:** Physical Acoustics Poster Session. Franklin Hall A

1:00 4pSA **Structural Acoustics and Vibration:** Lessons Learned: Mistakes, Missteps, and Miscalculations in Structural Acoustics and Vibration. Salon A

1:30 4pSCa **Speech Communication:** Cognition, Social Factors & Higher-Level Influences (Poster Session). Franklin Hall A

1:30 4pSCb **Speech Communication:** Bilingualism & L2 Learning (Poster Session). Franklin Hall A

1:30 4pUW **Underwater Acoustics:** Ocean Soundscapes and Underwater Noise. Salon B

Friday Morning

9:45 5aAA **Architectural Acoustics:** Acoustics Potpourri. Salon C

8:00 5aID **Interdisciplinary and Committee on Outreach and Education:** Listen Up and Get Involved. Salon F

7:55 5aPA **Physical Acoustics:** General Physical Acoustics. Franklin Hall 10

9:00 5aSA **Structural Acoustics and Vibration:** Damping, Dissipation, and Energy Attenuation. Salon A

9:00 5aSC **Speech Communication:** Speech Perception & Listening Conditions (Poster Session). Franklin Hall A

SCHEDULE OF COMITTEEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Sun, 10 May, 6:30 p.m.	ASA Board Dinner	404
Mon, 11 May, 9:00 a.m.	Executive Council	407-409
Mon, 11 May, 1:00 p.m.	Technical Council	407-409
Tue, 12May, 7:00 a.m.	Member Engagement	406
Tue, 12 May, 7:30 a.m.	Editorial Board	407-409
Tue, 13 May, 1:30 p.m.	CIRDI	404
Tue, 12 May, 1:30 p.m.	Meetings	410
Wed, 13 May, 7:30 a.m.	Finance	404
Wed, 13 May, 7:30 a.m.	Regional and Student Chapters	407-409
Wed, 13 May, 9:30 a.m.	Acoustical Society Foundation	407
Wed, 13 May, 12:15 p.m.	Public Relations	404
Thu, 13 May, 7:00 a.m.	Investments	407
Thu, 13 May, 7:30 a.m.	Practitioners and Industry	406
Fri, 14 May, 8:00 a.m.	Technical Council	407-409
Fri, 14 May, 11:00 noon	Executive Council	407-409

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 12 May, 5:30 p.m.	Engineering Acoustics	Franklin 5
Tue, 12 May, 5:30 p.m.	Musical Acoustics	Franklin 6
Tue, 12 May, 5:30 p.m.	Signal Processing in Acoustics	Franklin 8
Tue, 12 May, 7:30 p.m.	Acoustical Oceanography and Underwater Acoustics	Salon A
Tue, 12 May, 7:30 p.m.	Architectural Acoustics	Salon C
Tue, 12 May, 7:30 p.m.	Psychological and Physiological Acoustics	Franklin 8
Wed, 13 May, 7:30 p.m.	Animal Bioacoustics	Franklin 5
Wed, 13 May, 7:30 p.m.	Biomedical Acoustics	Franklin 9
Wed, 13 May, 7:30 p.m.	Education in Acoustics	Salon B
Wed, 13 May, 7:30 p.m.	Structural Acoustics and Vibration	Salon A
Thu, 14 May, 4:30 p.m.	Computational Acoustics	Franklin 11
Thu, 14 May, 5:30 p.m.	Physical Acoustics	Franklin 5
Thu, 14 May, 5:30 p.m.	Speech Communication	Franklin 8
Thu, 14 May, 7:30 p.m.	Noise	Salon D

STANDARDS COMMITTEES AND WORKING GROUPS

Tue, 12 May, 9:00 a.m.	Standards Plenary including TAGs	404
Tue, 12 May, 10:15 a.m.	ASC S12 Noise	404
Tue, 12 May, 11:30 a.m.	ASC S2 Mechanical Vibration and Shock	404
Tue, 12 May, 1:15 p.m.	ASC S3 Bioacoustics	404
Tue, 12 May, 2:30 p.m.	ASC S3/SC1 Animal Bioacoustics	404
Tue, 12 May, 3:45 a.m.	ASC S1 Acoustics	404
Wed, 13 May, 6:00 a.m.	ISO 1996 WG 45	404

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon, 11 May 7:00 a.m.–5:00 p.m. Tue-Thu, 12-14 May 7:30 a.m.–5:00 Fri, 15 May 7:30 a.m.–12:00 noon	Registration	Salons J-K
Mon–Fri, 11–15 May 7:00 a.m.–5:30 p.m.	A/V Preview	405
Mon–Fri, 11–15 May 7:00 a.m.–5:00 p.m.	Mothers' Room	Reg Desk 1
Mon–Fri, 11–15 May, 9:30 a.m.–11:00 a.m.	Morning Coffee Breaks	Franklin A
Tue, 12 May 2:00 p.m.-3:30 p.m.	Afternoon Coffee Break	Franklin A
Mon-Thu, 11-14 May 8:30 a.m.–11:30 a.m.	Accompanying Persons	415
Mon, 11 May 4:30 p.m.-5:30 p.m.	Creating an Acoustics Education Network - Open Discussion	Franklin 7
Mon, 11 May 5:30 p.m.–7:00 p.m.	Exhibit Opening Reception	Franklin A
Mon, 11 May 5:00 p.m.-5:30 p.m.	Students and First Time Attendees	Salon B
Mon, 11 May 5:45 p.m.-7:30 p.m.	Student Meet and Greet	Salon F
Mon, 11 May, 7:00 p.m.-8:00 p.m.	Women in Acoustics Reception	Salon G
Mon, 11 May 4:00 p.m.-7:00 p.m.	Using Computational Methods in Teaching Acoustics	404
Mon-Fri, 11-15 May	Gallery of Acoustics	Franklin A
Tue, 12 May 9:00 a.m.–5:00 p.m. Wed, 13 May 9:00 a.m.-12 noon	Exhibit	Franklin A
Tue, 12 May, 3:00 p.m.-4:00 p.m.	Women's Roundtable Discussion	407-409
Tue, 12 May, 6:00 p.m.–8:00 p.m.	Student Reception	Salon G
Wed, 13 May, 10:30 a.m.-1:30 p.m.	Philadelphia Orchestra Rehearsal	Kimmel Center 300 S Broad St
Wed, 13 May, 3:30 p.m.–5:30 p.m.	Plenary Session/Awards Ceremony	Salon F
Wed, 13 May, 6:00 p.m.–7:30 p.m.	Social	Salons G-H
Wed, 13 May, 8:00 p.m.–midnight	ASA Jam 1100 Chestnut St	Milkboy
Thu, 14 May 2:00 p.m.-4:00 p.m.	Strategic Plan Champions	411-412
Fri, 15 May, 7:30 a.m.-5:00 p.m.	Armstrong Tour Buses	12th St Exit
Fri, 15 May, 8:00 a.m.-12:00 noon	Listen Up and Get Involved	Salon F

190th Meeting of the Acoustical Society of America

The 190th meeting of the Acoustical Society of America will be held Monday through Friday, 11-15 May 2026 at the Philadelphia Marriott Downtown Hotel, Philadelphia, Pennsylvania, USA.

SECTION HEADINGS

1. REGISTRATION
2. TECHNICAL SESSIONS
3. TECHNICAL SESSION DESIGNATIONS
4. ITINERARY PLANNER AND MOBILE APP
5. EXHIBIT AND EXHIBIT OPENING RECEPTION
6. PRIZES AND PRIZE LECTURES
8. GALLERY OF ACOUSTICS
7. TECHNICAL COMMITTEE OPEN MEETINGS
9. WOMEN IN ACOUSTICS ROUNDTABLE
10. TECHNICAL TOURS AND NON-TECHNICAL TOURS
11. INFORMAL CHATS WITH ADMINISTRATIVE COMMITTEES
12. PLENARY SESSION AND AWARDS CEREMONY
13. ANNUAL MEETING OF MEMBERS
14. ANSI STANDARDS COMMITTEES
15. COFFEE BREAKS
16. A/V PREVIEW ROOM
17. MOTHERS ROOM
18. SOCIAL
19. STUDENT EVENTS: NEW STUDENTS/FIRST-TIME ATTENDEE ORIENTATION, MEET AND GREET, STUDENT RECEPTION
20. WOMEN IN ACOUSTICS RECEPTON\
21. JAM SESSION
22. ACCOMPANYING PERSONS PROGRAM
23. ASA WAYFINDERS
24. LOCAL RAIL TO AIRPORT
25. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)
26. TECHNICAL PROGRAM ORGANIZING COMMITTEE
27. MEETING ORGANIZING COMMITTEE
28. PHOTOGRAPHING AND RECORDING
29. ABSTRACT ERRATA
30. GUIDELINES FOR ORAL PRESENTATIONS,
31. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS
32. DATES OF FUTURE ASA MEETINGS

1. REGISTRATION

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

Registration will open on Monday, 11 May, at 7:00 a.m. in Salons I/J on the 5th floor

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

The registration fees (in USD) are $1000 for members of the Acoustical Society of America; $1350 for non-members, $350 for Emeritus members (Emeritus status pre-approved by ASA), $525 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $300 for ASA Student members, $400 for students who are not members of ASA, and $250 for accompanying persons. One-day registration is available at $525 for members and $700 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions.

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 $600.

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

2. TECHNICAL SESSIONS

The technical program includes over 1000 abstracts.

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

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

program schedule.

3. TECHNICAL SESSION DESIGNATIONS

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

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

1-Monday, 11 May
2-Tuesday, 12 May
3-Wednesday, 13 May
4-Thursday, 14 May
5-Friday, 15 May

The second character is a lower case “a” for a.m. or “p” for p.m. corresponding

to the time of day the session will take place. The third and fourth characters are

capital letters indicating the primary Technical Committee that organized the session using the following abbreviations or codes:

AA Architectural Acoustics

AB Animal Bioacoustics
AO Acoustical Oceanography
BA Biomedical Acoustics
CA Computational Acoustics
EA Engineering Acoustics
ED Education in Acoustics
ID Interdisciplinary
MU Musical Acoustics
NS Noise
PA Physical Acoustics
PP Psychological and Physiological Acoustics
SA Structural Acoustics and Vibration
SC Speech Communication
SP Signal Processing in Acoustics
UW Underwater Acoustics

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

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

4. ITINERARY PLANNER AND MOBILE APP

The itinerary planner can be accessed at: https://ativ.me/101i

The Mobile App can be accessed at: https://ativ.me/101h

5. EXHIBIT AND EXHIBIT OPENING RECEPTION

An instrument and equipment exhibition will be located in Franklin A on the 4th floor and will open on Monday, 11 May, with an evening reception serving a complimentary drink. Exhibit hours are Monday, 11 May, 5:30 p.m. to 7:00 p.m., Tuesday, 9 May, 9:00 a.m. to 5:00 p.m., and Wednesday, 10 May, 9:00 a.m. to 12:00 noon.

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

6. PRIZES AND PRIZE LECTURES

The 2026 Auditory Neuroscience Prize in Auditory Neuroscience and the 2025 Rossing Prize in Acoustics Education will be presented at the Plenary session on Wednesday, 13 May, in session 3pPP.

The 2026 Medwin Prize in Acoustical Oceanography at the Acoustical Oceanography/A. B. Wood Medal and Prize Lecture on Monday, 11 May, in session 1aAOb.

The Auditory Neuroscience Prize Lecture will be presented on Wednesday, 13 May, in session 3pPP. The 2026 Acoustical Education Prize Lecture will be presented on Wednesday, 13 May, in session 3pED.

7. 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 in the Schedule of Meetings of Other Events.

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.

8. GALLERY OF ACOUSTICS

The Technical Committee on Computational Acoustics will sponsor the 20th Gallery of Acoustics. Its purpose is to enhance ASA meetings by providing a setting for researchers to display their work to all meeting attendees in a forum emphasizing the diversity, interdisciplinary, and artistic nature of acoustics. The Gallery will consist of a collection of short videos detailing results generated via computational acoustics or describing the computational approach.

The Gallery of Acoustics provides a means by which we can all share and appreciate the natural beauty, aesthetic, and artistic appeal of acoustic phenomena: This is a forum where science meets art. Additionally, many computational aspects of acoustics are more easily explained in video form where additional dimensions can be displayed.

9. WOMEN IN ACOUSTICS ROUNDTABLE

A Roundtable sponsored by the Women in Acoustics Committee will be held on Tuesday, 12 May, 3:00 p.m. to 4:00 p.m. in 407-409.

10. TECHNICAL TOURS AND NON-TECHNICAL TOURS

Arrangements have been made for meeting participants to attend a rehearsal of the Philadelphia Orchestra on Wednesday, 13 May, from 10:30 a.m. to about 1:00 p.m. There is no charge to attend this event. Your ASA meeting badge must be shown to be admitted. The rehearsal will take place at the Kimmel Center, 300 S. Broad St. about a 6 block walk from the Marriott. All attendees must arrive and be in their seats by 10:20 a.m. It is not possible to depart from the rehearsal until the break which is expected to occur about mid-way through the rehearsal.

On Friday there will be an all-day tour to the Armstrong Campus in Lancaster, PA. There is no space for new registrants on this tour.

A guided tour the Kimmel Center for the Performing Arts, including Marian Anderson Hall, Perelman Theater, the Commonwealth Plaza, and other key spaces will be held on

Thursday, May 14 from 12:00 p.m. to 2:00 p.m. The tour will also extend to nearby venues along the Avenue of the Arts, including the historic Academy of Music and, if available, the Miller Theater, offering a broad look at Philadelphia's performance spaces and their acoustical environments. The tour will begin at Kimmel Center's Commonwealth Plaza, located at 300 S Broad St, precisely at 12 noon. For more information on these spaces, please visit https://www.ensembleartsphilly.org/plan-your-visit/our-venues.

11. INFORMAL CHATS WITH ADMINISTRATIVE COMMITTEES

During the coffee breaks on Monday, Tuesday, Thursday, and Friday, join the Member Engagement Committee for an opportunity to connect with representatives from various administrative committees across the ASA. This informal session is your chance to ask questions, share ideas, and learn more about each of the different committees. Please direct any questions to Christina Naify (christina.naify@gmail.com) and Martin Lawless (ME chair, martin.lawless13@gmail.com). Grab a coffee and come chat!

12. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 13 May, at 3:30 p.m. in Salon F. New Fellows will be introduced and ASA Prizes and Awards will be presented.

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

13. ANNUAL MEETING OF MEMBERS

The annual meeting of members will be held on Wednesday, 13 May 2026, at 3:30 p.m. in Salon F.

14. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be held at the Philadelphia meeting.

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

15. COFFEE BREAKS

Morning coffee breaks will be held daily from 9:30 a.m. to 11:00 a.m. and an afternoon break will be held on Tuesday from 2:00 p.m. to 3:30 p.m. in Franklin A on the 4th floor.

16. A/V PREVIEW ROOM

The A/V preview room will be available on Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 12:00 noon in 405.

17. MOTHERS ROOM

A Mothers Room for ASA meeting attendees will be available Monday to Friday in Reg Desk I The hours are Monday to Thursday, 8:00 a.m. to 5:00 p.m. and Friday, 8:00 a.m. to 12:00 noon.

18. SOCIAL

A Social will be held on Wesmesdau evening, 6:00 p.m. to 7:30 p.m. in Salons G-H.

The ASA hosts this social hour 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.

Philadelphia is a great walking town, and the Philadelphia Downtown Marriott Hotel is only 6 blocks away from the Liberty Bell, Independence Hall, the National Constitution Center, and the Independence Visitors Center in the Independence National Historic Park. No ASA-only tours are available, but meeting participants are welcome to tour on their own during usual visitors hours of 9 am to 5 pm, see www.nps.gov. Note that these sites are not open in the evenings. Liberty Bell Center does not require tickets, but some tours of Independence Hall do. Further, note that security screening is routine at these sites.

19. 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, 11 May, from 5:00 p.m. to 5:45 p.m. in Salon E. This will be followed by the Student Meet and Greet from 5:45 p.m. to 7:30 p.m. Salon F where refreshments and a cash bar will be available.

The Students' Reception will be held on Tuesday, 12 May, from 6:00 p.m. to 8:00 p.m. in Salon G. This reception will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

20. WOMEN IN ACOUSTICS RECEPTION

The Women in Acoustics reception will be held on Monday, 11 May, from 7:00 p.m. to 8:00 p.m. in Salon G. The reception is open to all attendees and tickets are not required to attend. A cash bar and lite snacks will be available.

21. JAM SESSION

You are invited to Milkboy (1100 Chestnut St, Philadelphia, PA 19107) for the ASA Jam, 8:00 p.m. to 12:00 midnight. 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.

22. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Philadelphia meeting. The on-site registration fee for accompanying persons is USD $250. A hospitality room for accompanying persons will be open in 415, 8:30 a.m. to 11:30 a.m. Monday through Thursday.

Registration entitles you access to the accompanying persons room, the social on Tuesday, the Jam Session, and the Plenary Session on Wednesday afternoon and the open rehearsal by The Philadelphia Orchestra, all on the Wednesday. In addition, a representative of the Philadelphia Visitors Office will be available to answer questions Monday and Tuesday between 9:30 a.m. and 10:00 a.m., and touring information will be available in the accompanying persons room at other times.

23. ASA WAYFINDERS

The ASA Philadelphia Local Committee is pleased to introduce an ASA Wayfinders program for the Philadelphia meeting. These Wayfinders will be student volunteers who will help point ASA attendees towards registration, meeting rooms, poster sessions, socials, tours, and other events in and around the Philadelphia Downtown Marriott hotel. You can identify them by their blue 'Directions? Ask Me!' buttons. This volunteer activity is sponsored by ASA's Penn State Student Chapter.

24. LOCAL RAIL TO/ROM PHILADELPHIA INTERNATION AIRPORT

Conference attendees arriving at Philadelphia International Airport (PHL) should make their way, followings the signs, to SEPTA, Philadelphia's commuter rail system, which has stations at PHL. Any train goes into the city and payment can be made several ways, see below. Get off at Jefferson Station, and it is a short walk to the downtown Marriott.

Options for Purchasing Tickets at PHL:

SEPTA Key Card (Recommended): Purchase from kiosks at the Airport Terminal (baggage claim level, near SEPTA train access). It costs approximately $4.95, which can be refunded as travel credit if registered online.

Quick Trip Ticket: A single-ride ticket available at kiosks for immediate travel.

Contactless Payment: Tap your credit/debit card, or phone (Apple/Google/Samsung Pay) directly on the train conductor's device or turnstile.

Cash: You can pay the conductor on the train with cash, but this is usually more expensive than using a Key card or contactless payment.

25. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Philadelphia meeting will have a published proceeding 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.

26. TECHNICAL PROGRAM ORGANIZING COMMITTEE

E. Carr Everbach, Technical Program Chair; Shima Abadi, Acoustical Oceanography; Gisela Giardino, Animal Bioacoustics; Kaitlin Hunt, Ben Shafer, Architectural Acoustics; John Cormack, James Kwan, Biomedical Acoustics; Philip Kaufinger, Stephanie Konarski Computational Acoustics; John Buck, Jill Linz, Education in Acoustics; Ahmed Allan, Randall Williams, Engineering Acoustics; Gary Scavone, Musical Acoustics; Andrew Christian, Alexandra Loubeau, Ze Feng (Ted) Gan, Noise; Raphael Hermann, Joel Lonzaga, Physical Acoustics; Nirmal Srinivasan, Christopher Stecker, Psychological and Physiological Acoustics; William Jenkins, Jing Lu, Signal Processing in Acoustics; Meghan Clayards, Keiko Ishikawa, Natasha Warner, Speech Communication; Feruza Amirkulova, Colby Cushing, Nathan Geib, Structural Acoustics and Vibration; Shima Abadi, Underwater Acoustics; Chirag Gokani, Student Council.

27. MEETING ORGANIZING COMMITTEE

Victor W. Sparrow, Chair; E. Carr Everbach, Technical Program Chair; Felicia Doggett , JAM; Members of the Pennsylvania State University Student Chapter.

28. PHOTOGRAPHING AND RECORDING

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

29. ABSTRACT ERRATA

This meeting program is Part 2 of the April 2026 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.

30. GUIDELINES FOR ORAL PRESENTATIONS

Preparation of Visual Aids

- See the guidelines for computer projection in section 41 below.
- Allow at least one minute of your talk for each slide (e.g., PowerPoint). No more than 12 slides for a 15-minute talk (with 3 minutes for questions and answers).
- Minimize the number of lines of text on one visual aid. 12 lines of text should be a maximum. Include no more than 2 graphs/plots/figures on a single slide. Too little information is better than too much.
- Presentations should contain simple, legible text that is readable from the back of the room.
- Characters should be at least 0.25 inches (6.5 mm) in height to be legible when projected. A good rule of thumb is that text should be 20 point or larger (including labels in
- inserted graphics). Anything smaller is difficult to read.
- Make symbols at least 1/3 the height of a capital letter.

- For computer presentations, use all of the available screen area using landscape orientation with very thin margins. If your institutions logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide.
- Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- Presentation
- Organize your talk with introduction, body, and summary or conclusion. Include only ideas, results, and concepts that can be explained in the allotted time. Four elements to include are:
- Statement of research problem
- Research methodology
- Review of results
- Conclusions
- No more than 3–5 key points can be covered adequately in a 15-minute talk so keep it concise.
- Rehearse your talk so you can confidently deliver it in the allotted time. Session
- Chairs have been instructed to adhere to the time schedule and to stop your presentation if you run over.
- An A/V preview room will be available for viewing computer presentations before your session starts. It is advisable to preview your presentation because in most cases you will be asked to load your presentation onto a computer which may have different software or a different configuration from your own computer.
- Arrive early enough so that you can meet the session chair, load your presentation on the computer provided, and familiarize yourself with the microphone, computer slide controls, laser pointer, and other equipment that you will use during your presentation. There will be many presenters loading their materials just prior to the session so it is important that you check that all multi-media elements (e.g., sounds or videos) play accurately prior to the day of your session.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They will not 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.

31. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

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

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

Design and layout

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

Lettering and text

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

Visuals

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

Presentation

- Prepare a brief oral summary of your poster and short answers to questions in advance.
- The presentation should cover the key points of the poster so that the audience can understand the main findings. Further details of the work should be left for discussion after the initial poster presentation.
- It is recommended that authors practice their poster presentation in front of colleagues before the meeting. Authors should request feedback about the oral presentation as well as poster content and layout.
- Other suggestions
- You may wish to prepare reduced-size copies of the poster (e.g., 8 1/2 × 11 sheets) to distribute to interested audience members.

32. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; E-mail: asa@acousticalsociety.org

191st Meeting, joint meeting, Baltimore Maryland, 15-17 December 2026

192nd Meeting, 4th joint meeting with the European Acoustics Association, 21-25 June 2027, Brussels, Belgium

ANNUAL GIVING TO THE ACOUSTICAL SOCIETY FOUNDATION FUND – 2026

The Acoustical Society of America Foundation Board is deeply grateful for all contributions received in 2026. To help express this gratitude, the list of donors to the Foundation Fund is published for all donations received in 2026.

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Session 1aAAa

Architectural Acoustics: Sustainability in Architectural and Building Acoustics

Matthew Agostino, Cochair

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

Invited Papers

8:00

1aAAa1. IEQ and sustainability: An acoustical consultant's perspective of building design trends. Josh Thede (Acoust. Consulting, BrightTree Studios, 301 Brush Creek Rd., Warrendale, PA 15086, jthede@brighttreestudios.com)

Sustainable building design is a sprawling hot topic, acoustics contributes to environmental performance, occupant-centered outcomes, and economics of project budgets and lifecycle cost. This presentation offers an acoustical consultant's perspective on Indoor Environmental Quality (IEQ) within the people-focused sub-category of sustainable acoustics. Drawing from interdisciplinary conference observations (including GreenBuild, Healthcare Design Expo, ASHRAE IEQ, and ASHRAE Winter Conference) and research literature, the presentation examines how acoustics interacts with other IEQ parameters to influence comfort, health, well-being, and sustainability across building types. Discussion includes rating systems LEED version 5, WELL version 2, and movements promoting healthy buildings, high-performance buildings, and occupant satisfaction for indoor environments. This work positions IEQ focused acoustics and noise control as a complementary counterpart to material and energy focused sustainable acoustics topics.

8:20

1aAAa2. "What do we do when there aren't clear regulations?" Regulatory ambiguity in design: A framework for good neighbor development. Scott Hamilton (Stantec, 233 S Wacker Dr #5300, Chicago, IL 60606, scott.hamilton@stantec.com)

When noise regulations are incomplete, inconsistent, or entirely absent, projects that affect their neighbors are forced to answer a harder question: *What does it mean to be a "good neighbor?" This presentation looks across sectors; urban data centers, industrial facilities, renewable energy, and other infrastructure, to examine how acoustics consultants navigate that question when the code book does not provide clear direction. The session contrasts prescriptive rules (objective, "how-to") with descriptive expectations (subjective goals like "no disturbance") and shows how each can fail when applied as a one-size-fits-all solution. Through past experiences, it explores situations where ordinances may be too vague, too narrow, or simply misaligned with local expectations, raising recurring questions: *Is the ordinance enough? Is it too much? Is there even a relevant standard?* Rather than offering a single, neat definition of "good neighbor," the presentation frames it as a moving target shaped by physical constraints, political context, and people issues, particularly trust, perception, and lived experience. It will provide a set of questions and strategies, but not a formula: tools for navigating the ambiguity where one size fits none and a "good neighbor" strategy is less an answer than an ongoing negotiation.

8:40

1aAAa3. Sustainability and acoustics: The common sense part. David S. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

In the wheel of sustainability, there are many topics to consider and it can be difficult to explain how all the facets provide direct benefit the owners/developers of buildings. Often when sustainability and acoustics are discussed in regard to buildings, the conversation turns to the very specific and limited performance requirements for indoor environmental quality standards. A more fundamental condition of sustainability is that a building is considered valuable as a functional or high performing building; it is desirable for occupancy as it fulfills the needs of its tenants. In acoustics this functionality can mean communication, privacy, or quiet among other criteria. The information in this paper may be useful in explaining the value of acoustics to architects, owners, and laypeople in regard to the value of a building, and how acoustics differs from other engineering disciplines.

9:00

1aAAa4. An acoustical-sustainability design framework for including embodied carbon of building materials. Arthur W. van der Harten (Acoust., TLC Eng./Open Res. in Acoust. Sci. and Education, 400 Main St., Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com) and Jonathan M. Broyles (Climate and Sustainability Consortium, Massachusetts Inst. of Technol., Broomfield, CO)

At the Spring 2025 meeting of the Acoustical Society of America (ASA), an addition to Pachyderm Acoustic was demonstrated to calculate the embodied carbon of acoustical materials, as well as the acoustic performance of finish systems. In addition to advocating for hearing comfort and noise control (noble goals in their own right), acousticians should be aware of their own role in mitigating global

warming and climate change through minimizing carbon emissions attributed to different building acoustic materials. This paper will demonstrate an acoustic modeling framework in which acousticians can be mindful of the environmental and health impacts of the materials that they specify. Using the open-source tool Pachyderm Acoustic as the modeling and simulation tool, and the open-source material information dataset by Broyles, it is possible to evaluate decisions not only in terms of their acoustical impact, but also in terms of their environmental impact. This paper will present case studies (e.g., a classroom), and a multi-objective design process to evaluate decisions not only in terms of their acoustical impact, but with an aim to reduce overall environmental impact.

9:20

1aAAa5. Decarbonizing acoustic performance: A comparative life cycle assessment of mass timber floor assemblies with dry subfloor systems. Pelin Tezcan (Pliteq, 4211 Yonge St., Toronto, ON M2P 2A9, Canada, ptezcan@pliteq.com), Simon Cleghorn (Tech. Services, Pliteq, Toronto, ON, Canada), and Aedan Callaghan (Pliteq Inc., Toronto, ON, Canada)

As mass timber projects push the envelope of low-embodied carbon construction and design for disassembly, challenges associated with acoustic floor assemblies have become a critical barrier to full building decarbonization. This work aims to use the embodied carbon calculation tool built into the Branch3D Software, analyzing both the Membertou Office Building in Cape Bretton as well as the Yale Divinity School's New Living Village. The embodied carbon footprint was calculated for a dry assembly in lieu of a business as usual concrete topping base case. As the largest university housing complex to pursue Living Building Challenge certification, the Yale Divinity School project replaced non-structural cementitious topping with a dry subfloor system utilizing GenieBoard with 100% recycled content. Cementitious materials are often used to increase mass to the floor-ceiling assemblies. This has a high contribution to the overall global warming potential (GWP) of the building. This study evaluates the GWP savings when incorporating recycled and bio-based materials into acoustic assemblies, while also achieving the project acoustic requirements without the use of wet-trade materials. The results provide a framework for selecting acoustic treatments that align with carbon-neutrality goals without compromising occupant comfort.

9:40–9:55 Break

9:55

1aAAa6. Quantifying trade-offs between acoustic absorption and embodied carbon in interior treatments. Milena J. Bem (School of Architecture, Rensselaer Polytechnic Inst., 1605 Hutton St. Apt 10, Troy, NY 12180, jonasm@rpi.edu) and Kristen Murphy (Acentech, Cambridge, MA)

The Architecture, Engineering, and Construction (AEC) sector contributes roughly 37% of global carbon emissions, with embodied carbon from building materials representing a significant share. One way acousticians can influence embodied carbon emissions is through thoughtful selection of acoustic treatments, yet the carbon implications of material choices are an often-overlooked design factor, and little is currently known about how to quantify and compare design decisions from an acoustical and embodied carbon perspective. This study provides a proposed framework to compare acoustical performance and embodied carbon using a Life Cycle Assessment (LCA) framework for reverberation control. First, this study shows the results of the calculated Global Warming Potential (GWP, A1–A3) of several common surface-applied acoustically absorptive materials (such as fabric-wrapped panels, acoustical ceiling tiles, *etc.*), compiled from available Environmental Product Declarations (EPDs) and verified databases. Second, an LCA was conducted of several theoretical retrofit scenarios applied to a museum gallery case study. The different scenarios represent different quantities and combinations of acoustically absorptive treatments and compare the design's A1–A3 emissions and resulting reverberation time (RT). With this example, this study aims to find strategies to help acousticians make decisions to optimize a design's carbon footprint and acoustic performance.

10:15

1aAAa7. Beyond design day: Quantifying acoustic exceedances and sustainability trade-offs in building systems noise control. Ben Loshin (Arup, 1191 Second Ave., Ste. 400, Seattle, WA 98101, ben.loshin@arup.com)

Project teams often design interior HVAC noise control to meet "Design Day" peak-load criteria, even though buildings seldom reach those extremes. This can yield over-engineered solutions with avoidable financial and environmental costs. Drawing on data collected while authoring office design standards for corporate clients, this study examines how sustainability-led design can drive portfolio-wide impact. First, we quantify the frequency and duration of acoustic exceedances under partial-load operation using operational building data. Second, we integrate embodied and operational carbon to compare conventional "always meet the limit" approaches with strategies optimized for typical operation. Third, we bridge technical metrics and human experience through calibrated spatial audio listening sessions in the Arup SoundLab, gauging occupant tolerance for occasional exceedances. The outcome is a practical framework that balances acoustic comfort, cost, and carbon in HVAC noise control.

10:35

1aAAa8. Acoustic barriers to mixed-mode ventilation in U.S. sustainability frameworks. Kamar Godoy (Intertek, 11 Broadway, Ste. 410, New York, NY 10004, kamar.godoy@intertek.com)

Mixed-mode ventilation has gained renewed attention in U.S. sustainable building design as a strategy to reduce energy consumption, improve occupant comfort, and align with decarbonization goals. However, there is a disconnect between air quality-based mixed-mode criteria and exterior acoustic constraints within current U.S. sustainability frameworks, such as LEED and WELL. A comparative energy sensitivity analysis is presented to illustrate the potential operational and energy impacts of mixed-mode ventilation under realistic acoustic limitations. The findings highlight a gap in current incentive structures, wherein exterior noise acts as an unaccounted barrier to mixed-mode adoption. The paper concludes by discussing opportunities for integrating exterior acoustic considerations into future sustainability standards to better align energy, health, and environmental performance goals.

10:55

1aAAa9. Sustainable pathways towards a healthy built environment with acoustic ventilation metamaterials. Caner Baydur (College of Design and Innovation, Tongji Univ., Siping Rd., No. 1239, Shanghai 200092, China, canerbaydur@tongji.edu.cn), Xu Wang (Inst. of Acoust., School of Phys. Sci. and Eng., Tongji Univ., Shanghai, China), Yongqi Lou (College of Design and Innovation, Tongji Univ., Shanghai, China), Jian Kang (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom), and Nicholas Fang (Dept. of Mech. Eng., Univ. of Hong Kong, Hong Kong, China)

The promise of overcoming the limitations of conventional materials by simultaneous noise reduction and air circulation using artificial acoustic ventilation metamaterials (AVMs) has attracted great attention in the physics community. However, the implementations into the built environment are currently limited. To address this gap, we conducted a systematic review of 108 articles on AVMs. Our analysis reveals that research has predominantly focused on acoustic and ventilation performance (technological dimension), while neglecting environmental, economic, and social dimensions. We then discuss the built environment-oriented challenges, demonstrating that AVMs have not yet matured to meet practical demands. Finally, we introduce our holistic design approach to AVMs for sustainable implementations. We believe that AVMs can pave the way for unique avenues for sustainability. Nonetheless, interdisciplinary collaboration among physics, engineering, and social sciences remains essential. [Work supported by the National Key R&D Program of China, the National Science Foundation of China, and the Fundamental Research Funds for the Central Universities.]

11:15

1aAAa10. Managing acoustic-ventilation tradeoffs in a high-performance sustainable office: A design collaboration. Arthur W. van der Harten (Acoust., Acoust. Distinctions/Open Res. in Acoust. Sci. and Education, 400 Main St., Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com), Stefano Capra (Ramboll, Copenhagen, Denmark), and Irene Gallou (Specialist Modelling Group, Foster + Partners, London, United Kingdom)

The Bloomberg London headquarters, designed by Foster + Partners and built by Sir Robert McAlpine, espouses some of the world's loftiest sustainability goals for a large office building (more than 4000 employees). It is said to have achieved a 98.5% score against BREEAM sustainability assessment method and was the winner of the 2018 RIBA Stirling Prize for architecture. The design effort required mechanical engineering, aeronautical engineering, and acoustical engineering in order to meet the very stringent ventilation criteria while controling for environmental noise ingress to the office space. This talk will cover the design and implementation of the natural ventilation system and discuss the delicate multi-disciplinary collaboration that was required in order to operate the natural ventilation effectively while still meeting requirements for background noise in an open office space in the heart of London.

MONDAY MORNING, 11 MAY 2026 SALON B, 10:30 A.M. TO 12:00 NOON

Session 1aAAb

Architectural Acoustics: New Challenges in Stage Acoustics Design

Francesco Martellotta, Cochair
Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy

Brandon Cudequest, Cochair
Threshold Acoust., 141 W Jackson Blvd Ste. 2080, Chicago, IL 60604

Chair's Introduction—10:30

Invited Papers

10:35

1aAAb1. What the orchestra needs and how acousticians can help? Brandon Cudequest (Threshold Acoust., 141 W Jackson Blvd Ste. 2080, Chicago, IL 60604, bcudequest@thresholdacoustics.com)

Historically, concert hall design has been audience focused. Less attention has been given to the orchestras that perform in these halls, and what conditions allow them to thrive when playing in these rooms on a regular basis. The presentation will start with general trends in orchestra sizes and repertoire and how that drives stage layouts. The needs of the orchestra and the production staff must be

considered in the design of orchestra shells and stage risers. The presentation will also discuss the challenges of producing larger works and how most halls have makeshift or retrofitted stage extensions to accomplish this. Acousticians can offer wholistic solutions that benefit both the orchestra and the audience if they are aware of the production challenges that orchestras regularly face.

10:55

1aAAb2. A comparative analysis of stage support parameters measured in different venues. Francesco Martellotta (Dept. of Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy, francesco.martellotta@poliba.it) and Dario D'Orazio (D'Orazio Acoust., Bologna, Italy)

Since their first appearance in the architectural acoustics field, stage support parameters slowly gained popularity but, at the same time, underwent small changes that slightly modified their definition (in particular to better fit hearing oneself and hearing others), while other parameters were proposed to better explore the specific acoustic features of stages. In fact, a limited, but significant, number of subjective studies pointed out that different parameters could better discriminate stage configurations while proving more robust to small stage differences and showing improved agreement with subjective preferences that pointed out the importance of the reverberant response of the main hall. The analysis of the measurements collected in a number of different venues is presented and discussed, showing that, as already pointed out by other authors, significant correlations appear among the different parameters, while comparable values (including uncertainties) correspond to very different configurations, suggesting that further investigations are required in this field, also involving subjective ratings, so to clarify which parameters could more reliably help acousticians in designing optimal arrangements for performers.

11:15

1aAAb3. Four decades of stage shells, a stroll through time and what's to come. Zachery O. L'Italien (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, zlitalien@mchinc.com), David A. Conant (McKay Conant Hoover Inc., Westlake Village, CA), and Jack P. Hagler (Schuler Shook, Dallas, TX)

This presentation recollects four decades of shell design, including notable features, progression of design approaches, and recent challenges. Commentary will address various acoustical aspects, such as performer comfort, on-stage energy distribution, and projection to audience, as well as various theatrical aspects, such as lighting, rigging, ventilation, storage, and operation.

Contributed Paper

11:35

1aAAb4. The influence of acoustic parameters and music from different periods for overall preference of concert hall acoustics. Fernando M. del Solar Dorrego (Acoust., Penn State, 445 Waupelani Dr., Apt. B10, State College, PA 16801, fsolar@gmail.com) and Michelle C. Vigeant (Acoust., Penn State, University Park, PA)

The study of the factors that dictate the overall preference for the acoustics of a performance venue has been undertaken by many researchers in the past. The purpose of the present study was to investigate the strength of the correlations between the acoustical parameters defined in the Annex A of the ISO 3382-1 standard, and acoustic preference, using higher order ambisonics reproductions. A perceptual study was undertaken in which participants had to rate a set of stimuli in terms of overall preference using a paired comparison and rating paradigm. The stimuli were obtained through the convolution of eight spatial room impulse responses (SRIRs) from eight different halls, with three anechoic musical excerpts, which are representative of the baroque, classical, and romantic musical styles. Significant correlations were obtained between preference and specific octave-band values of acoustic parameters, which provide some indication of the factors that influence acoustic preference for different musical styles. An additional objective of this investigation was to determine whether different preference groups exist. The statistical method of latent class analysis revealed that two preference groups of listeners exist within the sample, one favoring a more reverberant acoustic than the other.

11:50–12:00 Panel Discussion

Session 1aAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Effects of Hearing Loss and Abnormal Acoustic Experience on Cognitive and Affective Behavior in Animals and Humans

Amanda M. Lauer, Cochair
Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 521 Traylor Res. Bldg., 720 Rutland Ave., Balitmore, MD 21205

Micheal Dent, Cochair
Univ. at Buffalo, SUNY, B76 Park Hall, University of Buffalo-SUNY, Buffalo, NY 142260

Aravind Parthasarathy, Cochair
Univ. of Pittsburgh, 1414 Barnsdale St., Pittsburgh, PA 15217

Lina A. Reiss, Cochair
Oregon Health and Science Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239

Chair's Introduction—7:55

Invited Papers

8:00

1aAB1. Noise-induced hearing loss disrupts sensory-to-cognitive transformations in cortical decision-making networks. Justin D. Yao (Head and Neck Surgery & Commun. Sci., Rutgers Univ., 604 Allison Rd., Nelson Biology Lab., RM D418, Piscataway, NJ 08854, justin.yao@rutgers.edu)

Noise-induced hearing loss (NIHL) is a leading cause of sensory impairment resulting from exposure to loud noise and is accompanied by deficits in speech comprehension and cognitive function. However, the neural mechanisms linking NIHL to cognitive dysfunction remain unclear. Here, we examine how NIHL alters cortical representations supporting auditory decision-making by comparing neural activity in auditory cortex (AC) and parietal cortex (PC) under normal-hearing and NIHL conditions. Gerbils were trained to discriminate slow versus fast presentation rates of amplitude-modulated (AM) broadband noise while extracellular activity from cortical neurons were recorded during task engagement and passive listening. Permanent NIHL was induced *via* controlled broadband noise exposure, and post-NIHL stimuli were presented at matched sensation levels to pre-NIHL conditions to isolate central effects of NIHL. After NIHL, auditory decision-making was impaired, AC decoding accuracy of AM categorical representations during task engagement was reduced, and task-dependent modulation of AC was disrupted. In PC, task-related representations closely reflected behavioral performance under normal-hearing, but these representations were degraded after NIHL. Together, these findings suggest that NIHL disrupts the transformation of sensory representations into cognitive decision variables across cortical networks, providing a circuit-level account of NIHL-related cognitive impairment.

8:20

1aAB2. Neural correlates of binaural fusion. Lina A. Reiss (Oregon Health and Sci. Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239, reiss@ohsu.edu), Miles Carter (Univ. of Pittsburgh, Pittsburgh, PA), Michela Mondesir, and Stephen David (Oregon Health and Sci. Univ., Portland, OR)

Binaural fusion is the perception of different sounds across the two ears as a single auditory object. Many individuals with hearing loss experience abnormally broad binaural fusion, fusing tones differing by as much as 3–4 octaves between ears. Such broad binaural fusion leads to blending of speech from different voices and difficulties with understanding speech in background talkers. Understanding of mechanism is essential for treatment of broad binaural fusion. Here, we describe preliminary investigation of the neural correlates of binaural fusion in ferrets, using two novel illusions. The first illusion is the perception of the vowel /ae/ when the vowels /ah/ and /ee/ are presented dichotically at the same fundamental frequency, due to binaural fusion and perceptual averaging of the vowel spectra (Reiss and Molis, 2021). The second illusion is the perception of a single amplitude-modulated (AM) tone with binaural fusion of an AM and un-modulated tone. Results indicate perception of similar illusions in humans and ferrets, motivating studies of neural correlates in animals. [Work funded by the LaCroute Foundation, and NIH/NIDCD Grant Nos. R25GM134978, R01DC013307, and R01DC014950.]

8:40

1aAB3. Silence, solitude, and serotonin: Effects of noise exposure and social isolation on the serotonergic system. Megan M. Walker, Jessica M. Bush, Lauren M. Hadland (Biology, Indiana Univ., Bloomington, IN), and Laura M. Hurley (Biology, Indiana Univ., 1001 E 3rd St., Bloomington, IN 47405-7005, lhurley@iu.edu)

Hearing loss and social isolation can co-occur in older adults and may be associated with affective and depressive disorders. The serotonergic system is sensitive to hearing loss and the social environment, so could link these factors. To assess whether hearing loss and isolation affect behavior and the serotonergic system in older female mice, we exposed mice to noise or sham at 6 weeks of age. At 11 months of age, females were isolated or socially housed for 4 weeks, followed by measurement of vocal and nonvocal social behavior, behavior in the open field, serotonergic innervation of the auditory midbrain, and c-Fos expression by serotonergic neurons. Social and open field behavior, but not the density of serotonergic axons, were sensitive to social isolation. c-Fos expression by serotonergic neurons was increased in the noise-exposed group, but only in socially isolated individuals. c-Fos label in isolated individuals also showed a weak but significant correlation with hearing threshold and USVs during social interaction. These findings suggest that some aspects of the serotonergic system show less plasticity in older adults, and that noise exposure and social isolation interact to affect serotonergic function. [Work funded by a grant from the Royal National Institute for Deaf People.]

9:00

1aAB4. Tracking sustained effects of adverse listening conditions using resting electroencephalography. Cynthia Hunter (Speech-Language-Hearing, Univ. of Kansas, 1000 Sunnyside Ave., DHDC 3000, Lawrence, KS 66045, c.hunter@ku.edu)

Adverse listening conditions, such as those created by noisy environments or hearing loss, alter how the brain listens, for example requiring greater listening effort in order to comprehend speech. Sustained effort can lead to fatigue, and people with hearing loss commonly report that listening is fatiguing. Moreover, hearing loss is a risk factor for dementia, suggesting long-term links between hearing loss and brain function. This talk will describe two studies in which resting electroencephalography (EEG) was used to identify sustained effects of adverse listening conditions on brain function. In the first study, alpha (8–13 Hz) and theta (4–7 Hz) power and alpha peak frequency were measured in older adults ($n = 41$) before and after completing a challenging speech-in-noise task. Subjective fatigue and alpha power increased and alpha peak frequency decreased post-task, indicating that alpha power and peak frequency can track the development of cognitive fatigue induced by challenging listening scenarios. In a second, exploratory study with a sample of middle-aged and older adults ($n = 62$), greater degree of hearing loss was associated with lower pre-task oscillatory power in the alpha and theta bands, suggesting that reduced power of these neural rhythms may indicate changes to brain function associated with hearing loss.

9:20

1aAB5. Early loud noise exposure and background noise shape cognitive and exploratory behavior in mice. Sergio Vicencio-Jimenez (OHNS, Johns Hopkins School of Medicine, 720 Rutland Ave., Traylor Bldg. 515, Baltimore, MD 21205, svicenc1@jh.edu) and Amanda M. Lauer (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, Balitmore, MD)

Acute loud noise exposure early in life may have lasting effects on behavior. Additionally, hearing in noise is often among the first functions reported by patients, suggesting that background noise can reveal vulnerabilities that are otherwise difficult to detect. Here, we examined how early traumatic noise exposure and background noise influence behavior across the lifespan in mice. Animals were tested in behavioral tasks assessing exploration (open field), memory (Y-maze and object recognition), and decision-making under effort (effortful Y-maze). All testing was conducted under constant dim lighting ($\sim$50 lux) in an acoustically shielded room, using two background noise conditions: low ambient noise ($\sim$30 dB SPL) and moderate broadband noise ($\sim$60 dB SPL). Across tasks, behavioral performance differed significantly between noise conditions, with moderate noise showing behavioral deficits not found in quiet conditions. Moreover, with aging mice that experienced acute noise exposure at a young age showed worse performance than age-matched controls across all behavioral measures. These findings show that both early acoustic experience and auditory context shape behavioral outcomes. Finally, a brief survey of the literature suggests that sound testing conditions are rarely reported in mice behavioral studies, highlighting the need to treat sound as a biologically meaningful experimental variable.

9:40

1aAB6. Effect of asymmetric hearing on spatial selective attention. Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., 0141 Lefrak Hall, College Park, MD 20742, goupell@umd.edu) and Joshua G. Bernstein (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

More people are receiving cochlear implant (CIs) to partially restore hearing to both ears, either in cases of bilateral (BI-CI) or unilateral hearing loss (single-sided deafness, SSD-CI). Compared to monaural listening, this usually improves spatial-hearing abilities like sound localization and spatial release from masking (increased speech recognition for a target talker spatially separated from a masker). Yet spatial-hearing benefits with CIs are smaller than with acoustic hearing in both ears, and they appear to diminish with increasing functional asymmetry. We hypothesize that diminished spatial-hearing benefits are partly attributable to non-auditory-exclusive selective-attention abilities, whereby some individuals exhibit particular difficulty attending to a degraded target source in the presence of less degraded interfering source. We present data from several experiments in real and simulated human BI-CI and SSD-CI listeners that support this hypothesis. Measurements include behavioral binaural unmasking of speech and dichotic listening, pupillometry (an objective index of listening effort), and non-auditory cognitive tests. The results suggest that reducing functional asymmetry through novel programming approaches or training might improve spatial-hearing benefits with CIs. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy of the Department of War or U.S. Government.]

10:00–10:15 Break

10:15

1aAB7. Neuronal correlates underlying auditory categorization in mice. Anjali Sinha (Otorhinolaryngology, Univ. of Pennsylvania, Stemmler Hall G10, 3450 Hamilton Walk, Philadelphia, PA 19104, anjalikumari.sinha@pennmedicine.upenn.edu), Mark Aizenberg, and Maria Geffen (Otorhinolaryngology, Univ. of Pennsylvania, Philadelphia, PA)

Categorization of sensory stimuli is a fundamental computation in cognition and is critical for many aspects of learning, perception, and prediction. Auditory categorization has been linked to the neuronal activity in the auditory cortex (AC), however, it remains poorly understood how categorical representations emerge in population neuronal activity with learning. We recently showed that during categorization learning, mice use multiple complementary behavioral strategies (Collina *et al*., 2025). To understand neural correlates of auditory categorization, we recorded activity from AC of mice implanted with neuropixel electrodes while training them to perform auditory categorization task using two-alternative forced choice task. Approximately 30% of the recorded neurons exhibited category preference. Additionally, population decoding accuracy to categories increased with training. Training increased alignment between neurometric and psychometric thresholds, suggesting that neural activity reflected behavior. Moreover, a subpopulation of neurons was choice-selective, as their firing rate reflected the choice of the mouse across trials. Additionally, we also identified components in the population responses that signaled reward prediction and reward response in AC. Our results show that neuronal activity in the AC during categorization reflects behavioral performance. We are currently analyzing how the neuronal population activity evolves throughout auditory categorization learning.

10:35

1aAB8. Effects of hearing impairment on behavior in military operations. Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.s.brungart.civ@mail.mil), Gregory M. Ellis, and Benjamin Sheffield (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., Bethesda, MD)

Hearing is a critical sensory modality in military environments, where service members rely heavily on auditory cues to identify threats, coordinate movement, and exploit tactical opportunities on the battlefield. However, the effects of hearing impairment on behavior and performance during military tasks are complex and not always straightforward. In this study, West Point cadets wearing paintball gear were tasked with eliminating opponents while using a hearing loss simulator that systematically degraded auditory ability. It was initially hypothesized that participants with degraded hearing would adopt similar strategies but perform more poorly overall than those with normal hearing. Contrary to expectations, participants with severe hearing loss altered their tactics, shifting from active pursuit to a defensive, concealed strategy. Although this resulted in fewer opponents eliminated, it often improved their overall survival rate. In contrast, when hearing loss impaired sound localization rather than detection, performance declined substantially, likely because participants were unaware of the nature of their impairment. These findings have important implications for evaluating fitness-for-duty among hearing-impaired personnel in high-risk operational roles. [The views expressed in this presentation are those of the authors and do not necessarily reflect the official policy of the Department of War or the U.S. Government.]

10:55

1aAB9. Musical reward in hearing loss—Beyond perception to cognitive and affective factors. Alexander Chern (Otorhinolaryngology-Head & Neck Surgery, 19104, Philadelphia, PA, Alexander.Chern@PennMedicine.upenn.edu), Emily Gao, Joon Soo Kim, Celine Arar (Otolaryngol., Johns Hopkins Univ. School of Medicine, Baltimore, MD), Claire Wang (Johns Hopkins Univ. School of Nursing, Baltimore, MD), Maria Armache, Elaine Thompson (Otolaryngol., Johns Hopkins Univ. School of Medicine, Baltimore, MD), Marshall Chasin (Musicians' Clinics of Canada, Toronto, ON, Canada), Reyna Gordon (Otolaryngology-Head and Neck Surgery, Vanderbilt Univ. Medical Ctr., Nashville, TN), and Theodore McRackan (Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Current music-specific outcome measures in hearing loss (HL) research emphasize perceptual abilities while neglecting emotional, cognitive, and social dimensions of musical reward. We aimed to develop a patient-reported outcome measure (PROM) capturing the multidimensional nature of musical enjoyment across the hearing spectrum. Forty-seven participants across the hearing spectrum (ages 18–89; normal hearing, unaided HL, hearing aid users, and cochlear implant users) completed semi-structured interviews exploring music experiences per PROMIS guidelines. An abductive approach was used to refine a conceptual framework of musical reward. Two independent reviewers coded transcripts to identify key domains. An initial item pool was generated and refined through cognitive interviews with 16 additional participants. Five hearing-related domains emerged: music perception, preference, sound quality, listening environment, and social context. These factors influenced both passive and active music engagement pathways, which shaped emotional responses and mood regulation, ultimately moderating musical reward. An initial 88-item pool was developed representing these multidimensional domains. Musical reward in HL extends beyond auditory perception to encompass emotional, cognitive, and social factors. This multidimensional, HL-specific framework challenges perception-focused approaches and establishes a foundation for a PROM that can inform rehabilitation strategies and measure meaningful outcomes. Musical reward in HL extends beyond auditory perception to encompass emotional, cognitive, and social factors. This multidimensional, HL-specific framework challenges perception-focused approaches and establishes a foundation for a PROM that can inform rehabilitation strategies and measure meaningful outcomes.

11:15

1aAB10. Hearing severity implications on music perception, affective responses, and real-world environmental listening preferences. Emily Gao (Otolaryngol., Johns Hopkins Univ. School of Medicine, 1800 Orleans St., Baltimore, MD 21287, egao7@jh.edu), Maria Armache, Joon Soo Kim, Celine Arar, Elaine Thompson (Otolaryngol., Johns Hopkins Univ. School of Medicine, Baltimore, MD), Claire Wang (Johns Hopkins Univ. School of Nursing, Baltimore, MD), Marshall Chasin (Western Univ., Toronto, ON, Canada), Theodore McRackan (Otolaryngol., Medical Univ. of South Carolina, Charleston, SC), and Alexander Chern (Otorhinolaryngol., Hospital of the Univ. of Pennsylvania, Philadelphia, PA)

Music listening is a complex auditory task with cognitive and emotional benefits sensitive to hearing loss (HL), motivating adaptive environmental preferences. Adults ($\geq$18 years) across hearing spectrum (normal hearing, unaided HL, and hearing aid) completed the Barcelona Musical Reward Questionnaire, Goldsmiths Musical Sophistication Index, and select University of Canterbury Music Listening Questionnaire items (December 2024–January 2026). Associations between better-ear pure-tone average (PTAB) with music perception (pitch, rhythm, and song recognition) and affective responses (emotion, mood, and social) were examined using multivariable linear regression, while associations with environmental preferences were examined using multivariable logistic regression. Among 123 participants (mean age 58.2 years; mean PTAB 39.2 decibel hearing level [dB HL]; 58% female; and 59% aided), worse hearing (per + 20 dB HL) was associated with reduced pitch ($b = -6.7\%$, $p < 0.001$), rhythm ($b = -4.8\%$, $p = 0.04$), and song recognition ($b = -3.9\%$, $p = 0.03$) but not decreased affective responses. Worse PTAB was associated with reduced enjoyment at soft volumes (OR = 0.52, $p = 0.01$) and trended toward reduced enjoyment in reverberant environments or with high quality recordings, and greater enjoyment in quiet environments or when sitting near the front. HL may impair music perception while affective responses remain preserved potentially through adaptive environmental preferences, highlighting personalized counseling opportunities.

11:30

1aAB11. Perception of speech in noise and under memory load: Assessing the role of diurnal preferences in performance. Miranda Cleary (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., Ste. 0100 Lefrak Hall, College Park, MD 20742, micleary@umd.edu), Gerald Tabe, Cesmani Blango, Samira Anderson, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD)

As humans age, circadian/diurnal preferences typically shift to earlier times of day. Recent research identifies the fatigue associated with listening effort as a fundamental concern among listeners with hearing impairment. The impact of time-of-day on perceiving speech under challenging listening conditions is, however, not well established for older listeners with hearing loss. A study was therefore initiated to measure time-of-day effects. Experienced cochlear-implant users with postlingual onsets of hearing loss completed several effortful perceptual tasks either in the morning, mid-afternoon, or at both timepoints. Sentence pairs were presented in quiet listening conditions and listeners were cued either before or after presentation, to repeat the first, second, or both sentences. Speech-in-noise (+10 SNR) scores for single sentences were also recorded. Following this, information was collected regarding time-of-day preferences, sleep habits, and fatigue, along with standardized measures of visual attention, processing speed, and working memory. Preliminary data do not currently bear out participants' intuitions regarding performance in the morning versus afternoon. Although memory load and target uncertainty reduce scores, performance across timepoints appears similar, irrespective of time-of-day-preference. Additional data for speech-in-noise will be reported.

11:45

1aAB12. Somatostatin interneurons drive contrast gain control in mouse primary auditory cortex. Omer Zeliger (Dept. of Otorhinolaryngology, Univ. of Pennsylvania, 3450 Hamilton Walk, Philadelphia, PA 19104, omer.zeliger@gmail.com), Christopher Angeloni (Dept. of Neurobiology, Northwestern, Evanston, IL), Valerie Baubet, and Maria Geffen (Dept. of Otorhinolaryngology, Univ. of Pennsylvania, Philadelphia, PA)

In the natural environment, the auditory system faces the challenge of separating signal from noisy background. One method the central auditory system uses to address this challenge is to adapt to the characteristics of the current environment. In particular, neurons in the central auditory system adapt their gain in response to changes in sound level variability, known as contrast. Though contrast gain adaptation has been demonstrated across multiple brain regions in many different organisms, the circuit mechanisms of this adaptation are still unknown. In this study, I test the hypothesis that local inhibition drives contrast gain adaptation in the primary auditory cortex (A1). In order to test how local inhibition impacts neuronal response functions, I performed acute recordings in A1 of mice while optogenetically suppressing the activity of two genetic classes of interneurons: parvalbumin-positive (PV) interneurons and somatostatin-positive (SST) interneurons. In addition, to test how local inhibition contributes to functional outcomes, I tested how interneuron activity impacts perceptual sensitivity in environments with changing contrasts. In both experiments, I found that suppressing SST, but not PV, neurons reduced the degree of gain adaptation across stimuli with different contrasts, indicating that SST interneurons drive contrast gain adaptation in the mouse A1.

Session 1aAOa

Acoustical Oceanography, Underwater Acoustics, Signal Processing in Acoustics, and Animal Bioacoustics: Acoustic Observations from Uncrewed Platforms I

Scott Loranger, Cochair
Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 86 Water St., Woods Hole, MA 02543

Elizabeth Weidner, Cochair
Marine Sci., Univ. of Connecticut, 1080 Shennecossett Rd., Groton, CT 06340

Derek Olson, Cochair
Oceanography, Naval Postgraduate School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943

Invited Papers

8:30

1aAOa1. Verfied directed sampling: A new paradigm for mid-water exploration. Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, larry.mayer@unh.edu), Val Schmidt (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), Dana Yoerger, Annette Govindarajan (Woods Hole Oceanogr. Inst., Woods Hole, MA), Roland Arsenault (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH), Eric Hayden, and Nina Yang (Woods Hole Oceanogr. Inst., Woods Hole, MA)

Acoustic techniques have been used for years to derive critical fisheries, biological, and oceanographic information in the water column, but with the difficulty of knowing what, precisely, was ensonified. Similarly, eDNA measurements are quickly becoming a standard oceanographic tool to identify the species distribution and make biomass estimates by sampling at standard depths. However, these sampling intervals are typically selected without a-priori knowledge of the water column. To address these shortcomings, we have combined the operation of the Mesobot autonomous underwater vehicle and Oceanic-WHOI autonomous eDNA samplers with a DriX Uncrewed Surface Vessel equipped with an EK80 fisheries echosounder. A USBL system provides vehicle tracking and communication allowing remotely loctated operators to see, in real-time, the position of the vehicle with respect to acoustic targets in the water column. Using acoustic commands the Mesobot can be directed to specific acoustic targets and sampling systems (eDNA samplers, high-resolution cameras, *etc.*) can be turned on and off with complete knowledge of the position of the sample with respect to the acoustic target. This "Verified Directed Sampling" has proven very successful as eDNA results and the review of videos from targeted acoustic layers, give unambiguous information about the makeup of the layers, and estimates of biomass and species distribution retrieved from directed samples have shown much higher abundances than samples taken blindly at standardized depths.

8:50

1aAOa2. Integrated active acoustic observations from deep-ocean uncrewed platforms: Multi-scale mapping and real-time adaptive survey capabilities. Isaac Vandor (National Deep Submergence Facility, Woods Hole Oceanogr. Inst., Woods Hole, MA) and Christina Haskins (National Deep Submergence Facility, Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, haskins@whoi.edu)

The National Deep Submergence Facility's AUV Sentry and ROV Jason provide complementary active acoustic capabilities for oceanography at 6000 + meter depths. Sentry's multi-frequency acoustic suite (400–700 kHz multibeam, 120/230/540 kHz sidescan sonar, and 4–24 kHz sub-bottom profiler) achieves meter-scale seafloor resolution. ROV Jason enables sustained surveys with concurrent visual verification and sampling. Recent deployments demonstrate supervised autonomy enabled by acoustic imaging and communications. During Juan de Fuca Ridge operations, Sentry's multibeam acoustic backscatter detected hydrothermal plume structures while *in situ* chemical sensors (ORP, optical backscatter, methane) confirmed geochemical signatures. Vehicle-to-surface telemetry using the WHOI Micro-Modem and USBL positioning enabled real-time mission re-tasking. This workflow represents a flexible mission framework that dynamically shifts between fully autonomous, human-supervised, and direct intervention based on real-time acoustic observations. Multi-frequency acoustic observations with sub-10 m positioning accuracy enable integrated characterization of deep-ocean environments. This enables informed real-time decisions and meter-scale feature correlation across missions. The integration of comprehensive acoustic environmental sensing, precise navigation, and high-bandwidth/low-latency communications enable adaptive autonomy, expanding scientific capability while preserving safe and reliable deep-ocean operations.

9:10

1aAOa3. Impact of gyre-modulated sound-speed structure on acoustic arrivals measured with gliders in the Canada Basin. Luis O. Pomales Velázquez (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett Bay Campus 215 South Ferry Rd., Narragansett, RI 02882, luispomales@uri.edu), Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Two buoyancy-driven gliders recorded transmissions from an array of moored acoustic sources both outside and inside the Beaufort Gyre in the summers of 2016 and 2017 as part of the Canada Basin Acoustic Propagation/Glider Experiment. Concurrent observations of temperature and salinity provide a detailed look at the changes in the sound-speed environment within the array as the gliders crossed into the gyre. Additionally, observations made as the gliders navigated the gyre's inner periphery and towards its center reveal the sound-speed structure conditions that are modulated by the large-scale steady-state gyre dynamics. This work provides insight into the impact of the gyre's local structure on acoustic propagation by evaluating acoustic arrivals measured on the gliders in the context of the gyre-modulated sound-speed structure.

9:30

1aAOa4. How I learned to stop worrying and love uncalibrated sonar (on uncrewed platforms). Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu)

It is not uncommon to have side-looking sonar systems (*e.g.*, sidescan sonar or synthetic aperture sonar) mounted on autonomous underwater vehicles. Making quantitative measurements with side-looking imaging sonar systems has long been a goal of researchers. Bathymetry is one type of quantitative measurement that can be made with uncalibrated side-looking sonar systems using interferometric processing between signals obtained from two closely-space hydrophone arrays. Other potential methods for estimating relative topography also exist, including clinometry which uses intensity information in single images (*e.g.*, shape-from-shading techniques) and stereogrammetry, a technique based on optical photogrammetry which utilizes pairs of images of the same scene acquired from two different viewing angles (*i.e.*, parallax). Changes in image intensity statistics caused by random seafloor roughness may also be used for estimating seafloor parameters, such as ripple heights or rms slopes. Additionally, quantitative water column measurements using synthetic aperture sonar for the specific case of internal wave boluses have recently been explored. This talk reviews these different methods which use uncalibrated synthetic aperture sonar systems mounted on autonomous underwater vehicles to extract quantitative information and discusses their respective advantages, difficulties and constraints. We further assess the possibility of using combinations of these techniques as complementary sources of data.

Contributed Papers

9:50

1aAOa5. Extended autonomous multifrequency water column backscatter measurements on the NW Atlantic continental shelf captured the biological response to the retreat of a fresh/cold water anomaly in the New York Bight. Joseph Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy., Southampton, NY 11968, joe.warren@stonybrook.edu), Toniann Keiling, Megan Tavenner, Sophia Will, Delaney Costante, Michael Carlowicz, and Paige Tortorice (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY)

Wideband Acoustic Recording Bottom Landers for Ecosystem Research (WARBLERs) equipped with upward-looking multiple-frequency autonomous echosounders (along with a CTD and other sensors) were deployed in May 2024 in the NW Atlantic at two sites (in water depths of 25 and 50 m) south of Long Island, New York to monitor pelagic organism (zooplankton and fish) backscatter as part of an ecosystem monitoring program. The echosounders recorded approximately 30 s of data (at ping rates ∼1 Hz) at each frequency (both narrow and broadband modes) every hour. Thirteen months of data at the shallower site and 17 months of data from the deeper site were retrieved providing insights into a variety of oceanographic and ecological processes. These instruments provide a unique (and rare) view of conditions on the shelf including: measuring bubble injection depth during winter storms, presence of forage fish schools and their predators, and short (mesoscale) and long (seasonal) temporal changes in water column backscatter. This deployment overlapped with an anomalous oceanographic event on the continental shelf in this region with waters in 2024 being significantly fresher and cooler than normal. Data at both sites can be used to determine when normal conditions resumed which coincided with significant increases in water column backscatter at the sites (at all frequencies). Challenges with deployment, recovery, biofouling, interactions with commercial trawlers, and data analyses will also be discussed.

10:05

1aAOa6. The Mayflower autonomous ship voyage to the New England shelfbreak. Jennifer J. Johnson (Woods Hole Oceanogr. Inst., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, jjjohnson@whoi.edu), Dana Yoerger, Camrin D. Braun (Woods Hole Oceanogr. Inst., Woods Hole, MA), Andone C. Lavery (Oregon State Univ., Woods Hole, MA), Brett Phaneuf (Promare, Chester, CT), Matthew Shaw (MSubs Ltd., Plymouth, United Kingdom), Jake Southerland, and Ryan Draper (Submergence Group LLC, Cedar Park, TX)

Uncrewed surface vehicles (USVs) have become increasingly effective as research platforms in the last decade. Autonomous systems have demonstrated significant value in both real-world operational settings and scientific applications. When combined, uncrewed autonomous surface vehicles (ASVs) offer powerful capabilities for ocean research by complementing traditional crewed research vessels through extended observation periods, increased operational flexibility, and reduced costs. We present a technical and operational overview of a diesel-electric propelled ASV, known as the Mayflower autonomous ship (MAS). In 2024, MAS was equipped with a scientific echosounder payload and tasked with conducting an autonomous offshore acoustic survey. MAS transited autonomously to the New England Shelfbreak ∼100 nm from port to collect acoustic backscatter measurements, including surveying biological communities throughout the water column. We discuss the ecological context of the shelf-to-pelagic MAS echosounder observations and explore potential future scientific applications of MAS and similar ASVs.

10:20

1aAOa7. Characterization of a novel oceanographic buoyancy float for passive acoustic receiving. Savannah Passaretti-Fina (Ocean Eng., Univ. of Rhode Island, 220 S Ferry Rd., Narragansett, RI 02882, s.passaretti@uri.edu), John Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA), and Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

A new oceanographic buoyancy platform, the InfiniTE™ float, equipped with energy-harvesting technology and a programmable hydrophone expands deployment duration for passive acoustic monitoring. The

InfiniTE^{TM} float can also perform up to three 1000-m profiles a day increasing sound-speed profile sampling events compared with typical float platforms without affecting mission energy budgets. In addition to the hydrophone integration, customizable mission configurations and on-board processing optimize performance for ocean acoustics applications. The floats are typically not recovered, necessitating robust onboard processing. Two temporary short-term deployments of the InfiniTE^{TM} float, in Oceanside and Monterey, CA, are used to evaluate the float's self-noise characterization and acoustic sensing performance. Preliminary results with datasets sampling at 256 kHz will be presented.

MONDAY MORNING, 11 MAY 2026 SALON A, 11:00 A.M. TO 12:00 NOON

Session 1aAOb

Acoustical Oceanography and Underwater Acoustics: Acoustical Oceanography Prize Lecture/A.B. Wood Medal Lecture

Karim G. Sabra, Cochair
Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405

David R. Dowling, Cochair
Mech. Eng., Univ. of Michigan, Dept. of Mech. Eng., Univ. of Mich., 1231 Beal Ave., Ann Arbor, MI 48109

Chair's Introduction—11:00

Invited Paper

11:05

1aAOb1. Advancing ocean acoustic observations with distributed acoustic sensing. Shima Abadi (Univ. of Washington, 185 Stevens Way, Paul Allen Ctr., Rm. AE100R, Seattle, WA 98195, abadi@uw.edu)

Recent advances in sensing technologies are reshaping the field of ocean acoustics, enabling observations at spatial and temporal scales that were previously impractical. Distributed acoustic sensing (DAS) has emerged as a promising new approach, transforming fiber-optic cables into dense arrays of acoustic sensors capable of continuous measurements over tens of kilometers and generating data volumes far exceeding those of traditional hydrophone deployments. This talk provides an overview of DAS technology from the perspective of ocean acoustics and summarizes recent efforts to understand and apply DAS to a range of underwater acoustic problems. Applications are presented spanning ambient noise analysis, ship noise characterization, and marine mammal monitoring, illustrating how DAS enables large-scale, long-duration observations of acoustic variability across space and time. Measurements collected using fiber-optic cables deployed in a range of configurations are presented, demonstrating the versatility of DAS beyond existing telecommunications infrastructure. Potentials of DAS for underwater acoustic communication are also discussed. Key challenges are addressed, including efforts to calibrate strain measurements against acoustic pressure using co-located hydrophones and to quantify the effects of acquisition parameters on sensitivity and frequency response. The talk concludes by demonstrating how DAS offers a more integrated view of the ocean by capturing acoustics alongside other co-recorded processes.

Session 1aBAa

Biomedical Acoustics, Physical Acoustics and Signal Processing in Acoustics: Advanced Ultrasound Vascular Imaging and Vessel Characterization I

John M. Cormack, Cochair
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Kang Kim, Cochair
Bioengineering, Univ. of Pittsburgh, 623A Scaife, 3550 Terrace St., Cardiology/Medicine, Pittsburgh, PA 15213

Zhiyu Sheng, Cochair
Medicine, Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 624, Pittsburgh, PA 15213

Invited Papers

8:00

1aBAa1. Photoacoustic mapping of vascularization and tissue oxygenation during wound healing. David Lemonnier (Bioengineering, George Mason Univ., Fairfax, VA) and Parag V. Chitnis (Bioengineering, George Mason Univ., 4400 University Dr., 1J7, Fairfax, VA 22030, pchitnis@gmu.edu)

Wound healing is a multistage process that relies heavily on proper tissue oxygenation, which governs extracellular matrix formation, angiogenesis, collagen synthesis, and wound contraction. Insufficient oxygen supply can disrupt tissue regeneration, and uneven oxygen distribution adds complexity, especially in chronic wounds. Despite its importance, the spatiotemporal behavior of oxygen during normal and impaired healing is not well characterized. In this study, longitudinal photoacoustic imaging was used to map oxygenation dynamics *in vivo*. Full-thickness 1-cm excisional wounds were made on the dorsum of 10 healthy C57BL/6 mice and 10 diabetic db/db mice, a standard impaired-healing model. Using a Fujifilm VisualSonics F2-LAZR system with 30-MHz ultrasound and dual-wavelength excitation (750 and 850 nm), blood oxygen saturation (SpO_2) and hemoglobin concentration were measured over 28 days. Healthy mice closed wounds by day 14 and showed peak oxygenation at day 4, during the late inflammatory phase, with elevated oxygen levels persisting after closure and extending into deeper tissue. Diabetic mice healed by day 28 and showed a delayed oxygenation peak about two weeks post-wounding. Hemoglobin levels increased steadily in both groups, moving inward from wound margins. These results provide spatially resolved insights into oxygen dynamics and highlight differences between normal and impaired healing.

8:20

1aBAa2. Photoacoustic imaging of liver vasculature. Muyinatu Bell (Johns Hopkins Univ., 3400 N. Charles St., Barton 208, Baltimore, MD, mledijubell@jhu.edu)

Photoacoustic imaging has the potential to provide visualization of major blood vessels during surgery to avoid severe hemorrhage, which could possibly cause patient death. The PULSE Lab investigated the feasibility of *in vivo* blood vessel visualization for photoacoustic-guided liver surgeries. *In vivo* photoacoustic imaging of major blood vessels was successfully achieved after a laparotomy was performed on swine. Three-dimensional photoacoustic imaging with a robot-controlled ultrasound (US) probe and color Doppler imaging were used to confirm vessel locations. We observed that photoacoustic signals were more focused when the light source encountered a major vessel in the liver. This observation can be used to distinguish major blood vessels in the image plane from the more diffuse signals associated with smaller blood vessels in the surrounding tissue. Histopathological analyses were then performed to explore possible laser-related damage. Results are generally promising for photoacoustic-guided abdominal surgery when the US probe is fixed and the light source is used to interrogate the surgical workspace.

8:40

1aBAa3. Robot-assisted laparoscopic photoacoustic imaging for neurovascular bundle visualization. Haichong Zhang (Robotics Eng. and Biomedical Eng., Worcester Polytechnic Inst., 100 Inst. Rd., Worcester, MA 01609, hzhang10@wpi.edu), Ryo Murakami, Yernar Zhetpissov, and Shang Gao (Robotics Eng., Worcester Polytechnic Inst., Worcester, MA)

In this talk, we will introduce a spectroscopic photoacoustic (PA) imaging system for robot-assisted radical prostatectomies. PA is an emerging imaging modality that combines the deep tissue penetration of ultrasound (US) imaging with the high optical contrast of light-based techniques. Using PA imaging, a surgical robot can provide intraoperative neurovascular guidance to the operating physician, alerting them to the presence of critical anatomical structures, such as neurovascular bundles (NVBs), that are invisible to the naked eye. This capability helps prevent complications, such as hemorrhage and paralysis. Our proposed framework integrates seamlessly with the robot-assisted surgical system: real-time PA images generated by the framework are superimposed on the endoscopic video feed using an augmented

reality overlay, enabling intuitive 3-D localization of vital anatomy. We have built a prototype of the PA probe and successfully detected NVBs in an *ex vivo* rat model. Additionally, our recent efforts include developing a miniaturized drop-in PA probe for robotic surgical systems and creating a robust PA augmentation pipeline for endoscopic vision to compensate for tissue deformation. We believe this technology will help reduce postoperative complications, prevent NVB damage, and ensure complete tumor removal.

9:00

1aBAa4. Transforming medical ultrasound for continuous monitoring of neurovascular pathophysiology. Jeeun Kang (Johns Hopkins Univ., 733 Rutland Ave., Traylor 822, Baltimore, MD 21205, kangj@jhmi.edu)

Ultrasound imaging plays a crucial role in critical care medicine by providing real-time, non-invasive assessments of dynamic pathophysiological states. Our research at Hopkins aims to bridge the gap between episodic diagnostics and continuous physiological surveillance, leading to early identification of at-risk patients with actionable insights for timely interventions. In this lecture, I will outline the evolution of our research. Initially, we conducted foundational investigations into non-invasive transcranial photoacoustic imaging of perinatal hypoxic-ischemic encephalopathy (HIE) and stroke in clinically relevant animal models, and artificial intelligence unveiled its feasibility to identify at-risk brain with high accuracy. These early studies suggest a feasibility for a physiologically grounded approach for identifying early brain distress with high sensitivity and specificity. Building upon these discoveries, our research has progressed toward translational applications. We have developed wearable ultrasound/photoacoustic imaging systems for continuous fetal and neonatal brain monitoring. These devices are designed to meet the unique spatial, temporal, and physiological requirements in critical care applications, such as labor and delivery, the neonatal intensive care unit (NICU), and pediatric critical care settings. [Work supported by the National Institutes of Health Blueprint for Neuroscience Research (Grant Nos. U54EB015408 and U54EB033650)].

9:20

1aBAa5. Multiparametric imaging of brain hemodyamics using advanced ultrasound and photoacoustic imaging methods. Sri-Rajasekhar Kothapalli (Biomedical Eng., Penn State Univ., 325 Chemical Biomedical Eng. Bldg., 375 Sci. Dr., State College, PA 16803, srkothapalli@psu.edu), Prameth Gaddale, Shubham Mirg, Mahaan Mitra, Haoyang Chen (Biomedical Eng., Penn State Univ., State College, PA), Aayushi Laliwala, Agata A. Exner (Dept. of Radiology, Case Western Reserve Univ., Cleveland, OH), Jessica Sallavanti, Wei Li (Pediatrics, Penn State College of Medicine, Hershey, PA), Bruce Gluckman, and Patrick Drew (Eng. Sci. Mech., Penn State Univ., State College, PA)

Understanding stimulus-evoked cerebral hemodynamics is critical for elucidating brain function and neurological disease mechanisms. However, existing neuroimaging approaches are often limited by spatial or temporal resolution, imaging depth, sensitivity, or the ability to simultaneously capture multiple hemodynamic parameters. To address these challenges, we developed a hybrid functional ultrasound–photoacoustic (fUSPA) imaging platform integrating ultrafast ultrasound with multispectral photoacoustic techniques. This system enables real-time quantitative mapping of cerebral blood volume (CBV), cerebral blood flow (CBF), and blood oxygen saturation (SO_2) at high spatiotemporal resolution using a compact head-mounted probe. In addition to intrinsic hemodynamic imaging, the platform supports microbubble-enhanced super-resolution ultrasound for microvascular flow measurements and photoacoustic contrast agent-based imaging, such as indocyanine green, for vascular perfusion assessment. Using fUSPA, we investigated brain-wide cerebrovascular reactivity (CVR) at single-vessel resolution by quantifying relative changes in CBV, CBF, and SO_2 during hypercapnic stimulation. Functional analyses reveal distinct CVR responses between cortical arteries and veins, as well as anti-correlated CBV oscillations during resting conditions. We demonstrate the potential of multiparametric fUSPA imaging to interrogate complex cerebrovascular dynamics and advance studies of brain function and neurodiseases, such as brain cancer.

Contributed Papers

9:40

1aBAa6. Numerical investigation of a frequency-domain model-based method for 3-D transcranial photoacoustic computed tomography. Hyungjoo Park (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hyungjoo4408@gmail.com) and Yun Jing (Graduate Program in Acoust., Penn State Univ., State College, PA)

Photoacoustic computed tomography (PACT) offers rich optical contrast combined with the penetration depth of ultrasound for transcranial imaging. However, the skull introduces significant phase aberrations that degrade image quality. While the time-reversal (TR) method can mitigate these acoustic barriers, its computational intensity makes it unsuitable for time-sensitive applications. To overcome these limitations, we propose a frequency-domain model-based (FDMB) method. The performance of the FDMB method is evaluated against the TR method across various sensor densities, data sampling rates, and perturbations in speed-of-sound maps and sensor locations using 3-D numerical modeling. The results show that the proposed method is three orders of magnitude faster than the TR method while maintaining better image quality, highlighting its potential for efficient and accurate 3-D transcranial PACT.

9:55

1aBAa7. Improving photoacoustic imaging through the skull using deep learning: Considering 3-D effects. Matthew Olmstead (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mjolmstead@mac.com), Hyungjoo Park (Acoust., Penn State Univ., University Park, PA), MD Rizwanul Kabir, Aiguo Han (Biomedical Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), and Yun Jing (Acoust., Penn State Univ., State College, PA)

Human brain mapping has become one of the most exciting fields of research for medical imaging. Among other imaging techniques, photoacoustic computed tomography (PACT) has shown great promise due to its rich optical absorption contrast, high spatial and temporal resolutions, and relatively deep penetration. However, before the acoustic signal makes it to the transducer array for image reconstruction, it becomes distorted by the porous skull, resulting in blurry images. In recent years, deep learning (DL) has been proposed as a solution to achieve images of higher quality, with U-Net being a common architecture. This numerical study utilized a large collection of blood vessel models obtained from an online database and a micro-CT scan of an *ex vivo* human skull. Simulations were run in a 3-D space, accounting for out-of-plane acoustic phenomena. While the results show improved images, the model performs noticeably better when tested on images generated under the same conditions as the training data. These findings demonstrate U-Net's robustness and provide insights into future architectural enhancements.

10:10–10:25 Break

Invited Papers

10:25

1aBAa8. Contrast-agent-free ultrafast Doppler imaging of embryo–placental hemodynamics in pregnant mice. Mark Burgess (Dept. of Medical Phys., Memorial Sloan Kettering Cancer Ctr., New York, NY), Orlando Aristizabal (Ctr. for Adv. Imaging Innovation and Res. (CAI2R) and Bernard and Irene Schwartz Ctr. for Biomedical Imaging, Dept. of Radiology, New York Univ. Grossman School of Medicine, New York, NY), and Jeffrey A. Ketterling (Dept. of Radiology, Weill Cornell Medicine, Dept. of Radiology, 416 E 55th St. MR-008, New York, NY 10022, jek4011@med.cornell.edu)

The placenta is a critical interface that regulates exchange of nutrients, gases, and waste while maintaining separation of fetal and maternal blood. Many placental defects lead to embryonic lethality. In the mouse, the fetal and maternal vascular spaces closely interdigitate within the placental labyrinth, making transport across this barrier highly relevant to maternal–fetal health. However, *in vivo* assessment of embryo–placental hemodynamics remains challenging due to the small caliber of the blood vessels. We demonstrate a contrast-agent-free approach for imaging embryo–placental flow in a pregnant mouse at embryonic day 13.5 (E13.5) using high-frequency ultrafast ultrasound. A Verasonics system with a 15-MHz linear array was used to acquire ultrafast plane-wave data. B-mode, power Doppler, color Doppler, and Doppler waveforms were generated. Data were collected across a series of imaging planes to enable 3-D reconstruction. Ultrafast Doppler provided detailed visualization of circulation in the embryo, placenta, and yolk sac, and fetoplacental vascular tree. Doppler waveforms demonstrated pulsatile flow in the umbilical cord and vitelline vessels. These results support ultrafast power Doppler as a sensitive, non-contrast technique for mapping small-vessel flow *in vivo* suggest the possibility of separating blood flow contributions on the maternal versus fetal sides of the mouse placenta.

10:45

1aBAa9. Wide-field-view vector flow imaging approaches for visualization of aortic blood flow. Takuro Ishii (Graduate School of Biomedical Eng., Tohoku Univ., 6-6-05, Aramaki Aza Aoba, Aoba ward, Sendai, Miyagi 9893127, Japan, takuro.ishii@tohoku.ac.jp), Kei Mitsui, Kaya Takakusagi, Tomu Morishita, Yasuharu Funamizu, and Yoshifumi Saijo (Graduate School of Biomedical Eng., Tohoku Univ., Sendai, Miyagi, Japan)

Accurate evaluation of aortic blood flow dynamics is important for understanding pathological conditions such as aortic aneurysms and for improving interventional strategies for circulatory support. However, conventional vector flow imaging (VFI) techniques are limited by a restricted field of view, hindering comprehensive assessment of large and deep vessels including the thoracic and abdominal aorta. This presentation introduces our recent studies aimed at achieving a wide field of view in ultrasound VFI for aortic flow visualization. First, a dual-angle flow vector measurement approach using a curved linear array transducer is presented. Archimedean spiral wave transmission enables extension of the observable flow vector field while maintaining sufficient angular separation between transmit wave fronts, allowing reliable flow vector visualization over a large area. Second, a 2C–3-D flow vector estimation framework using a matrix array transducer is presented. A 3-D sector wave transmission strategy is proposed to drive array elements under hardware constraints, enabling visualization of complex 3-D flow structures from a limited acoustic aperture. These wide-field-view VFI approaches extend the observable region of both abdominal and thoracic aortic flow and provide a framework for comprehensive assessment of complex aortic hemodynamics in large vessels.

11:05

1aBAa10. Contrast-free super-resolution microvascular imaging based on high frequency ultrasound for human applications. Chih-Chung Huang (Dept. of Biomedical Eng., National Cheng Kung Univ., No. 1, University Rd., Tainan 70101, Taiwan, cchuang@mail.ncku.edu.tw)

Ultrasound localization microscopy (ULM) enables super-resolution microvascular imaging but is limited by microbubble dependence. Erythrocyte-based ULM (eULM) offers a contrast-free alternative using ultra-high-frequency ultrasound to localize and track erythrocytes. In this study, we developed a flow-aware Hessian eULM framework for quantitative, contrast-free super-resolution microvascular imaging in tendonitis. By integrating flow-aware erythrocyte tracking with Hessian-based vessel filtering, FA-Hessian eULM enhances spatial resolution, microvessel continuity, and signal-to-noise ratio. In a rat kidney microembolization model, the proposed method sensitively captured dose-dependent microvascular perfusion loss after imipenem/cilastatin injection, improving the FRC-based in-plane resolution to approximately 6.5 μm and increasing image SNR from about 23 to 82 dB compared with conventional eULM, with only 30 s of data acquisition. In three patients with refractory tendonitis undergoing transcatheter arterial microembolization, FA-Hessian eULM visualized dense peritendinous neovascular networks at baseline and demonstrated marked post-treatment microvascular reductions, correlating strongly with improvements in pain and wrist function. These results suggest that FA-Hessian eULM is a promising noninvasive imaging biomarker for monitoring treatment response.

11:25

1aBAa11. f^{2}Ultrasound: Functional and focused ultrasound theranostic platform for brain disorder. Jaesok Yu (DGIST, 333 Technojoonangdaero Hyunpung-eup Dalsung-gun, E5-309, Daegu 42988, Korea, jaesok.yu@dgist.ac.kr)

Ultrasound offers a non-invasive, non-ionizing modality for deep brain monitoring while preserving physiological states. This talk introduces "f^{2}Ultrasound," a unified framework designed to integrate stimulation and monitoring within a single platform to elucidate brain mechanisms. A core component is ultrasound localization microscopy (ULM), which overcomes diffraction limits by localizing microbubbles to reconstruct microvascular networks transcranially. To advance this technology for practical and translational applications, we present key technical evolutions. First, we implement sub-Nyquist sampling to reduce data volume by two-thirds without

compromising image quality. Second, coded-excitation techniques are utilized to achieve high SNR without contrast agents. Third, deep-learning pulse compression enhances penetration depth and signal clarity. Finally, GPU-accelerated processing enables real-time imaging with speeds increased by >1000-fold, facilitating observations in semi-freely moving rodents. These innovations pave the way for simultaneous neuromodulation (*e.g.*, sonogenetics) and functional monitoring. By combining data efficiency, AI-driven signal processing, and high-speed computing, this research establishes a new paradigm in neuroscience, bridging the gap between preclinical animal studies and clinical applications for brain disorders.

Contributed Paper

11:45

1aBAa12. Sub-millimeter 3-D-printed mouse brain vascular phantom for ultrasound imaging validation. Luca Giaccone (Univ. of Trento, Via Sommarive, 9, Trento 38123, Italy, luca.giaccone@unitn.it), Giulia Tuccio (DISI, Univ. of Trento, Povo, Trento, Italy), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

Ultrasound (US) imaging has emerged as an effective technique for visualizing the vasculature, offering a real-time, low-cost, and non-invasive solution. Nevertheless, US-based methods are susceptible to artifacts and reconstruction errors, making *in vitro* validation essential before clinical translation. Despite significant progress in US imaging, such as ultrasound localization microscopy that pushed spatial resolution to the micrometric scale, the development of vascular phantoms for *in vitro* validation remains limited. In particular, existing phantoms often fail to capture complex, realistic, and micrometric-scale vascular structures. While 3-D printing offers wide flexibility in phantom design, material selection remains a critical challenge due to low ultrasound transparency. In this study, we developed realistic vascular phantoms using 3-D printing, starting from an anatomically faithful CAD model of a mouse brain vasculature. We tested different photopolymeric materials using a polyjet 3-D printer (stratasys j750), achieving minimum vessel diameters of 800 μm. Our results identify Agilus30 as a candidate material with sufficient acoustic transparency, allowing imaging up to 4 cm depth. These 3-D-printed phantoms successfully replicate realistic vascular architectures while enabling ultrasound penetration, offering a robust platform for validating emerging US reconstruction techniques.

MONDAY MORNING, 11 MAY 2026 FRANKLIN HALL 7, 10:15 A.M. TO 12:00 NOON

Session 1aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics: Advanced Methods

Benjamin Wood, Chair
Graduate School of Biomedical Sci., Mayo Clinic, 200 1st St. SW, Rochester, MN 55902

Contributed Papers

10:15

1aBAb1. MICROBEAD: A novel Doppler detectable device with suspended microparticles for use as a biopsy marker. Benjamin Wood (Dept. of Radiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55902, wood.benjamin@mayo.edu), Charles Capron, Christine Lee, and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

For patients with metastatic axillary lymph nodes from breast cancer, the common treatment method is neoadjuvant systemic therapy followed by targeted axillary dissection. Markers are preferably visualized by ultrasound but are lost in up to 50% of cases. The MICROBEAD (Microparticles Inside a Cylinder ReOrganized By Externally Applied Doppler) design, comprised of a capsule filled with a suspension of deionized water, Tween 20 surfactant, and polyethylene microspheres, was developed as a novel Doppler-enabled marker. This design uses ultrasound-induced streaming as a source of color Doppler signal. We evaluated this design over a range of capsule sizes down to 1.4 mm outer diameter (OD) over a range of Doppler frequencies and transducers (9L, ML6-15; GE Logiq E9) in degassed water and *in situ* porcine environments. We also explored different concentrations of spheres and surfactant in the fluid solution. Results show substantial Doppler signal down to the 1.4 mm OD capsule at Doppler frequencies between 3.1 and 12.5 MHz. *In situ* scans of MICROBEAD markers with ML6-15 and 9L probes showed robust Doppler signal. These results show that the MICROBEAD design is distinct and readily distinguishable from background tissue B-mode at the required scale for clinical translation.

10:30

1aBAb2. Assessment of collateral damage during histotripsy procedures. Bradley Burden (U.S. Food and Drug Administration, Arden, NC), Seyed Ahmad Reza Dibaji (U.S. Food and Drug Administration, 10903 New Hampshire Ave., WO64 Rm. 3032, Silver Spring, MD 20993, ahmad.dibaji@fda.hhs.gov), and Matthew R. Myers (U.S. Food and Drug Administration, Phoenix, AZ)

Histotripsy relies on mechanical effects generated by short ultrasound emissions, and it can be used to treat various conditions including cancer tumors and neuroglial, cardiac, and valvular diseases. However, non-linear propagation of the high-pressure pulses leads to significant energy absorption, and there is a risk for undesired heating and thermal damage. Assessing the thermal impact during histotripsy procedures is challenging, as the device utilizes highly non-linear waves, which makes the phenomenon difficult to model. The goal of this study is to develop a regulatory science tool to assess thermal safety of these devices. In this study, a combined

computational and experimental approach was taken to estimate thermal effects of histotripsy procedures. A tissue phantom embedded with thermocouples was fabricated to record the ultrasound induced temperatures at remote locations during histotripsy procedure. To estimate the temperature elevation over the entire field, a distribution function in the form of a double-Gaussian was used to model intensity. The constants of this distribution function were calculated using an optimization routine. Temperatures from the double-Gaussian model and the thermocouples agreed to within 5%. The technique will be applied to further evaluate the effect of different ultrasound parameters on potential thermal damage during histotripsy procedures.

10:45

1aBAb3. Ultrasound synthetic aperture image reconstruction by phase cancellation in the Fourier domain. Raghu Raghavan (Sonovance, Inc., 4203 Somerset Pl., Baltimore, MD 21210, raghu@sonovance.com) and Jeremy Dahl (Radiology, Stanford Univ., Palo Alto, CA)

Reconstruction of a 3-D ultrasound volume from freehand planar scans is generally done by *incoherently* "stacking" the planar scans. We describe an approach suitable for combining beams and planar scans from focused transmissions where we introduce coherence in the off-beam and off-plane reconstructions, as an alternative to synthetic aperture. This involves phase cancellations in the (spatial) one-dimensional Fourier domain corresponding to the off-beam or off-plane dimension. Through Field II simulations, we show the quantitative sharpening of the images at different axial depths. Modeling an L12–3V transducer with a transmit focus at 20 mm, the estimated full width at half maximum of the original and improved PSF in the elevation dimension is 2.7 and 0.48 mm, respectively, at a depth of 5 and 3.7 and 0.38 mm, respectively, at a depth of 40 mm. Like virtual source synthetic aperture, we do not obtain improvement at the focal depth. We further demonstrate the validity of the approach using planar scans from tissue mimicking phantoms. We also show the method is applicable to improving lateral resolution from in-plane B-mode lines. We also compare the method to synthetic aperture with coherent compounding from different poses and beams.

11:00

1aBAb4. Evaluating the likelihood of treatment success using the passive cavitation signal from a test pulse prior to focused ultrasound treatments of tendinopathies. Grace M. Wood (Graduate Program in Acoust., Penn State Univ., 1 Old Main, University Park, PA 16802, gmw5253@psu.edu), Katelyn M. Peachey (Penn State Univ., State College, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Tendons are resistant to fractionation by focused ultrasound (FUS); however, our prior research has shown the combination of tendinopathy and histotripsy can cause mild fractionation. Here, we grouped tendons by cavitation activity observed during a 10 bovine tendons were injected with collagenase in fibrin and left overnight to induce tendinopathy. Tendons were treated with 1.5- or 2.45-MHz FUS and 5 ms pulses repeated at 1 Hz with $p+ \leq 156/p- \leq 20$ MPa or $p+ \leq 213/p- \leq 25$ MPa ($n = 8$/amplitude–frequency). Prior to treatment, the passive cavitation signal from the 10-ms test pulse was evaluated and grouped as ≤ 5, 6–14, or ≥ 15 change points, where the change points were defined as the change in slope of the time domain signal envelope; treatment depth and radius of curvature were also recorded. After treatment, samples were collected for histological analysis. Results showed a trend of increased fractionation on histology as change points decreased, but results were not significant ($p = 0.58$). Treatment depth increased with decreasing change points ($p = 0.008$); while significant ($p = 0.02$), no trend was observed between change points and radius of curvature. These results suggest that the cavitation signal present during a test pulse may indicate the likelihood of fractionation in tendons. [Work supported by NIH R01EB032860.]

11:15

1aBAb5. Impact of media filling acoustic traps on the spectral characterization of lung ultrasound vertical artifacts, an experimental study. Federico Mento (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Povo, Trento, Trento (TN) 38123, Italy, federico.mento@unitn.it) and Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

Lung ultrasound (LUS) is nowadays widely utilized by healthcare professionals to evaluate the state of the lung surface. This evaluation is mainly performed by analyzing vertical artifacts (VAs), which are often observed in patients affected by pathologies characterized by air spaces' reduction. VAs' genesis is linked to the resonance phenomena occurring within the acoustic traps (ATs) formed along the pathological lung surface. As a consequence, a spectral characterization of VAs could indirectly estimate ATs' dimensions and shape. However, no experimental studies showed the possibility to characterize the material (or medium) filling ATs with a spectral analysis of VAs. In this study, we designed and tested two materials characterized by different attenuation coefficients. The first material (MAT1) consists of a mix of gelatine and water, whereas the second (MAT2) consists of a mix of gelatine and milk. MAT2 resulted to be more attenuating than MAT1. Specifically, a slope difference of 0.335 dB/cm*MHz was estimated between the two materials, over a 4-MHz bandwidth (from 3 to 7 MHz). These materials can be used to fabricate lung mimicking phantoms having different attenuation coefficients and verify whether there exists a correlation between VAs' intensity and attenuation coefficient of materials filling ATs.

11:30

1aBAb6. Optimization methods for trajectory estimation in low-cost 3-D ultrasonic imaging. Brian Boitnott (Radiology, Stanford Univ., 235 Ayrshire Farm Ln., Apt. 105, Stanford, CA 94305, bboitnot@stanford.edu), Richard T. Lee (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., Stanford, CA), Raghu Raghavan (Sonovance, Inc., Baltimore, MD), and Jeremy Dahl (Radiology, Stanford Univ., Palo Alto, CA)

Medical access in underserved settings benefit from low-cost clinically viable ultrasound imaging. Generating accurate 3-D reconstructions from a freehand sweep of 2-D ultrasound images requires accurate estimation of the probe trajectory. Low-cost sensors, however, are not accurate enough without further improvements in the estimation. We improve the accuracy of the pose (six parameters for translation and rotation) by formulating the spatial estimation process as a sequential optimization problem based on a signal model in the complex domain. In this method, a low-cost electromagnetic sensor is used to initialize the spatial parameters. We describe a novel two-step algorithm, using first a correlation maximization for three and then a cost function minimization for the remaining three parameters. We validate the method using simulations in Field II and phantom experiments using an Interson GP-C01 transducer. For the simulations, the mean absolute error (MAE) for the rotational degrees of freedom about the x, y, and z axes are 0.22, 0.17, and 1.16 deg, respectively. The MAE for the lateral and elevational degrees of freedom are 0.04 and 0.09 mm, respectively. For the phantom experiments, the average MAE per pose for the elevational and lateral parameters are 0.09 and 0.06 mm, respectively.

11:45

1aBAb7. Sound levels from an ultrasonic angioplasty device. Sara P. Marcello (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, saram27@byu.edu), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT, saram27@byu.edu), and William C. Tang (Biomedical Eng., Univ. of California Irvine, Irvine, CA)

Accurate measurements of underwater sound levels are vital when using medical technology in fluid environments. Quantifying sound levels helps ensure device performance, safety, and reliability. The goal of this project is to measure and characterize the sound pressure levels, expressed in decibel, produced by a commercially available ultrasonic angioplasty device. Testing is performed by operating the device in a controlled water tank at Brigham Young University and recording the emitted acoustic signals using calibrated hydrophones positioned at known distances. Challenges associated with device placement in the tank and measurement positions are

addressed using a 3-D print holder and precision hydrophone placement. The recorded signals are analyzed to determine sound pressure levels across relevant frequencies. These measured levels are compared against ambient noise levels to assess consistency and signal stability. Additionally, attention is also given to factors such as distance-dependent attenuation based on the position of the hydrophone relative to the length of the device. This work provides a foundation for comparing and validating acoustic output in new angioplasty devices. The findings will support device optimization and contribute to the safe and effective development of sound-based medical technologies.

MONDAY MORNING, 11 MAY 2026 FRANKLIN HALL 11, 10:00 A.M. TO 11:55 A.M.

Session 1aCA

Computational Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Computation in Phononics and Metamaterials

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

Matthew D. Guild, Cochair
Naval Res. Lab., 4555 Overlook Ave. SW, Acoustics Div., Code 7160, Washington, DC 20375

Chair's Introduction—10:00

Invited Papers

10:05

1aCA1. Multiple scattering in phononic metamaterials with localized defects. Charles Dorn (Aeronautics and Astronautics, Univ. of Washington, Guggenheim Hall 211, 3940 Benton Ln. NE, Seattle, WA 98195, cdorn@uw.edu)

While wave dispersion in periodic metamaterials has been well studied, deviation from periodicity due to defects complicates the modeling of wave motion. However, efficient modeling of defects is critical both for expanding the metamaterial design space and for characterizing defective or damaged metamaterials. This work develops a semi-analytical multiple scattering model capturing how incident elastic waves within the passband of a metamaterial scatter due to (non-periodic) localized defects. Our model is built from the far-field asymptotics of the lattice Green's function expressed in terms of Bloch eigenmodes. To regularize the singularity in the lattice Green's function, the local perturbation method (assuming that defect length scales are smaller than the wavelength) is used to obtain a tractable multiple scattering system. This model applies to metamaterials with complex unit cells, as long as their dispersion relations are available numerically (*e.g.*, *via* finite element modeling of a unit cell). Additionally, large defect magnitudes can be considered if the defect length scale is subwavelength. For validation, comparisons with large transient finite element simulations on a finite domain show close agreement. The computational efficiency of this framework allows for optimization of defects in a metamaterial, which is demonstrated for waveguide design.

10:25

1aCA2. Controlling the phase velocity in the topology optimization of sonic black holes. Martin Berggren (Dept. of Computing Sci., Umeå Univ., Campustorget 5, Umeå 90187, Sweden, martin.berggren@cs.umu.se), Balaje Kalyanaraman (Dept. of Computing Sci., Umeå Univ., Umeå, Sweden), Abbas Mousavi (Dept. of Fluid Mech., KTH Royal Inst. of Technol., Stockholm, Sweden), and Eddie Wadbro (Dept. of Mathematics and Comput. Sci., Karlstad Univ., Karlstad, Sweden)

A gradual thinning of a plate creates a monotonic propagation speed reduction and amplitude increase of transversal waves, creating a focusing effect towards the thin end. The resulting device is known as a *vibrational black hole*. Mironov and Pislyakov (*Acoust. Phys.*, **48**, 2002) suggested an analogous device for longitudinal waves, the waveguide *sonic black hole*. Although their design works as an absorber, it does not provide the focusing properties of the vibrational black hole. Using material-distribution topology optimization, we recently introduced (*JASA* **155**, 2024) a waveguide sonic black-hole design providing broadband wave energy focusing at the distal end of the device. Unfortunately, the axial phase speed, although reduced on average, turned out to vary wildly throughout the device (*Acta Acoust.* **9**, 2025). This property is not entirely surprising since the objective function used in the optimization only considered focusing. Based on ideas from geometric acoustics and ray theory, we derived an expression for the effective axial phase velocity from solution of the Helmholtz equation for the acoustic pressure. Utilizing this quantity as optimization objective, in addition to the previous one, we show that it is possible to design a waveguide sonic black hole resembling both properties of its vibrational counterpart.

Contributed Paper

10:45

1aCA3. Abstract withdrawn.

Invited Papers

11:00

1aCA4. Period-finding computation in a nonlinear acoustic metamaterial structure. Ilia Kuk (Dept. of Mathematics, and New Frontiers of Sound Sci. and Technol. Ctr., The Univ. of Arizona, 617 N Santa Rita Ave., Tucson, AZ 85721, ilyakuk@arizona.edu), Ivan B. Djordjevic (Dept. of Elec. and Comput. Eng., FAMU-FSU College of Eng., Florida State Univ., Tallahassee, FL), Keith Runge (Dept. of Mater. Sci. and Eng., and New Frontiers of Sound Sci. and Technol. Ctr., The Univ. of Arizona, Tucson, AZ), Ildar R. Gabitov (Dept. of Mathematics, The Univ. of Arizona, Tucson, AZ), and Pierre A. Deymier (Dept. of Mater. Sci. and Eng., and New Frontiers of Sound Sci. and Technol. Ctr., The Univ. of Arizona, Tucson, AZ)

We present a quantum-inspired computational method in which information is encoded and processed in the relative geometric phases of acoustic fields. The method is implemented in a nonlinear topological acoustic metamaterial structure and uses phase bits (phibits), where logical states correspond to phase relationships between spectrally defined nonlinear mixing components. To enable reliable multistep computation, we introduce two tools: a phase cache that dynamically tracks phase updates across a computation, and an operator spectra shift that provides a consistent mapping between experimentally applied physical manipulations and the intended computational operations. Using this framework, we realize the period-finding core of Shor's integer factorization algorithm and experimentally demonstrate factoring of the composite integers 15 and 35. Measured probability distributions are benchmarked against theoretical predictions and show strong agreement, indicating robustness of the nonlinear acoustic implementation. The results establish phase-encoded phibit computation as a practical way for implementing quantum-inspired algorithms on an acoustic wave platform at room temperature. [Work supported by the Science and Technology Center New Frontiers of Sound (NewFoS) through the U.S. National Science Foundation (NSF) Cooperative Agreement No. 2242925.]

11:20

1aCA5. Spectral-state computation in a nonlinear metamaterial resonator using logical elastic bits. M Arif Hasan (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, Hasan.Arif@Wayne.Edu), Pierre A. Deymier, Keith Runge (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ), and Joshua A. Levine (Comput. Sci., The Univ. of Arizona, Tucson, AZ)

Mechanical metamaterials with engineered nonlinearity can compute by encoding information in harmonic spectra. We demonstrate a logical elastic bit in a two-mass conical-spring resonator operating at room temperature. Under steady large-amplitude drive, the unit cell produces a stable, phase-coherent harmonic ladder. Time-resolved laser Doppler vibrometry yields velocity signals whose Fourier components are projected onto orthogonal in-phase and out-of-phase eigenmodes, producing complex modal coefficients that map directly to a Bloch-sphere state. Pairing equal-frequency components yields stationary, phase-defined states suitable for memory, while pairing detuned harmonics introduces a beat frequency that drives deterministic Bloch precession, implementing Pauli-X and Hadamard-like single-bit rotations with time as the intrinsic gate clock. By partitioning the harmonic spectrum into multiple frequency blocks, several elastic bits can coexist within a single resonator, expanding the effective Hilbert space dimension without additional hardware. A nonlinear mass-spring reduced-order model reproduces measured Bloch trajectories and provides closed-form expressions for state preparation and gate timing, enabling computational design and optimization of metamaterial computing elements. [Work supported by NSF Grant Nos. 2204382, 2204400, and 2242925.]

Contributed Paper

11:40

1aCA6. Compact low-frequency wave-based computing enabled by singular topological states. Ahmad Aljabali (Mech. Eng., Wayne State Univ., Detroit, MI) and Mohammad A. Bukhari (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., 2140, Detroit, MI 48202, bukhari@wayne.edu)

As digital computers near their performance limits, continued advances in AI require new computing architectures. Wave-based analog mechanical computers (AMCs) enabled by metamaterials offer fast operation and low power, but they remain highly sensitive to imperfections and struggle to operate at low frequencies in compact form. In this work, we overcome these barriers by introducing "singular topological states" that arise from tailored spatial modulations in "locally resonant metamaterials" (LRMMs), leveraging their inherent subwavelength functionality. We first analyze a one-dimensional LRMM chain with embedded local resonators and demonstrate the emergence of robust edge-state modes (ESMs). We then extend the study to 2-D waveguiding configurations to verify topological-like protection, showing strong suppression of backscattering and resilience to disorder and fabrication imperfections. Remarkably, these properties persist even though conventional calculations yield a zero Chern number, challenging standard topological classifications. Finally, we harness these ESMs to design a topological mechanical solver for first-order differential equations. The resulting system delivers high accuracy and robustness, providing a promising route toward practical, compact, low-frequency, and noise-resilient analog mechanical computing.

MONDAY MORNING, 11 MAY 2026 FRANKLIN HALL 12, 8:45 A.M. TO 12:00 NOON

Session 1aEA

Engineering Acoustics: Acoustic Transducers and Sensors

Ahmed Allam, Cochair
Mechan. & Mater. Eng., Univ. of Cincinnati, 2851 Woodside Dr., Rhodes Hall 500, Cincinnati, OH 45221

Michael R. Haberman, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

8:45

1aEA1. Contact microphone testing: Variability between devices of the same brand and model. Fernanda P. Figueroa-Martinez (Univ. of Iowa, 302 S. Grand Ave., Iowa City, IA 52246, fernanda-figueroa@uiowa.edu), Adrian Castillo-Allendes (Michigan State Univ., Iowa City, IA), Mark L. Berardi, and Eric J. Hunter (Univ. of Iowa, Iowa City, IA)

High background noise environments remain a major challenge for voice evaluations. Contact microphones present a promising, affordable alternative, but the most accessible models often lack manufacturer data on frequency response and device variability. There is limited literature regarding the consistency of devices from the same brand and model, highlighting a need for further research in this area. We assessed device variability using a standardized shaker platform, using an accelerometer as reference. Devices were tested with different stimuli (logarithmic sine sweep and white noise) and two surfaces (platform and neck): (1) microphones on the platform using built-in loudspeaker stimuli, (2) microphones on the platform with stimuli from a loudspeaker one meter away, and (3) microphones on the neck with stimuli from a loudspeaker one meter away. Devices of the brand Alomejor showed the strongest correlations and agreement (Bland–Altman) between devices in terms of corrected frequency response. All the devices maintained a low sensitivity and good flatness in background noise conditions, and sensitivity was even lower when the microphones were attached to a neck instead of the testing platform. Alomejor contact microphones are a reliable low-cost alternative to regular microphones in challenging contexts with high background noise. [Work partially supported by the National Institutes of Health Grant R01 DC012315 from the National Institute on Deafness and Other Communication Disorders, awarded to Eric J. Hunter, Ph.D.]

9:00

1aEA2. Acoustic velocity field measurements and simulations of flat-panel loudspeakers. Aaron T. Bundy (Elec. & Comput. Eng., Univ. of Rochester, 500 Joseph C. Wilson Blvd., 413 Comput. Studies Bldg., Rochester, NY 14627, abundy@ur.rochester.edu) and Michael C. Heilemann (Elec. & Comput. Eng., Univ. of Rochester, Rochester, NY)

The acoustic velocity vector field is an objective measurement that indicates the direction of wavefront propagation. Sound fields comprised of velocity vectors pointing toward the same location in space, such as those produced by a conventional loudspeaker in an anechoic chamber, indicate a well-defined position of the acoustic source. Alternatively, a sound field where each velocity vector is random in both time and direction, such as that produced by late reflections in a reverberant space, may be described as "diffuse," making the position of the acoustic source impossible to determine. Distributed mode loudspeakers (DMLs) employ a small force actuator to excite the sound-radiating bending modes of a flat plate and have been described as diffuse radiators in many early publications. We present a

collection of acoustic velocity field measurements and simulations comparing traditional loudspeakers with both single and multi-exciter flat-panel loudspeakers. This comparison demonstrates that heavily damped flat-panel loudspeakers employing actuator arrays to selectively excite or cancel the vibrations of low-frequency bending modes can produce acoustic velocity vector fields with the same directional behavior as those produced by conventional loudspeakers.

9:15

1aEA3. Speaker characterization testing for acoustic-boundary layer receptivity experiment. Riley K. Soule (Appl. Res. Lab., Penn State Univ., University Park, PA 16802, rks6013@arl.psu.edu), David E. Mayo, and Zachary Berger (Appl. Res. Lab., Penn State Univ., University Park, PA)

High powered external acoustic forcing is being explored as a method of modifying the lifting performance and boundary layer development over airfoils, both as a method to increase flight efficiency and to decrease lift, depending on the goals of the end user. In order to properly assess the efficacy of such forcing, the sound source needed to be characterized. A compression driver has been tested in an anechoic chamber with four different horn designs. An array of seven microphones placed in an arc 2 m away from the horn exit plane to assess the sound pressure level (SPL) and directivity of the source. The compression driver was operated with tonal forcing at select frequencies between 500 and 5000 Hz as well as white noise forcing with the goal of selecting the horn design that produced the highest SPL at the target at the frequencies of interest while minimizing harmonic tones at driving frequency. [Work supported under ONR Contract N00024-25-F-8363.]

9:30

1aEA4. Simulation of electroacoustic devices with open source software. Mark Kahrs (ECE, Univ. of Pittsburgh, 1238 Benedum Hall, 3700 O'Hara St., Pittsburgh, PA 15261, kahrs@pitt.edu)

Historically, the use of equivalent circuit models dates back more than a century. Leach (2003) was a strong proponent of SPICE modeling of electroacoustic circuits. And although the use of two-ports to represent electroacoustic devices dates back at least to Lampton (1978), it's only recently with the 2nd edition of Beranek (and Mellow) (2019) that two-port modeling has emerged. We will discuss the use of public open source software for modeling lumped electroacoustic systems. Octave, a not-quite compatible MATLAP look-alike can be used to model two-port transfer matrices and perform frequency analysis. Gnu SPICE (ngspice) can be used to model electroacoustic circuits using controlled sources with passive components and perform time-domain analysis. These two resources can be used to simulate lumped electroacoustic circuits that model common devices, such as loudspeakers, headphones, earphones, and microphones. The two approaches will be compared and contrasted with an eye toward expressivity, complexity and ease of debugging.

9:45

1aEA5. Spatial sampling method for laser-based comparison calibrations of the smaller type of laboratory standard microphones. Randall Wagner (NIST, NIST—100 Bureau Dr., MS 6833, Gaithersburg, MD 20899, randall.wagner@nist.gov) and Richard Allen (NIST, Gaithersburg, MD, randall.wagner@nist.gov)

Laboratory standard microphones are condenser microphones used for highly accurate sound pressure measurements and calibrations. Specifications for these microphones are provided in a nationally adopted International Electrotechnical Commission standard. The larger type, designated as type LS1P, has a diaphragm diameter that is nominally 18.6 mm, while the smaller type, designated as type LS2P, has a diaphragm diameter, that is, nominally 9.3 mm. The larger type provides a sensitivity that is nominally 11 dB higher than the smaller at low frequencies, but the smaller provides a relatively flat frequency response to higher frequencies. We previously demonstrated a precision laser-based comparison calibration method for type LS1P microphones that uses one such microphone calibrated by the pressure reciprocity method as a transfer standard to calibrate other type LS1P microphones using the electrical drive current and the microphone diaphragm velocity measured while each microphone is driven as a sound source. Microphone diaphragm velocity is measured over a central region of each microphone using a laser-Doppler vibrometer. Here, we extend the approach to type LS2P microphones, which due to the different diaphragm sizes, requires different spatial sampling methods. Spatial sampling approaches, sensitivity calibration data, and uncertainties for both microphone types will be discussed.

10:00

1aEA6. Improved fluid handling and manipulation using a floating electrode unidirectional transducer. Sujith Jayakumar (Mech. Eng. and Mater. Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, sujith@wustl.edu), Ofer Manor (Chemical Eng., Technion – Israel Inst. of Technol., Haifa, Israel), and James Friend (Mech. Eng. and Mater. Sci., Washington Univ. in St. Louis, La Jolla, CA)

Recently, surface acoustic waves (SAW) have found their way into various microfluidic applications to provide useful functionality and efficiency. Traditionally, the standard interdigital transducer (IDT) design is used in acoustofluidics to generate propagating SAW later used to manipulate fluids or particle suspensions. In this configuration, the acoustic energy generated in the fluid is a narrow beam, much narrower than the wave's propagation length in the fluid. This gives rise to issues like reflections and caustics, making it difficult to control and use. To overcome this, we employ a floating-electrode unidirectional transducer (FEUDT) that produces a unidirectional surface acoustic wave, replacing the beam-like attenuation (in case of an IDT) with a continuous pressure SAW over a longer length scale in a desired direction. We establish the differences between the acoustic wave field in the absence and presence of a fluid medium for both these transducers, illustrating vortex and shear flows uniquely generated by the FEUDT system. Additionally, we show the efficiency and unidirectionality of the FEUDT in pumping fluids in a semi-infinite volume of fluid. We have observed fluid velocities up to 18 mm/s, compared to 2.5 mm/s for a standard IDT using the same experimental settings.

10:15–10:30 Break

10:30

1aEA7. Azimuth bearing estimation for a Mach–Zehnder interferometric acoustic sensor. Michael S. McBeth (Res. and Appl. Sci., Naval Information Warfare Ctr. Atlantic, NASA Langley Res. Ctr., 22 West Taylor St., M.S. 060, Hampton, VA 23681, michael.s.mcbeth.civ@us.navy.mil)

A push-pull Mach–Zehnder interferometric acoustic sensor is composed of two parallel fiber optic line sensor elements separated by one half acoustic wavelength. The maximum signal response or optical path difference occurs for a plane acoustic wave with an acoustic wave angle of ± 90 deg since one line sensor will be in a sound pressure peak while the other is in a sound pressure trough. The minimum signal occurs with an acoustic wave angle of 0 deg or 180 deg since both line sensors experience the same sound pressure variations. Here, we show how to estimate the acoustic wave arrival direction or azimuth bearing angle to within a few degrees of accuracy with acoustically short line sensor elements that are one fifteenth of an acoustic wavelength long. This is accomplished by creating a coherent reference of the signal of interest and observing the phase shift as the interferometer baseline is rotated clockwise and counterclockwise a prescribed number of degrees.

10:45

1aEA8. Titanium-based packaging of microelectromechanical systems accelerometers for fully implantable auditory prostheses. Yuetong Hao (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, yuetongh@umich.edu) and Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI)

Hearing aids and cochlear implants restore auditory perception for many patients but remain limited by environmental susceptibility and restricted daily use. Completely implantable auditory prostheses promise continuous, unobtrusive operation. However, their widespread adoption is constrained by the lack of implantable acoustic sensors that are simultaneously low-noise, mechanically robust, and suitable for long-term implantation. Recent

work from our group has demonstrated the feasibility of ultra-miniature piezoelectric microelectromechanical systems (MEMS) accelerometers for middle-ear sound sensing. Speech-relevant noise performance was achieved over the 0.1–8 kHz bandwidth through analytically guided and experimentally validated transducer design. Building on this validated MEMS sensor design, the present work addresses implantable packaging as the next-in-queue critical challenge. Titanium-based package architectures are developed using precision titanium stamping and additive manufacturing, integrated with alumina ceramic substrates and metallized electrical feedthroughs. The resulting packages aim to achieve sub-millimeter form factors while maintaining mechanical robustness and hermetic encapsulation. Ongoing and planned evaluations consider seal integrity, material biocompatibility, and the influence on sensing performance under physiologically relevant mechanical loading, guided by established standards for implantable medical devices.

11:00

1aEA9. Miniature hydrophones based on MEMS microphones operating at depth. Souleymane Cissokho (Mech. Eng., Tufts Univ., 200 College Ave.—Robinson Hall Rm. 150, Medford, MA 02155, souleymane.cissokho@tufts.edu), James Vlahakis (Mech. Eng., Tufts Univ., Medford, MA), Jonathan Anderson, Michael Val (MIT Lincoln Lab., Lexington, MA), Miles Woodward (Sea Land Aire Technologies, Jackson, MI), Daniel Freeman (MIT Lincoln Lab., Lexington, MA), Michael Klein (Sea Land Aire Technologies, Jackson, MI), and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA)

MEMS-based hydrophones offer advantages in size, weight, and power consumption (Ohata, 2007). Recent efforts have explored the use of coated commercial MEMS microphones as hydrophones, although previous testing has been restricted to shallow depths (Travaglione, 2018). We present here the design, packaging, and experimental evaluation of two MEMS-microphone based hydrophones operated at depths up to 400 feet (122 m) in Seneca Lake, New York. The designs use commercially available MEMS microphones, combined with custom preamplifiers and a molded polyurethane encapsulation procedure that includes internal stiffening elements. The encapsulation provides hydrostatic pressure survivability and general robustness and, importantly, includes an internal trapped air cavity to allow the MEMS microphone to operate efficiently. Transmission to the internal cavity is governed by the size of the internal cavity and the structural stiffness of a "window" region that admits acoustic energy while withstanding hydrostatic pressure. Six devices operated at depths up to 400 feet (122 m) achieved sensitivities at 1 kHz between −154 and −142 dB *re* 1 V/μPa, with an operating bandwidth of at least 100 Hz to 20 kHz. Self-noise is as low as 5 dB below sea state zero. Variation of performance with frequency and depth will be discussed.

11:15

1aEA10. Validation of an automated protocol for measuring source directivity in a laboratory water tank. Benjamin L. White (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, bw392@byu.edu) and Traci B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The BYU Underwater Acoustics Laboratory utilizes a water tank that measures 3.6 m long by 1.2 m wide with water depths up to 1 m. The tank's walls are lined with echo-reducing paneling, and the tank is equipped with two robotic arms capable of positioning acoustic transducers. A protocol has been developed to obtain source directivity with an automated spherical positioning and the timegated analysis of propagated signals to remove reflections. This protocol facilitates the obtaining of source directivity, providing logistical and financial benefits. This protocol is now being applied to commercially available transducers to verify its ability to obtain source directivity in a medium-sized water tank. This work summarizes the methodology of the protocol, reports on the source directivity obtained for these transducers, and identifies sources of uncertainty, confidence bounds on the measurements, and future steps for refining the protocol. [Work supported by the College of Computational, Mathematical, and Physical Sciences at Brigham Young University.]

11:30

1aEA11. Design and performance evaluation of spherical microphone arrays for 3-D acoustic imaging. Xinzhang Xiong (ZXONIC, 777 N 1st St., Ste. 440, San Jose, CA 95112, xiong@zxonic.com), Fang Yuan, and Wenbo Xiong (ZXONIC, San Jose, CA)

A spherical microphone array system was developed and evaluated for 3-D acoustic imaging in enclosed sound fields. The array consists of 36 condenser microphones mounted on a 0.2-m-diameter rigid sphere with approximately uniform angular distribution, interfaced to a ZXONIC multichannel data acquisition unit (up to 80 channels at 48 kHz) and PC-based beamforming software. Delay-and-sum beamforming under the plane-wave assumption was implemented in the spherical coordinate domain, and array performance was characterized using mainlobe width and maximum sidelobe level as key metrics. Numerical simulations compared different spherical element layouts (uniform single-layer versus coaxial double-layer configurations) and array diameters (0.2 m vs. 0.4 m) over 2–6 kHz to quantify trade-offs between angular resolution and sidelobe suppression. Results show that increasing the array diameter significantly narrows the elevation mainlobe, improving source separability in 3-D space, while smaller diameters provide stronger sidelobe suppression at higher frequencies. Laboratory measurements in a semi-anechoic chamber (background $<$ 15 dBA) using single- and dual-loudspeaker configurations validated the simulations, demonstrating stable 3-D source localization for stationary and separated broadband tonal sources. The proposed spherical array provides platform for automotive interior noise diagnosis, industrial leak detection, and general spatial acoustics research.

11:45

1aEA12. Programmable CMUT-based acoustic manipulation platform (PCAMP). Maijie Xiang (Mech. Eng., Tufts Univ., 485 Foley St., Unit 1706, Somerville, MA 02145, Maijie.Xiang@tufts.edu)

Precise manipulation of microscopic particles and cells is fundamental for studying biological processes and performing biomedical diagnostics and assays in microfluidic systems. Acoustophoresis provides an efficient and label-free approach through acoustic radiation forces generated by ultrasound fields. Here, we present a programmable capacitive micromachined ultrasound transducer (CMUT)-based acoustic manipulation platform (PCAMP) that employs an 8×8 array of micromachined elements on a 1.25 mm pitch to generate reconfigurable acoustic fields for microscale particle manipulation. The platform was fabricated using a surface micromachining process and comprises individually addressable elements integrated beneath a microfluidic chamber. By actuating the acoustic element array with prescribed amplitude and phase, spatially varying acoustic fields are generated, providing programmable acoustic radiation forces. Characterization using laser vibrometry confirms resonant operation of array elements near 400 kHz. Finite element analysis combined with analytical models was used to compute acoustic fields, acoustic radiation force distributions and particle trajectories. Simulations predict nN-level acoustic radiation forces, sufficient to trap and manipulate micrometer-scale particles and swimming cells. Ongoing experiments will show the impact of water loading on the CMUT's performance and programmable particle manipulation, to provide a scalable and reconfigurable approach to microfluidic acoustophoretic manipulation.

Session 1aPAa

Physical Acoustics: Acoustics at the Synchrotron and Neutron Beamlines

Christopher M. Kube, Cochair
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Ralph T. Muehleisen, Cochair
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Raphael Hermann, Cochair
ORNL, 1 Bethel Valley Rd., Oak Ridge, TN 37830-8050

Chair's Introduction—8:15

Invited Papers

8:20

1aPAa1. *In situ* resonant ultrasound spectroscopy at low and high temperatures for neutron scattering. Victor R. Fanelli (Neutron Sci. Directorate, Oak Ridge National Lab., Bldg. 8600, C-483, P.O. Box 2008, MS6475, Oak Ridge, TN 37831-6475, fanellivr@ornl.gov), James R. Torres (Neutron Sci. Directorate, Oak Ridge National Lab., Oak Ridge, TN), Yuya Shinohara (Physical Sci. Directorate, Oak Ridge National Lab., Oak Ridge, TN), Lisa Debeer-Schmitt, Mariano Ruiz-Rodriguez, Dunji Yu (Neutron Sci. Directorate, Oak Ridge National Lab., Oak Ridge, TN), Ercan Cakmak (Physical Sci. Directorate, Oak Ridge National Lab., Oak Ridge, TN), Ke An (Neutron Sci. Directorate, Oak Ridge National Lab., Oak Ridge, TN), and Raphael Hermann (Physical Sci. Directorate, Oak Ridge National Lab., Oak Ridge, TN)

Resonant ultrasound spectroscopy (RUS) is a high-precision, non-destructive technique that enables rapid measurements of the elastic properties of solid materials. In this talk, we present the development of a neutron-compatible RUS probe integrated with both low- and high-temperature sample environments, expanding the capabilities for *in situ* characterization of small (mm–cm scale) solid samples during neutron scattering experiments. The rapid evolution of the ultrasonic response near phase transitions allows intrinsic, high-precision tracking of the sample state *in situ*, often with greater sensitivity than conventional temperature or magnetic field sensors. The macroscopic elastic information obtained by RUS (including acoustic attenuation, shear modulus, and bulk modulus) provides a powerful complement to the atomic-scale structural information accessible through neutron diffraction. The key advantage of combining *in situ* RUS with neutron scattering is that elastic and atomic-scale information are acquired under identical experimental conditions. This enables accurate correlation studies and improved assessment of phenomena such as irreversible processes and transient behavior. [Work supported by Oak Ridge National Laboratory, managed by UT-Battelle, LLC, for the U.S. Department of Energy. A portion of this research used resources at the High Flux Isotope Reactor and Spallation Neutron Source, DOE Office of Science User Facilities operated by the Oak Ridge National Laboratory.]

8:40

1aPAa2. *In situ* synchrotron analysis on ultrasonically assisted laser melting process. Xun Liu (Mater. Sci. and Eng., The Ohio State Univ., 1248 Arthur E. Adams Dr., Columbus, OH 43212, liu.7054@osu.edu)

Metal additive manufacturing (AM) is advantageous in the rapid prototyping of complex geometries. However, the undesired as-solidified microstructure is one of the main challenges for widespread AM applications. Power ultrasound, which typically operates at frequencies of 20 or 40 kHz and power outputs of 1–5 kW, has shown various benefits in melting and solidification behavior. It has recently been utilized to improve AM processes in promoting columnar to equiaxed transition, minimizing segregation and reducing anisotropy. Despite these promising results, fundamental understandings of the mechanisms of ultrasonically assisted (UA) have remained elusive. In this study, a UA system compatible with a laser-based AM replicator is developed for *in situ* synchrotron X-ray diffraction (SXRD) analysis. Under 40 kHz and 6 μm vibration amplitude, SXRD shows a faster melting and solidification rate for AA4043 aluminum alloy. The static SXRD diffraction patterns in the UA condition exhibit higher continuities, indicating a more refined grain structure, which is verified with optical microscope and electron backscatter diffraction characterization. Thermal histories determined from the lattice parameter evolution demonstrate an increased cooling rate with UA. With a recent update of the UA system with 20 kHz frequency and 14 μm amplitude, the solidification structure of commercially pure Ti is significantly refined.

9:00

1aPAa3. Investigating the link between keyhole formation and acoustic signatures. Rik Vaerenberg (Dept. of Mech. Eng., LMSD, Flander Make @ KU Leuven, KU Leuven, Celestijnenlaan 300—Box 2420, Leuven 3001, Belgium, rik.vaerenberg@kuleuven.be), Milad Hamidi Nasab (Amnovis, Aarschot, Belgium), Lucas Schlenger (Thermomechanical Metallurgy Lab.—PX Group Chair, École Polytechnique Fédérale de Lausanne—EPFL, Neuchatel, Switzerland), Steven Van Petegem (Structure and Mech. of Adv. Mater. (SMAM) group, Ctr. for Photon Sci., Paul Scherrer Inst. PSI, Villigen, Switzerland), Ashish Ganvir (Digital Manufacturing and Surface Eng. (DMS) Res. Group, Mech. Eng., Univ. of Turku, Turku, Finland), Alexander Rack (European Synchrotron Radiation Facility—ESRF, Grenoble, France), Yunhui Chen (School of Eng., Ctr. for Additive Manufacturing, Royal Melbourne Inst. of Technol.—RMIT, Melbourne, Victoria, Australia), Antti Salminen (Digital Manufacturing and Surface Eng. (DMS) Res. Group, Mech. Eng., Univ. of Turku, Turku, Finland), Federica Marone (PSI Ctr. for Photon Sci., Paul Scherrer Inst. PSI, Villigen, Switzerland), Florian Aymanns (Ctr. for Imaging, École Polytechnique Fédérale de Lausanne—EPFL, Lausanne, Switzerland), Roland E. Logé (Thermomechanical Metallurgy Lab.—PX Group Chair, École Polytechnique Fédérale de Lausanne—EPFL, Neuchatel, Switzerland), Bey Vrancken (Dept. of Mech. Eng., Maakprocessen en-Systemen (MaPS), Flanders Make @ KU Leuven, Belgium, KU Leuven, Leuven, Belgium), and Konstantinos Gryllias (Dept. of Mech. Eng., LMSD, Flander Make @ KU Leuven, KU Leuven, Leuven, Belgium)

To tackle the issue of quality assurance in powder bed fusion–laser beam (PBF–LB), *in situ* acoustic monitoring has emerged as a promising cost effective solution. However, as most work focuses on data driven approaches, there is still a lack of a clear understanding of the physical link between the recorded acoustic signal and the events inside the melt pool. In order to investigate this link the formation of keyholes is examined using synchrotron-based high-speed X-ray imaging accompanied with measurements from an optical microphone. It is hypothesized that the vapor gas generation is the main source of the airborne acoustics, and an amplitude modulation based phenomenological model is created in order to explain the (unstable) keyhole acoustic emissions. Based on these insights, a signal processing pipeline is constructed to detect the formation of (unstable) keyholes inside the melt pool. Furthermore, it is shown that when employing a pulsed laser, an additional amplitude modulation step is introduced which aids with the classification accuracy as it makes it easier to distinguish between the background noise and the signal generated by the keyhole. [Work supported by FWO fellowship 1SE9325N.]

Contributed Paper

9:20

1aPAa4. From bubble to pore: Acoustic signatures of keyhole cavity collapse. Christopher M. Kube (Penn State Univ., 212 Earth and Eng. Sci. Bldg., University Park, PA 16802, kube@psu.edu), Nathan Kizer (Penn State Univ., University Park, PA, kube@psu.edu), Ian C. Bacon (Penn State Univ., State College, University Park, PA, kube@psu.edu), Olivia C. Van Fleet (Penn State Univ., State College, PA, kube@psu.edu), Lovejoy Mutswatiwa (Penn State Univ., University Park, PA, kube@psu.edu), Sihang Feng, Kyle A. Mumm, Dukyong Kim (Northwestern Univ., Evanston, IL), Samuel J. Clark (Argonne National Lab., Lemont, IL), Andrea P. Arguelles, Allison Beese (Penn State Univ., University Park, PA, kube@psu.edu), and Tao Sun (Northwestern Univ., Evanston, IL)

This work investigates acoustic signatures of vapor cavity dynamics during laser powder bed fusion of AlSi10Mg. The experimental approach combines passive acoustic detection with active excitation to characterize transient cavity resonances in real time. Three laser optical microphones employing Fabry–Perot interferometers are positioned with respect to the center of the 4×10 mm sample cross sectional area, ensuring distinct acoustic arrivals from any melt pool location. The microphones passively detect natural resonance modes of the millimeter scale vapor depression while a substrate coupled ultrasonic transducer delivers broadband excitation pulses at 25 kHz to actively amplify cavity resonances. Three channels of photodiode data are simultaneously collected with the acoustic measurements. All sensing is acquired through a single national Instruments chassis as part of a modular system utilizing a secondary vacuum chamber with feed throughs to port cabling into the printer. Acoustic measurements were acquired simultaneously with high-speed synchrotron X-ray imaging at APS beamline 32-ID, captured at 300 000 frames per second, enabling direct observation of cavity evolution correlated with acoustic response. Wavelet transform analysis correlates acoustic emissions from both passive and active excitation cases with observed cavity behavior. Preliminary results reveal distinct spectral features associated with keyhole pore pinch-off events.

9:35–9:55 Break

Invited Paper

9:55

1aPAa5. Understanding the acoustic signatures of stressed fractured rock by simultaneous *in situ* synchrotron X-ray imaging and acoustic monitoring. Parisa Shokouhi (Eng. Sci. and Mech., The Penn State Univ., 212 EES Bldg., University Park, PA 16802, pxs990@psu.edu), Evan Bozek, Prabhav Borate, Asef I. Sadaf (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Mark Rivers (Univ. of Chicago, Chicago, IL), and Jacques Riviere (Penn State Univ., University Park, PA)

Subsurface fracture characterization using seismic imaging is critical to understanding fluid transport within geological formations, which is vital for effective geothermal energy extraction and CO_2 sequestration, among other applications. Controlled laboratory-scale experiments help enhance the interpretation of seismic attributes in relation to fracture characteristics. Here, we introduce a novel pressure cell that allows simultaneous *in situ* synchrotron X-ray imaging, ultrasonic, and acoustic emission recording as well as flow measurements under both static and dynamic loading. Using this apparatus, we have conducted a series of sophisticated experiments at Argonne National Laboratory (GSECARS) that couples synchrotron X-ray computed tomography and time-lapse radiography with linear and nonlinear ultrasonic and acoustic emission testing of fractured Westerly granite (some thermally damaged) and Berea sandstone specimens to: (1) relate the ultrasonic response (i.e., wave amplitude, velocity, and frequency) to fracture characteristics, such as average

fracture aperture, contact area, and contact size distribution; (2) decouple the contributions of bulk rock versus fracture to the measured linear and nonlinear ultrasonic signatures of fractured rock; and (3) constrain the mechanisms behind fault reactivation and acoustic precursors to shear failure for faults of different roughness. This talk provides an overview of these experiments and discuss a subset of our findings.

Contributed Papers

10:15

1aPAa6. Direct observation of acoustic cavitation in laser additive manufacturing of aluminum using high-speed synchrotron X-ray imaging. Lovejoy Mutswatiwa (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci., University Park, PA 16802, lpm5609@psu.edu), Dukyong Kim (Mech. Eng., Northwestern Univ., Evanston, IL), Ian C. Bacon (Eng. Sci. and Mech., The Penn State Univ., State College, University Park, PA), Nathan Kizer (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Sihang Feng (Mech. Eng., Northwestern Univ., Evanston, IL), Olivia C. Van Fleet (Acoust., Penn State Univ., State College, PA), Kyle A. Mumm (Mech. Eng., Northwestern Univ., Evanston, IL), Samuel J. Clark, Kamel Fezzaa (X-ray Sci. Div., Argonne National Lab., Lemont, IL), Judith A. Todd, Andrea P. Argüelles (Eng. Sci. and Mech., The Penn State Univ., State College, PA), Tao Sun (Mech. Eng., Northwestern Univ., Evanston, IL), and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

High-intensity ultrasound application during metal additive manufacturing (AM) promotes microstructural refinement and reduces defects. These benefits are attributed to acoustic cavitation-induced shock waves, acoustic streaming, and localized thermal effects. Conventional ultrasonic processing employs high-power excitation near 20 kHz, where increased acoustic intensity enhances cavitation and microstructural benefits. However, in AM, high-intensity ultrasound results in increased surface roughness and reduced dimensional accuracy. To address these limitations, high-frequency ultrasonic excitation with sub-10-nm amplitudes was applied to melt pools generated under powder bed fusion–laser beam (PBF–LB) conditions. Melt pool dynamics and solidification behavior with and without excitation were directly observed using high-speed X-ray imaging. This work investigates whether mechanisms responsible for microstructural refinement and defect reduction under conventional 20 kHz excitation persist at higher frequencies. A stepped cylindrical resonator with a longitudinal resonant frequency of 570.79 kHz was fabricated *via* PBF–LB. Simultaneous laser melting, excitation at resonance, and X-ray imaging were done at Argonne National Lab. The results reveal rapid bubble nucleation, collapse, and transport away from the solidification front toward the keyhole and melt pool surface, reducing keyhole instability and porosity. A pathway toward realizing ultrasonic processing benefits in AM while preserving process stability is demonstrated.

10:30

1aPAa7. X-ray imaging-informed acoustic emission modeling of melt pool and keyhole dynamics in laser powder bed fusion. Ian C. Bacon (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci., State College, University Park, PA 16802, icb5130@psu.edu), Lovejoy Mutswatiwa (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Tao Sun (Mech. Eng., Northwestern Univ., Evanston, IL), Samuel J. Clark (Adv. Photon Source, Argonne National Lab., Lemont, IL), Allison Beese (Mater. Sci. and Eng., The Penn State Univ., University Park, PA), Andrea P. Arguelles, Nathan Kizer (Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Kyle A. Mumm, Sihang Feng, Dukyong Kim (Mech. Eng., Northwestern Univ., Evanston, IL), Olivia C. Van Fleet (Acoust., Penn State, State College, PA), and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

This work develops a physics-based digital twin for powder bed fusion–laser beam (PBF–LB) to improve the interpretation of acoustic signatures and support process monitoring. A hybrid modeling framework is advanced to simulate acoustic emissions during PBF–LB of AlSi10Mg. Computational fluid dynamics (CFD) simulations capture melt pool and keyhole dynamics under conditions replicating experiments at the Advanced Photon Source (APS 32-ID-C), where high-speed X-ray imaging enables direct, *in situ* observation of melt pool and keyhole behavior. These X-ray observations provide ground truth for melt pool geometry and keyhole evolution, allowing calibration and validation of the CFD model, ensuring the simulations accurately replicate the APS experimental conditions and thermal-fluid processes. Surface velocity fields obtained from the CFD simulations serve as initial conditions for a time-domain wave propagation model, supporting the prediction of both passive and actively excited acoustic responses within the build chamber. Ongoing efforts compare simulated and experimental acoustic data to evaluate model fidelity across excitation conditions and investigate near-field acoustic behavior. This framework connects melt pool events to their acoustic signatures, supporting identification of acoustic indicators of vapor depression and keyhole dynamics while providing labeled training data for machine learning-based process monitoring. This approach can be extended to other PBF–LB alloys.

Session 1aPAb

Physical Acoustics: General Physical Acoustics: Manufacturing

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Christopher M. Kube, Cochair
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Contributed Papers

11:00

1aPAb1. High-intensity focused ulrasound (HIFU) for advanced product assembly. Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu), Amy Swanson, Cody Birkland (Univ. of Washington, Seattle, WA), Adam D. Maxwell (Urology, Univ. of Washington, Seattle, WA), and Mike Pinch (Kirkland, WA)

Li-ion batteries (LiBs) were introduced by Sony in 1991. Today, LiBs dominate the market, and can be found in cell phones, wearables, computers, EVs, sensors, *etc.* LIBs are often placed in the assembly and then placed in an oven for bonding. However, LiBs degrade at higher cure temperatures. To reduce LIB damage, cure temperatures must be reduced and cure times increased. However, lower cure temperatures can lead to reduced bond strength, while increased cure times reduce efficiencies. Moreover, unintended movement within the assembly enclosure leads to poor performance. Instead of applying traditional curing techniques to modern technologies, we contend that there is a need to apply modern curing technologies to bond modern assemblies. This implies targeting heat to the bond joint so that nearby heat-sensitive components remain at nominal temperatures. The critical unmet need is therefore a technology that targets bond sites precisely, in the presence of different plastic materials, with a few millimeters' precision. Such a technology exists, called high-intensity focused ultrasound (HIFU) and is used to ablate tissues, including the prostate and brain tissues. Our goal is to develop HIFU-bonding for industrial use. This talk will discuss initial studies to develop HIFU for bonding modern assemblies. [Work supported by University of Washington CoMotion gap fund, and Royalty Research Fund.]

11:15

1aPAb2. Estimation of anisotropic transport properties for porous media. Kyle G. Dunn (U.S. Army ERDC, 72 Lyme Rd., Hanover, NH 03755, kyle.g.dunn@erdc.dren.mil), Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), and Gordon M. Ochi (U.S. Army ERDC, Champaign, IL)

The acoustic response of a porous medium fundamentally depends on its transport properties. The Johnson–Champeaux–Allard model and its extensions require several transport properties to acoustically characterize porous media in the low- and high-frequency limits. In existing work, estimation of transport properties is largely restricted to isotropic media. For anisotropic materials, some of these important properties become tensor valued because the pathways navigated by acoustic waves are direction dependent. In this work, tensor-valued results are presented for static viscous tortuosity, static viscous permeability, inertial tortuosity, and viscous characteristic length with no assumptions regarding the isotropy of the sample. Transport properties are numerically estimated using an anisotropic generalization of the hybrid multiscale method. Within the pore space, Poisson's equation is solved for a temperature field, Laplace's equation for potential flow fields, and the Stokes equations for viscous flow fields. Transport properties are estimated using the resulting temperature and velocity fields. Results are presented for select test cases, including a periodic arrangement of parallel fibers in two dimensions and the analysis of a 3-D polyethylene filter foam sample.

11:30

1aPAb3. Impedance measurements of 3-D printed porous material. James C. Morrison (Phys., Astronomy, and Eng., Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72035, jmorrison11@cub.uca.edu) and Carl Frederickson (Phys., Astronomy, and Eng., Univ. of Central Arkansas, Conway, AR)

Understanding the acoustic impedance of porous media is fundamental to acoustic engineering. This study investigates the relationship between acoustic impedance and key material characteristics, including porosity, pore shape, and pore length. We leveraged 3-D printing technology to precisely vary these characteristics and uniformly distribute the pores across the material. Experiments were conducted using a traditional impedance tube setup with a top-mounted speaker and dual microphones, measuring frequencies from 100 Hz to 1.5 kHz. For this experiment, a custom Python script was developed that utilizes binary data acquisition and the Fourier transform for accelerated data collection, allowing for rapid iteration without sacrificing accuracy. Furthermore, recent optimizations have led to a marked improvement in the quality of our impedance data, enabling us to compare acoustic properties with greater precision. This study can be used to influence the design of novel acoustic materials.

11:45

1aPAb4. Measurement of the airflow resistance of 3-D printed porous materials. Braden Lee (Phys., Astronomy, and Eng., Univ. of Central Arkansas, 201 Donaghey Ave., Conway, AR 72035, blee13@cub.uca.edu), Armando Montoya Pina, and Carl Frederickson (Phys., Astronomy, and Eng., Univ. of Central Arkansas, Conway, AR)

The concept of acoustic impedance of porous media holds significant importance in acoustical engineering. An important characteristic of porous media that influences the acoustic impedance is the media's airflow resistance. However, due to airflow resistance measurements being highly sensitive to experiment and method design, it is a challenge to determine the airflow resistance precisely. The goal of this study is to modify a previously designed and 3-D printed airflow device that more closely follows engineering standards EN ISO 9053-1 and stabilizes the airflow through 3-D printed porous plugs being studied to reduce uncertainty of measured airflow resistance values. The studied 3-D printed porous plugs vary in multiple properties, including pore length, pore shape, pore number, and overall porosity. Using the modified airflow device and the porous plugs with varying properties, a relationship between material properties and airflow resistance can be quantified. The engineering process and developments of the airflow device, along with airflow resistance values of various porous plugs will be presented.

Session 1aSP

Signal Processing in Acoustics, Acoustical Oceanography and Animal Bioacoustics: Signal Processing and Modeling for Distributed Acoustic Sensing (DAS) I

Abner C. Barros, Cochair
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Léa Bouffaut, Cochair
K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Cornell Lab of Ornithology, 159 Sapsucker Woods Rd., Ithaca, NY 14850

Chair's Introduction—8:00

Invited Paper

8:05

1aSP1. Principles of distributed acoustic sensing for underwater acoustic communications. Shima Abadi (Univ. of Washington, 185 Stevens Way, Paul Allen Ctr.—Rm. AE100R, Seattle, WA 98195, abadi@uw.edu), Samantha Juber (Univ. of Washington, Seattle, WA), Isabelle Brandicourt (Oceanogr., Univ. of Washington, Seattle, WA), Angeliki Xenaki (Foundation for Res. and Technol.—Hellas Heraklion, Crete, Greece), Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA), and Ethan Williams (Earth & Planetary Sci., Univ. of California, Santa Cruz, CA)

Distributed acoustic sensing (DAS) transforms fiber-optic cables into dense arrays of acoustic sensors, enabling high-resolution sampling of underwater wavefields over tens of kilometers. The quality and characteristics of DAS measurements depend strongly on acquisition parameters, including gauge length, channel spacing, sampling rate, and interrogator design. Understanding how these parameters influence the recorded data is critical for interpreting DAS measurements and optimizing system performance. In this study, the effects of key acquisition settings are evaluated using multiple experimental data sets collected with different DAS configurations. The analysis combines theoretical modeling of DAS strain sensing and spatial averaging with signal-processing-based characterization of experimental recordings. Comparisons across parameter configurations reveal how acquisition choices affect sensitivity, frequency response, noise floor, and spatial coherence, providing a quantitative foundation for understanding DAS measurement capabilities. Building on these insights, we discuss the emerging potential of DAS for underwater acoustic communication and conclude by highlighting DAS as a versatile platform for both sensing and information transmission in complex ocean environments.

Contributed Papers

8:25

1aSP2. Vertical array distributed acoustic sensing for higher frequency waterborne signals. Samantha Juber (School of Oceanogr., Univ. of Washington, 1501 NE Boat St., #220, Seattle, WA 98195, sjuber@uw.edu), Isabelle Brandicourt (School of Oceanogr., Univ. of Washington, Seattle, WA), Brad P. Lipovsky (Earth and Space Sci., Univ. of Washington, Seattle, WA), Quentin Goestchel, William S. Wilcock, and Shima Abadi (School of Oceanogr., Univ. of Washington, Seattle, WA)

Distributed acoustic sensing (DAS) has become widely used in passive acoustic monitoring, but most applications focus on frequencies below 1 kHz and on horizontally deployed cables. Expanding DAS to vertical configurations and higher frequencies would broaden its utility, particularly for biological monitoring of fish and marine mammals. Looped and spiral cable configurations can improve the signal-to-noise ratio (SNR) of higher frequency signals by suppressing noise and decreasing the directional response of the system. Two vertical cable configurations—incorporating loops and spiraling around a fiberglass tube—were tested in Puget Sound, Washington. Impulsive signals from a bubble pulser and a 50 Hz to 1.5 kHz sequence of signals from a Lubell VC2C transducer were recorded up to the Nyquist frequency of 3.3 kHz, verified with hydrophones. SNR was estimated in the time-frequency domain to account for noise that was unequally distributed in frequency. Both configurations improved SNR compared to straight sections. Different interrogator settings were tested, showing that data recorded at gauge lengths less than 4.79 m were too noisy for signal detection. This study shows that DAS can detect waterborne acoustic signals up to 3.3 kHz and highlights the importance of cable geometry on data quality.

8:40

1aSP3. Characterizing underwater ambient sound using distributed acoustic sensing. Lucy Brock (School of Oceanogr., Univ. of Washington, 1431 NE Boat St., Marine Sci. Bldg.—Rm. 218, Seattle, WA 98195, lcbrock@uw.edu) and Shima Abadi (School of Oceanogr., Univ. of Washington, Seattle, WA)

Distributed acoustic sensing (DAS) is an emerging technique for measuring strain using fiber-optic cables, enabling dense and continuous measurements of acoustic signals. By transmitting laser pulses through the cable and interpreting strain-induced phase variations in the backscattered light, DAS effectively forms a large array of acoustic sensors. This study uses data from an experiment conducted in Puget Sound, Washington, during which a submarine DAS cable and a co-located hydrophone collected passive acoustic data for 9 days in October 2022. The DAS system sampled at 2 kHz with a channel spacing of 6.38 m, spanning approximately 3.5 km of

cable length. By benchmarking DAS measurements against hydrophone observations, we assess the frequency-dependent capability of DAS to characterize underwater ambient sound across a range of wind and tidal conditions. This analysis aims to establish a calibration between the strain measurements from DAS and acoustic pressure levels.

8:55

1aSP4. Measurements of the acoustic pressure sensitivity of the GeoLab DAS cable off Madeira. Ernst M. Uzhansky (Phys., Naval Postgrad. School, SP-120, University Circle, Monterey, CA 93943, ernstuzhansky@gmail.com), Oleg A. Godin, TSUWEI TAN (Phys., Naval Postgrad. School, Monterey, CA), Matthew Alford, Gunnar Voet, Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), Jennifer Neale, Mohammad Belal, Carl Spingys (National Oceanogr. Ctr., Southampton, United Kingdom), Alberto N. Garabato (Dept. of Ocean and Earth Sci., Univ. of Southampton, Southampton, United Kingdom), and Afonso Loureiro (ARDITI—Agência Regional para o Desenvolvimento da Investigação, Tecnologia e Inovação, Lisboa, Portugal)

Distributed acoustic sensing (DAS) is a novel tool in underwater acoustics that measures strain in an optical fiber from the phase change of backscattered coherent light to estimate acoustic pressure. This technology transforms fiber-optic cables into dense, synchronized arrays of thousands of acoustic sensors, offering unique opportunities for marine research. However, the directivity and frequency response of DAS measurements remain poorly understood. An experiment was conducted in 2025 to characterize the DAS acoustic pressure sensitivity of the GeoLab cable off the southern coast of Madeira Island. Sound source was deployed from R/V Discovery at 14 m depth, with a reference hydrophone suspended 1 m below the source. Data acquisition covered three short towed transects and six stationary stations at 100–3000 m ocean depths. The source repeatedly transmitted a signal comprising chirps in the 105–160 Hz band and 11 tonals between 55–195 Hz. Ambient and DAS self-noise were evaluated during periods between the transmissions. Gaussian beam and wide-angle parabolic equation-based propagation modeling was performed to account for sloping bathymetry and surface reflections. Results of the measurements of the frequency and angular dependence of DAS pressure sensitivity and signal-to-noise ratio will be presented. [Work supported by NSF and NERC.]

9:10–9:25 Break

9:25

1aSP5. DAS4Orcas: Detecting marine mammal vocalizations up to 2.2 kHz with distributed acoustic sensing. Samantha Juber (School of Oceanogr., Univ. of Washington, 1501 NE Boat St., #220, Seattle, WA 98195, sjuber@uw.edu), Isabelle Brandicourt (School of Oceanogr., Univ. of Washington, Seattle, WA), Scott Veirs (Beam Reach, Seattle, WA), val veirs (Beam Reach, Friday Harbor, WA), Brad P. Lipovsky (Earth and Space Sci., Univ. of Washington, Seattle, WA), William S. Wilcock, and Shima Abadi (School of Oceanogr., Univ. of Washington, Seattle, WA)

Distributed acoustic sensing (DAS) has emerged as a powerful tool for marine acoustic monitoring, yet its application to marine mammal bioacoustics has largely been limited to low-frequency baleen whale vocalizations. This is partly due to the relatively high noise floor and directional response of linear DAS systems, which reduce sensitivity to higher-frequency signals. Looped cable geometries can mitigate this directionality and suppress noise to improve detection of higher frequencies. We deployed a lightly-armored, 2 km seafloor fiber optic cable with incorporated loops in Haro Strait, a passage for marine mammals in the Salish Sea. With a sampling frequency of 5 kHz, we successfully detected both humpback and killer whales calls up to 2.2 kHz for the first time on a subsea fiber optic cable. These results show that DAS can record higher-frequency marine mammal vocalizations than previously documented and highlight how experimental design and cable geometry can improve signal-to-noise ratios to extend the effective bandwidth of DAS.

9:40

1aSP6. Interferometry of fin whale calls using distributed acoustic sensing on OOI seafloor cables. Chih-Chieh Chien (Univ. of Washington, 4453 Stone Way N, Apt. 313, Seattle, WA 98103, ccchien@uw.edu) and Shima Abadi (Univ. of Washington, Seattle, WA, ccchien@uw.edu)

Distributed acoustic sensing (DAS) transforms existing fiber-optic cables into dense arrays of acoustic sensors, providing unprecedented spatial coverage for ocean monitoring. This study uses DAS recordings collected from November 1–5, 2021, on two seafloor cables of the Ocean Observatories Initiative Regional Cabled Array off the coast of Oregon. The dataset contains abundant fin whale calls, with arrivals clearly observable across multiple DAS channels. We apply an interferometric approach to these calls to extract travel-time information for direct arrivals and to explore the possibility of identifying reflected arrivals from the sea surface and seafloor. Cross-correlation of the DAS recordings is then performed to assess how accurately inter-channel travel times can be used to estimate the Green's function between DAS channels. The results are compared with simulations to evaluate the method's performance, and the Green's function is estimated along the full cable to analyze propagation characteristics across the array.

9:55

1aSP7. Sea ice property inversion using distributed acoustic sensing on Arctic landfast ice. Junsu Jang (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 98 Water St., Bigelow 4th Fl., MS#9, Woods Hole, MA 02543, junsu.jang@whoi.edu), Gil Averbuch, and Maddie Smith (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA)

Distributed acoustic sensing (DAS) has enabled new scientific insight into processes that are otherwise difficult to resolve spatially. Here, we leverage its unique capability to investigate ice conditions using a bespoke fiber-optic cable deployment on landfast sea ice in Utqiaġvik, Alaska, during the spring of 2025. The 2-km cable was installed in an L-shaped geometry, enabling beamforming with the virtual array of strain measurements. We estimate the arrival angle of flexural-gravity waves from ocean swells propagating within the ice plate using frequency–wavenumber analysis. This approach identifies frequency-dependent slowness vectors, from which arrival angles and phase velocities are obtained, allowing characterization of dispersive behavior. These observations were subsequently used to invert for sea ice properties, including effective Young's modulus and ice thickness. The inversion is performed using a deterministic misfit-based approach, and the results motivate a Bayesian statistical formulation of the dispersion-based inverse problem. Inversion results obtained from swell-induced flexural-gravity waves are compared with those derived from controlled hammering experiments, and suggest differences in effective Young's modulus. A follow-up DAS deployment at the same site is planned for the spring of 2026, with the goal of extending these observations to assess temporal variability in inferred ice properties.

10:10

1aSP8. Physics-based hierarchical Bayesian inference for localization in passive seismoacoustic sensing. Abner C. Barros (Naval Undersea Warfare Ctr. Div. Newport, 285 Old Westport Rd., Dartmouth, MA 02747, abarros1@umassd.edu) and Paul J. Gendron (ECE, Univ. of Masschusetts Dartmouth, Dartmouth, MA)

Distributed acoustic sensing (DAS) systems are increasingly employed for passive detection and localization in geophysical and underwater acoustic applications. However, fiber-optic sensing presents several challenges related to strain-based measurements, sensitivity, and angular response, limiting the direct application of conventional array processing methods. Uncertainty in propagation speeds, complex dispersive propagation, and imperfect knowledge ofcable position further complicate accurate source localization. To address these challenges, this study considers coherent combining techniques within a hierarchical Bayesian localization framework tailored to DAS arrays. Localization approaches are developed for an

impulsive seismic source on the Earth's surface in the near field of a DAS array, as well as for passive underwater acoustic sources. Marginalization over the weakly known source disturbance yields a posterior probability density of the source location. For the terrestrial case, the forward model incorporates a near-field Green's function with surface-wave dispersion appropriate for layered media and accounts for the DAS angular responseto surface waves. Modal information is extracted using frequency–wavenumber analysis and Radon transform, and empirical dispersion curves are fitted with a two-term exponential model. Application to DAS observations showspromising localization accuracy.

10:25

1aSP9. Revisiting matched field processing (MFP), given large apertures provided by DAS arrays. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

Matched field processing (MFP) estimates source range and depth by matching measured signals to signal replicas produced by wave propagation models. Complex pressure $p = U \times a$, where U contains modes on the receive array, and mode amplitude a is the mode sampled at source depth with phase proportional to horizontal wavenumber km times range. Mode amplitudes determine source range and depth. Mode amplitudes can be calculated by projecting p onto the modes in U, using orthogonality of modes over fully spanning VLA. Horizontal line arrays (HLAs) are easier to deploy. Mode amplitudes can be isolated on HLAs by transforming the complex pressure over range to a wavenumber spectrum. In previous efforts, HLAs with enough aperture to resolve the wavenumbers have not been available, resulting in poor MFP results with short HLAs. Using DAS arrays at endfire provides the long apertures needed to resolve the wavenumbers and normal mode amplitudes, so that the resulting ambiguity surface in range and depth has significantly lower sidelobes than have been produced by shorter HLAs. We will present analysis of using DAS arrays for MFP at endfire, and results of testing this concept on publicly available datasets.

MONDAY AFTERNOON, 11 MAY 2026 SALON C, 1:00 P.M. TO 5:00 P.M.

Session 1pAA

Architectural Acoustics, Engineering Acoustics and Noise: Memorial Session Honoring Robert Coffeen

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

Shane J. Kanter, Cochair
Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604

Robin Glosemeyer Petrone, Cochair
Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604

Ben Bridgewater, Cochair
D.L. Adams Associates, 1536 Ogden St., Denver, CO 80218

Logan Pippitt, Cochair
Talaske, 32560 W 171st St., Gardner, KS 66030

Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. Memorial session honoring Robert Coffeen. Sharon C. Leatherman (1213 Stone Meadows Dr., Lawrence, KS 66409, sharon.leatherman@gmail.com)

Bob Coffeen was an acoustical pioneer, consultant, educator, and family man. He was never at rest! There was always something to "jot down," a call to make, a report to prepare, or project papers to grade. Bob was driven to succeed, but more than that, he was driven to provide solutions and to share his knowledge and expertise. (Sharon recalls him making work notes on the bulletin while in church!) Bob was a contributing member of ASA. He always had their meetings on the calendar, and deeply valued his connections within the industry. While no longer with us, his legacy is shown through those he mentored and educated. The Coffeen family members would be

honored to share a few moments of perspective on Bob's life through the lens of the early business days, the later business success, and his move to academia. We will also share our thoughts on the best things he taught us—perseverance, dedication, commitment, and love for family.

1:25

1pAA2. A retrospective of Bob Coffeen's professional consulting career from the perspective of two of his longtime associates. Robert Ledo (D.L. Adams Assoc., 15323 W. 95th St., Lenexa, KS 66219, Rledo@AvantAcoustics.com), Edouard Charland, and Ben Bridgewater (D.L. Adams Assoc., Denver, CO)

This session will cover Bob Coffeen's professional consulting career starting from his first project at Washington Dulles International Airport and covering his work on airports, stadiums, arenas, and performing arts facilities throughout the U.S., Japan, and Germany. The session will include first-hand accounts of select projects and the development of Coffeen's acoustical consulting techniques.

1:45

1pAA3. Memorial session honoring Bob Coffeen—Joint session. Jason K. Pittman (School of Architecture, Univ. of Kansas, 1465 Jayhawk Blvd., Marvin Hal, Lawrence, KS 66045, pittman.jason.k@ku.edu)

Robert (Bob) Coffeen (1932–2024) was an artist in architectural acoustics and electro-acoustics, serving as a long-time faculty member at the University of Kansas (KU) School of Architecture and Design. Following a distinguished 35-year career as a consultant at Coffeen Fricke and Associate, hen transitioned to academia in 1992. His work at KU focused on the integration of complex physics into architectural design, specifically room acoustics, noise control, sound isolation, and sound reinforcement systems. Coffeen's research was particularly influential regarding stadium acoustics and the impact of air-supported fabric roofs on sound quality. He was a pioneer in using computer modeling and auralization, allowing architects to "hear" a space before it was built. At KU, he was renowned for his hands-on teaching style, mentoring over 50 students who entered the professional acoustics field. His contributions were recognized with numerous accolades, including the Acoustical Society of America's Rossing Prize in Acoustics Education and the NSCA Educator of the Year award multiple times. His legacy at the University of Kansas is defined by his ability to bridge the gap between engineering precision and architectural art, ensuring that thousands of graduates entered the profession with a specialized understanding of the art of sound.

2:05

1pAA4. Honoring Bob Coffeen: Teacher, mentor, and friend. Lauren M. Ronsse (Durham School of Architectural Eng. and Construction, Univ. of Nebraska – Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, lronsse@unl.edu), Andrew N. Miller (BAi, LLC, Austin, TX), and Shane J. Kanter (Threshold Acoust. & AV, Chicago, IL)

Bob Coffeen made numerous significant contributions to the field of acoustics and AV throughout his lifetime. One very special contribution was the outstanding education that Bob provided for many students at the University of Kansas. This presentation recalls Bob's dedication to teaching and mentoring and shares some of his unique educational techniques. This presentation also explores the profound impact Bob had on forming the careers of his students and expresses our gratitude for his influence on our lives.

2:25

1pAA5. "Well! Now!"—A legacy of curiosity, enthusiasm, and kindness. Liz Lamour Croteau (Acentech, 33 Moulton St., Cambridge, MA 02138, elamourcroteau@acentech.com)

I had no idea that a career in acoustics was even a possibility when I completed my bachelor's degree. I knew that I was fascinated by sound and that I wasn't ready to end my formal education. That curiosity led me to my first Acoustical Society of America (ASA) conference in Portland, Oregon, in May 2009, where I was quickly overwhelmed by the magnitude of the field. During that meeting, I was introduced to Bob Coffeen and his group of students and something immediately resonated with me (pun not intended). I didn't recognize it at the time, but that was the beginning of a journey that would take me from graduate school to professional consulting and eventually to the classroom as a professor. My work as an audiovisual consultant and educator is deeply shaped by Bob's teaching style and by how he interacted with us as people, the future of acoustics. I am proud to share Bob's passion and play a role in continuing his legacy.

2:45–3:00 Break

3:00

1pAA6. Development of the classroom acoustics booklet for the acoustical society. Ben Seep (Wave Eng., 1100 W Littleton Blvd #420, Littleton, CO 80120, bens@waveengineering.us) and Robin Glosemeyer Petrone (Threshold Acoust., Chicago, IL 60604)

In the mid-to-late 1990s, the acoustics profession began a heightened focus on improving acoustics in classrooms in the U.S., which led to the development of the ANSI S12.60 standard. As part of this movement, the ASA's Technical Committee on Architectural Acoustics requested proposals to prepare presentation materials on acoustics of learning facilities. Bob Coffeen was awarded the project and led a team of KU Architectural Engineering students in the development of a booklet on Classroom Acoustics as part of their senior studio project. Two authors of the booklet look back at the process of developing the booklet under Bob's mentorship.

3:20

1pAA7. Adventures with Bob Coffeen, teaching professionals about modeling, sound system design, and acoustic design, music and speech. Bruce C. Olson (AFMG Services North America LLC, 8717 Humboldt Ave. North, Brooklyn Park, MN 55444, Bruce.Olson@afmg.eu)

I had the distinct honor of collaborating with Bob for more than 20 years. We started as strangers brought together by a mutual desire to teach and ended up great friends. Collaborations over the years involved teaching the use of simulation and measurement tools to a diverse group of professionals. He and I created auralization source files and explored different ways to measure characteristics of rooms to both verify designs and give greater insight into the needs of acoustic simulation. I'll share a few miscellaneous highlights of other adventures as well.

3:40

1pAA8. Bob Coffeen—Friend, colleague, and teacher. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Joseph Keefe (Ostergaard Acoust. Assoc., Woodbridge, NJ)

Bob Coffeen was near and dear to so many of us. We will offer some tales and anecdotes about Bob that reach back over several decades, including some personal stories, professional society connections, computer modeling, and warm collegial relationships. Maybe most importantly, we hope to emphasize Bob's love for teaching, with the abundant benefits to his many devoted students and to audio and acoustics.

4:00–5:00 Discussion

MONDAY AFTERNOON, 11 MAY 2026 FRANKLIN HALL 5, 1:00 P.M. TO 3:50 P.M.

Session 1pAB

Animal Bioacoustics: Learning and Coordination of Vocal Signaling

Julia Hyland Bruno, Cochair
Biological Sci., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102

Eduardo Mercado, Cochair
Univ. at Buffalo, SUNY, 4360 Gunnville Rd., Clarence, NY 14031

Invited Papers

1:00

1pAB1. The social origins of vocal sequence learning in songbirds and human infants. Steven L. Elmlinger (Psych., Princeton, 113 Cranbury Rd., Princeton Junction, NJ 08550, elmlinger@princeton.edu), Samantha Carouso-Peck, Annalise Wilk, and Michael H. Goldstein (Psych., Cornell, Ithaca, NY)

From birdsong to human language, acoustic communication by vocal learners involves concatenation of sounds into sequences. Developmental compression of syllables into rapid sequences may reflect both social learning and motor maturation. Here, we test whether social feedback may guide the sequential structure of vocal forms. In study 1, 50 caregiver–infant ($M^{age} = 9.70$ months) dyads participated in 15-minute free-play. Analyses revealed that infant vocal sequences elicited significantly higher rates of caregiver verbal responses than single syllables. In an independent sample from study 1, study 2 followed 30 caregiver–infant dyads longitudinally from 5 ($M = 5.32$) to 10 months ($M = 9.74$). Caregiver responses to 5-month-old's vocal sequences predicted faster increases in syllable rate over development, which in turn predicted infants' vocabulary outcomes. In study 3, juvenile zebra finch elicited contingent social feedback on their song during vocal development. Early social feedback predicted finches' sequence compression and (2) compression only predicted song maturity (similarity to tutor) when birds received contingent social feedback to their immature vocalizing. The ability to change the immediate learning environment by producing early immature behavior guides vocal development across two distantly related species. Our study provides evidence of a shared social mechanism for vocal development in both songbirds and humans.

1:20

1pAB2. Timescales of plasticity: Vocal flexibility in a closed-ended learner. Julia Hyland Bruno (Biological Sci., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102, j.hylandbruno@gmail.com)

In theory, vocal learning allows for increased communicative flexibility compared with species whose vocalizations are not learned. Despite this, mechanistic research into vocal learning has been dominated by laboratory studies of the zebra finch, a closed-ended learner with stereotyped singing behavior. Relative to mimics and other songbirds with large or fluctuating repertoires, zebra finches sing simple songs: each male learns one song during sexual maturation and then repeats it for the rest of his life. However, extended recordings of individual adult birds reveal that mature song bouts exhibit acoustic variability above the level of the imitated motif. Here, I report on how a zebra finch's song bouts can vary across timescales ranging from seconds to years, focusing on the expressed bout repertoires and timesharing patterns of birds in dyadic interactions. While individual birds differ in the degree of variability within their bout repertoires, dyads sometimes exhibit repertoire convergence. Surprisingly, greater bout variability expressed by both birds is associated with more singing and rapid coordination of turns. These findings suggest that bout structure is under vocal control and indicate that zebra finch communication may be a tractable system for studying the co-emergence of vocal flexibility and imitative vocal learning during development.

Contributed Paper

1:40

1pAB3. Quantifying the development of infant–caregiver dyadic interaction using information-theoretic measures of vocal contingency. Gordon Ramsay (Dept. of Pediatrics, Emory Univ., Marcus Autism Ctr., 1920 Briarcliff Rd. NE, Atlanta, GA 30329, Georgia, gordon.ramsay@emory.edu)

Social communication is scaffolded on early vocal signaling and the emergence of dyadic interaction between infant and caregiver over the first months of life. A key component of this process is the establishment of vocal contingency—the mutual predictability of signaling and response—between mother and child. In this study, we use information theory to quantify the development of vocal contingency over time, as a potential biomarker for communication disorders. As part of an NIH Autism Center of Excellence (NIH P50 MH10029), we tracked vocal development in 45 infants later diagnosed with autism and 100 typically developing controls, using LENA recorders to collect all-day home audio recordings from each child every month from 0–24 months. Using multitaper harmonic analysis, we identified the onset and offset of each infant and caregiver vocalization. From these events, we calculated the transfer entropy, an information-theoretic metric of statistical causality, to derive a new index of directional vocal contingency between infant and caregiver, and caregiver and infant. Developmental trajectories for each infant–caregiver dyad, analyzed across groups, indicate that asymmetries in vocal contingency emerge naturally in typical development, with caregivers leading initially until turn-taking by infants begins, whereas this pattern is disrupted in autism.

Invited Papers

1:55

1pAB4. Are communicative functions of rodent ultrasonic vocalizations incidental? Eduardo Mercado (Univ. at Buffalo, SUNY, 4360 Gunnville Rd., Clarence, NY 14031, emiii@buffalo.edu)

Interest in the vocal repertoires of rats and mice has increased exponentially over the last three decades, largely because of the potential value of rodents as models of human disorders. To the extent that rodents' calls mediate social interactions, studies of their vocalizations can potentially shed light on neural mechanisms of vocal communication and the evolution of vocal interactions. If the primary function of rodents' vocalizations is something other than social communication, however, then proposed parallels between rodent and human vocal acts are more questionable. The ultrasonic sieving hypothesis proposes that rodents produce many ultrasonic vocalizations (USVs) primarily to enhance olfactory reception of pheromones. A corollary of this hypothesis is that social cues listeners extract from USVs may be the result of eavesdropping rather than involving intentional vocal communication. Determining whether rodent USVs contribute to active olfactory sensing is thus critical to interpreting the role USVs play in rodents' social interactions.

2:15–2:30 Break

2:30

1pAB5. Gesture–vocal interactions and its effect on acoustic output in a songbird. Christina Bardjis, Eric Tao, D'Juan Moreland, Kate Swett (Biology, Univ. of Pennsylvania, Philadelphia, PA 19104), and Marc Schmidt (Biology, Univ. of Pennsylvania, 431 University Ave., Philadelphia, PA, marcschm@sas.upenn.edu)

Natural communication often requires coordination of vocal production with pronounced body movements. However, musculoskeletal coupling between gestural limb movements and abdominal muscles are capable of impacting respiratory mechanics and thus destabilizing acoustic output. How such mechanical interactions are compensated to maintain stable vocal control remains unknown. Here, we investigate the highly stereotyped song of the brown-headed cowbird (*Molothrus ater*), which is associated with a dramatic postural display when directed toward conspecific males but not females. We evaluate the effect of wingspread on acoustic quality by careful quantification of several different acoustic features, including fundamental frequency, spectral entropy, syllable timing, song duration as well as phonatory pressure, from hundreds of song renditions performed both with and without display. We find that although some acoustic features, such as duration, pitch, and entropy, differed between social contexts, we observed no effect of wingspread amplitude on measured acoustic parameters or phonatory air-sac pressure. These findings suggest that social context modifies song acoustics, but that cowbirds can effectively compensate for wingspread-related perturbations during vocal communication in natural social interactions.

2:50

1pAB6. Abstract withdrawn.

3:10

1pAB7. Coordination of vocal signals in duetting wrens. Melissa J. Coleman (Dept. of Natural Sci., Scripps College, 925 N Mills Ave. Claremont, CA 91711, mcoleman@scrippscollege.edu), Nancy Day (Dept. of Psych., Whitman College, Walla Walla, WA), Carlos Rodriguez-Saltos (Dept. of Biological Sci., Illinois State Univ., Normal, IL), and Eric Fortune (Biological Sci., NJIT, Newark, NJ)

Male and female plain-tailed wrens sing duet songs in which participants alternate syllable production so rapidly, it sounds like a single bird is singing. This behavior requires that birds learn their own unique song syllables. The precise timing between individuals depends on sensory input from the sender that is matched with motor output from the receiver. We asked how auditory and visual cues influence duet production. To examine the influence of acoustic cues, we separated pairs of birds in cages and placed them at different distances, from 1 to 7 m apart. We found that the interval between both females and males syllables increased as a function of distance, suggesting both sexes use acoustic cues from their partner to coordinate the duet. Visual cues may also be used in the coordination of duet performances. To test if visual cues play a role in coordination of the duets, we placed a visual barrier between the two birds. Surprisingly, male but not female responses were affected by visual cues. We interpret these results to suggest that females have a greater role in "leading" duet performances, since females rely solely on acoustic cues from males for the coordination of duet singing.

3:30

1pAB8. Simple heuristics underpin complex and flexible chorusing interactions in túngara frogs. Luke C. Larter (The Inst. at Brown for the Environment and Society, Brown Univ., Dept. of Ecology, Evolution, and Organismal Biology, Providence, RI 02912, luke_larter@brown.edu)

Many fitness-enhancing behaviors require successful interactions with conspecifics and successful navigation of diverse social environments. In acoustic choruses, males call to attract females, and male call properties and call-timing interactions crucially impact their attractiveness. Furthermore, competing selection pressures scale in complex ways with rival density, necessitating that call properties and interaction patterns be adjusted on the fly within and across different social environments. Do such complex, flexible interaction patterns necessitate high-level cognitive processes and responses to highly specific conspecific cues? In túngara frogs, adaptive adjustments to calling behavior are in fact guided by surprisingly generic acoustic cues. This reliance on generic cues likely reduces computational costs and response latencies, as cue appraisal can be offloaded to peripheral auditory tuning and subcortical brain regions. However, these gains could be offset by increased error rates. Yet, in túngara frogs, certain generic cues act as reliable guides to rival behavior and social context, because the stereotyped nature of rival calls and calling strategies mean they predictably generate certain acoustic regularities in certain social contexts. This suggests that the cognitive architecture underlying complex social communication may be surprisingly minimalist, relying on "fast and frugal" perceptual heuristics that leverage predictable patterns in the social environment.

Session 1pAO

Acoustical Oceanography, Underwater Acoustics, Signal Processing in Acoustics, and Animal Bioacoustics: Acoustic Observations from Uncrewed Platforms II

Scott Loranger, Cochair
Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 86 Water St., Woods Hole, MA 02543

Elizabeth Weidner, Cochair
Marine Sci., Univ. of Connecticut, 1080 Shennecossett Rd., Groton, CT 06340

Derek Olson, Cochair
Oceanogr., Naval Postgraduate School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943

Contributed Papers

1:00

1pAO1. Observations of underwater ambient noise from wind and waves in the fall Arctic Ocean. Junsu Jang (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 98 Water St., Bigelow 4th Fl., MS#9, Woods Hole, MA 02543, junsu.jang@whoi.edu), Maddie Smith (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Jim Thomson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In open water, breaking surface waves entrain air and inject bubbles, whose oscillations are a primary source of underwater ambient acoustic noise. Sea ice formation fundamentally alters air-sea coupling by attenuating surface waves and suppressing the transfer of wind energy to the ocean surface. Simultaneous, co-located measurements of atmospheric forcing, sea ice and surface wave dynamics, near-surface bubble activity, and ambient sound were collected using a surface wave instrument float with tracking (SWIFT), a passive drifting platform deployed during a research cruise near Prudhoe Bay, Alaska, in November 2025. Open-water observations confirm the classic increase in ambient sound levels with increasing wind speed and additionally reveal a secondary dependence on wave development. This dependence appears to be related to relative bubble penetration depth, consistent with recent findings in the literature. Measurements collected under ice-covered conditions are markedly quieter than those in open water, reflecting reduced wind forcing, attenuated wave activity, and the absence of bubble plumes. The datasets are limited in duration and influenced by acoustic interference, including ship noise and mechanically induced noise. These observations demonstrate both the potential and the practical challenges of using uncrewed platforms to obtain multi-physics datasets for investigating coupled ocean–atmosphere–acoustic processes in ice-affected regions.

1:15

1pAO2. Passive acoustic identification tags to enhance underwater navigation accuracy of uncrewed platforms. Nizar Somaan (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, J. Erskine Love Bldg., Office 131 & Rm. 130, Atlanta, GA 30332, nsomaan3@gatech.edu), John Wilson (Mech. Eng., Georgia Inst. of Technol., Birmingham, AL), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Accurate navigation remains a critical challenge for uncrewed underwater platforms, particularly during homing and docking in optically opaque and acoustically cluttered environments. Passive acoustic identification (AID) tags provide an alternative to optical markers and active acoustic beacons by generating unique, frequency-dependent acoustic signatures without requiring onboard power or active transmission. Previous work demonstrated the feasibility of AID tags in scaled laboratory and field experiments [Soman *et al.* (2024), *JASA-EL* **4**(12)]. This study extends prior results through full-scale lake experiments using large-format, multilayer AID tags operating in the 100–600 kHz band. Tags with diameters up to 24 in. were interrogated using a custom impedance-matched transducer deployed from a surface vehicle equipped with a winch and gimbal system, enabling controlled 3-D scanning. Field measurements demonstrate reliable passive detection at ranges up to 50 m, exceeding earlier demonstrations. Energy budget analysis indicates the potential for detection at substantially longer ranges under favorable conditions. In addition to range validation, wide-angle spatial scanning combined with correlation-based processing was used to generate angular likelihood maps, enabling coarse tag localization. These results demonstrate the potential of passive AID tags to enhance autonomous underwater operations, such as docking, homing, and path tracking.

1:30

1pAO3. Real-time ambient noise monitoring using drifting and moored buoy systems. EeShan Bhatt (Appl. Ocean Sci., 5242 Port Royal Rd. # 1032, Springfield, VA 22151, eeshan.bhatt@appliedoceansciences.com), Julian Blanco (Appl. Ocean Sci., Newport, RI), Richard Campbell (Appl. Ocean Sci., Seattle, WA), James Murray (Appl. Ocean Sci., Stafford, VA), Kevin D. Heaney, and Christopher Verlinden (Appl. Ocean Sci., Fairfax Station, VA)

Real-time ambient noise data collection was demonstrated using Sofar Ocean Spotter buoys equipped with Applied Ocean Sciences' BOREALIS acoustic edge-processors in waters offshore Montauk, NY, in September 2025. Three drifting systems (10–15 NM offshore, 40–60 m depth) and one moored system (3 NM offshore, 20–30 m depth) transmitted deci-decade band noise levels every 30 minutes *via* iridium SBD and LTE telemetry, with data and descriptive statistics available *via* an online dashboard. Raw timeseries were available upon platform recovery. BOREALIS's extremely low power consumption enables indefinite solar-powered operation. All four systems collected high-quality, self-consistent acoustic data. Surface decoupling hardware on one drifter effectively reduced low-frequency noise. Acoustic observations agreed well with parabolic equation models incorporating AIS vessel traffic and NCOM wind fields, demonstrating feasibility for data assimilation into real-time noise prediction products. Multiple vessel detections showed consistent timing across platforms, indicating potential for coherent target localization. This demonstration shows how distributed networks of low-cost, solar-powered buoys can provide operationally relevant global ambient noise observations, substantially improving historical databases through real-time and forecasted products. [Work supported by NAVOCEANO.]

1:45

1pAO4. Variability of target strength distribution of mesopelagic organisms in the deep scattering layer. Zhaozhong Zhuang (MIT-WHOI, 266 Woods Hole Rd., Woods Hole, MA 02543, zhuangz@mit.edu) and Andone C. Lavery (Oregon State Univ., Woods Hole, MA)

Accurate quantification of mesopelagic communities from acoustic backscatter requires knowledge of *in situ* target strength (TS) distributions, yet such measurements at depth remain scarce. We deployed an upward-looking broadband split-beam echosounder (36–44 kHz) at ∼580 m depth in the Northwest Atlantic slope water for two years (2021–2022), tracking ∼820 000 individual organisms to ranges up to 45 m above the echosounder. The TS distribution reveals two persistent, acoustically distinct groups with mean TS of approximately –53 and –49 dB, stable across both years despite shifts in relative abundance. Vertically migrating organisms exhibited lower TS than non-migratory populations, with diel modulation driven primarily by the migratory group. Non-migratory organisms showed increasing TS with depth, consistent with pressure-dependent resonance of gas-bearing organisms. TS composition varied with water mass origin and changed markedly across warm-core ring boundaries, where the migratory group was notably absent at observation depths inside the eddy. The measured mean TS values are substantially higher than global assumptions used in biomass estimates, highlighting the necessity for region-specific TS calibration. These findings demonstrate that TS distributions vary systematically with depth, diel phase, and oceanographic context—factors that must be accounted for when converting volume backscatter to abundance estimates for ecosystem and biogeochemical applications.

2:00

1pAO5. Study of acoustic effects from seafloor depressions using an autonomous underwater vehicle. Robert W. Doerner (Mech. Eng., UCSD, 8820 Shellback Way, La Jolla, CA 92037, rodoerner@ucsd.edu) and Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA)

This study examines how seafloor depressions (pockmarks) and their accompanying disruptions to sub-bottom stratigraphy influence acoustic wave propagation and scattering. To estimate material properties of a representative pockmark on the New England shelf break, an inversion of sub-bottom data was performed using an autonomous underwater vehicle (AUV) equipped with an acoustic source and towed hydrophone array. The inversion resulted in spatially varying estimates of density and sound speed in the region surrounding the pockmark. An idealized pockmark model was then constructed, and its acoustic response was simulated by computing normal modes and pressure fields with the coupled-mode method. These results were compared to simulations of an uninterrupted stratigraphy to quantify overall propagation and scattering differences and mode-specific effects. Finally, a realistic stratigraphic model derived from the inversion was used to simulate propagation and scattering using a parabolic-equation model. These analyses provide insight into how pockmarks modify acoustic scattering, transmission loss, and mode coupling. [Work supported by the Office of Naval Research.]

2:15

1pAO6. Acoustic out-of-plane scattering off landslide bathymetry using an autonomous underwater vehicle at the New England shelfbreak. Natalie Kukshtel (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543, nkukshtel@whoi.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), and Andone C. Lavery (Oregon State Univ., Corvallis, OR)

During the New England shelf break acoustics (NESBA) experiment in May 2021, an acoustic autonomous underwater vehicle (AUV) was deployed to investigate bathymetric scattering processes at the shelfbreak. The AUV, consisting of a modified REMUS 600 vehicle with an onboard 2.5–4.5 kHz transducer and towed hydrophone array, passed over a submarine landslide site characterized by a steep escarpment and rocky debris field. AUV source signals received on nearby mooring hydrophones showed a complex interference pattern of overlapping bottom returns caused by this intricate bathymetry. A series of computational propagation models were tested to understand the bottom return data and scattering physics of this site. Ray tracing as well as 2-D and 3-D broadband parabolic equation (PE) models were used to simulate acoustic arrivals, which were then compared to the data for validation. While all models reproduced the primary features observed in the data, features attributed to out-of-plane 3-D scattering effects were best reproduced with the 3-D broadband PE model. [Work supported by the Office of Naval Research.]

2:30–2:45 Break

2:45

1pAO7. Localizing a mid-frequency source on a seamount using autonomous surface vehicles. McKenna Taylor (Georgia Inst. of Technol., 000714569 Georgia Tech. Station, Atlanta, GA 30332-1400, mtaylor363@gatech.edu), Davis Rider (Georgia Inst. of Technol., Atlanta, GA), Laurent Grare, Luc Lenain (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Karim G. Sabra (Georgia Inst. of Technol., Atlanta, GA)

The focus is to investigate sound propagation in the vicinity of the Atlantis II seamount. We examine a mid-frequency 2.5–4.5 kHz bandwidth LFM source mounted on A2SRC, 900 m below the surface, and acoustic vehicles are used to allow sampling of the spatial variability of the acoustic field. A question that arises is how precisely defined should the environment be in simulation to reasonably replicate the true wave physics in these regions. In the case of the Atlantis II seamount complex, there has been extensive research through the NESMA 2024-IOP2 cruise. The experiment features two time-synchronized moving unmanned autonomous surface vehicles on independent tracks, each with a towed acoustic module at depths of 75 and 9 m, respectively. Each receiver has a tetrahedral microphone array that is used to beamform and estimate the direction of arrival. A matched filter is used to extract the channel impulse response. Jointly, the arrays are coherently processed to localize the source. The typical range for detection is up to ∼20 km. Localization results are compared to Bellhop 2-D and 3-D simulations, focusing on single-bottom predictions, to quantify the environmental mismatch effects. A range-dependent geoacoustic model with 25 m bathymetric resolution is compared with a range-independent 1.02 km bathymetric resolution model. [Work supported by ONR.]

3:00

1pAO8. Wave property parameterization of wind-generated ocean surface noise source levels. Davis Rider (Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, drider3@gatech.edu), Laurent Grare (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), James S. Martin (Georgia Inst. of Technol., Atlanta, GA), Luc Lenain (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Karim G. Sabra (Georgia Inst. of Technol., Atlanta, GA)

Ambient noise monitoring in the ocean can provide insight into physical oceanographic processes, such as wind and wave generation. Additionally, accurate environmental parametrization of the sea surface noise source levels is needed for SONAR systems performance prediction. This presentation will present a method to estimate *in situ* the actual mid-frequency noise source level—and not just the sound pressure level—based on a quarter wave-length dipole mode and assess noise source level parametrization as a function of physical variables measured concurrently (*e.g.*, wind speed and wave field properties) To do so, an autonomous surface vehicle (liquid robotics wave glider, WG) instrumented for upper-ocean and marine-boundary-layer measurements and fitted with a compact four-element tetrahedral hydrophone array at 100 m depth was deployed for a 6-day experiment in the North Atlantic where winds varied from 1 to 12 m/s. Three-dimensional beamforming is used to isolate surface-generated noise in the 1–5 kHz band and is compared to a modeled spatial noise field to infer source levels produced by local sea-surface conditions. Estimated source levels are evaluated against conventional wind metrics (10 m wind speed, friction velocity) and against breaking-wave parameters (*e.g.*, breaking dissipation energy) inferred from the wave glider, yielding insight into surface-noise generation mechanisms not captured by standard metrics. [Work supported by ONR.]

3:15

1pAO9. Echofilter 2: A deep learning tool for automated air bubble and bottom detection in active acoustics. Angus Galloway (Benthic Lab., 192 Spadina Ave., Toronto, ON M5T 2C2, Canada, agalloway@benthiclab.com), Thomas M. Evans (DNRE, Cornell, Ithaca, NY), Hannah Blair (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY), Kade Keranen (DNRE, Cornell, Ithaca, NY), Jeremy Holden (ON Ministry of Natural Resources and Forestry, Peterborough, ON K9J 8M5, Canada), David M. Warner, Timothy P. O'Brien, Peter Esselman (Great Lakes Sci. Ctr., U.S. Geological Survey, Ann Arbor, MI), Suresh A. Sethi, Lars Rudstam (DNRE, Cornell, Ithaca, NY), and Scott Lowe (Vector Inst., Toronto, ON, Canada)

Echograms are image-like representations of acoustic backscatter amenable to machine vision analysis. Hydroacoustic surveys of biological organism abundance and biomass seek to isolate the backscattering intensity above the lake or seabed and below any entrained air bubbles near the water surface. Locating these non-biological scattering sources is often a bottleneck in producing fish biomass estimates. Commercial algorithms exist to automate fitting these boundaries; however, their outputs may require considerable manual adjustment and review. Echofilter (Lowe *et al.*, 2022) is a machine learning (ML) method that automatically fits entrained air and bottom exclusion lines directly from echograms. Echofilter was initially trained on upfacing echosounders moored in tidal energy streams in the Bay of Fundy, Nova Scotia, Canada, and it may perform poorly when applied to new datasets from different environments. Here, we discuss practical considerations in adapting Echofilter to define entrained air and bottom exclusion lines for echograms collected by downfacing echosounders carried by uncrewed surface vessels in the Great Lakes. We estimate that the updated Echofilter 2.0 algorithm reduced the time taken to preprocess a 90-day acoustic survey from 6–8 months to 2–3 weeks. Site-specific training data conferred the best performance, but requisite data volumes are much less than originally reported.

3:30

1pAO10. Concurrent physical and acoustical observations across the North Wall of the Gulf Stream with autonomous surface vehicles. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu), Davis Rider (Georgia Inst. of Technol., Atlanta, GA), Laurent Grare, and Luc Lenain (SIO, La Jolla, CA)

The rapid evolution of the physical properties of the upper ocean expected in regions with strong submesoscale features, such as the Gulf Stream's North Wall can lead to complex acoustic propagation effects. A fully instrumented autonomous surface vehicle—ASV (Wave Glider, Liquid Robotics) was deployed near the New England Sea Mounts and collected concurrent physical observations (*e.g.*, ADCP, CTD and air-sea interaction measurements) and GPS-synchronized acoustic observations using a compact 4-element hydrophone array at a depth of approximately 11 m. The ASV initially moved eastward along the Gulf Stream along a 200 km track passing over the Atlantis II Seamount and was ultimately directed northward to cross the North Wall. Energetic submesoscale features were observed along this track with notably strong horizontal gradients of the sound speed field while the ASV crossed the North Wall. During this track, low-frequency (500–600 Hz) transmissions from a remote moored acoustic source located in the SOFAR channel (∼depth 1200 m) were clearly received by the ASV's hydrophone during most of the track in both the Sargasso Sea and Slope Sea. The Crossing of the North Wall by the ASV was found to influence the amplitude of the long-range receptions. [Work Sponsored by ONR].

3:45

1pAO11. Volume scattering and acoustic single targets detected from VAMPIRE: A Lagrangian profiling vehicle with down- and horizontal-looking transducers. Benjamin D. Grassian (Univ. of Rhode Island, 45 Water St., Falmouth, MA 02543, bgrassian@gmail.com), Martha I. Ortiz-Torres (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Julian Blanco, Roy Gilboa (Univ. of Rhode Island, Narragansett, RI), Joseph Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., Southampton, NY), and Chris Roman (Univ. of Rhode Island, Narragansett, RI)

The mesopelagic environment is understudied relative to its size and ecological importance due to sampling challenges. Observations of key variables like mesopelagic animal abundance and biomass have high uncertainty in large part due to the unknown acoustic scattering properties (target strength, TS) of mesopelagic organisms which are difficult to measure using conventional sampling strategies. To help close this data gap, we have developed VAMPIRE, a 0–1000 m rated Lagrangian profiling vehicle equipped with a broadband split-beam echosounder (38, 70, and 200 kHz transducers), optical cameras, and environmental sensors. The acoustic transducers are actuated and can be oriented downward or horizontal to measure animal scattering properties and tracks from orthogonal sampling perspectives within a single survey. We present the first results from this acoustic sampling strategy from deployments in the Sargasso Sea and the Gulf of California from 0 to 600 m depth. We find significant TS distribution differences at specific depth bands between the horizontal and down-looking sampling perspectives useful to infer aspects of animal behavior, such as preferred depth ranges, orientation, and movement patterns within the measured environment. This work expands knowledge of the depth-dependent acoustic scattering properties of mesopelagic animals which are needed for accurate interpretation of acoustic measurements.

4:00

1pAO12. Statistical modeling of bio-physical interactions in the Gulf of California using a high-resolution acoustic profiling vehicle. Martha I. Ortiz-Torres (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, mi.ortizt@uri.edu), Benjamin D. Grassian (Biology, AOPE, Woods Hole Oceanogr. Inst., Falmouth, MA), Amy Maas (School of Ocean Futures, Arizona State Univ., St. George's, Bermuda), Brad Seibel (Univ. of South Florida, St. Petersburg, FL), and Chris Roman (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

The mesopelagic ecosystem in the Gulf of California (GOC) is characterized by a pronounced and shallow Oxygen Minimum Zone (OMZ) which influences animal distributions and behavior. While this region harbors significant biomass of midwater fishes and invertebrates, the environmental and biological drivers of their spatial and temporal distribution remain poorly understood, particularly regarding how organisms interact with sharp environmental gradients. This study leverages vehicle-borne acoustic data to investigate the partitioning of mesopelagic communities with respect to hydrographic variability resolved at high resolution using the Wire Flyer, a towed vehicle equipped with environmental sensors and an EK80 echosounder with side-looking 70 and 200 kHz transducers. From this data, we assessed the response of deep scattering layers to key environmental variables like oxygen, temperature, and salinity at meso- and submesoscales. Using Generalized Additive Models (GAMs) to capture multivariate and non-linear relationships, our results show how the structure of the midwater community responds to the environment, particularly to fine-scale gradients at the OMZ boundaries. These findings highlight the specific sensitivities of mesopelagic fauna to their physical environment and provide new insights into how midwater communities may respond to the expansion of OMZs in a changing ocean.

Session 1pBA

Biomedical Acoustics, Physical Acoustics and Signal Processing in Acoustics: Advanced Ultrasound Vascular Imaging and Vessel Characterization II

John M. Cormack, Cochair
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Zhiyu Sheng, Cochair
Medicine, University of Pittsburgh, 3550 Terrace St., Scaife Hall 624, Pittsburgh, PA 15213

Kang Kim, Cochair
Bioengineering, Univ. of Pittsburgh, 623A Scaife, 3550 Terrace St., Cardiology/Medicine, Pittsburgh, PA 15213

Invited Papers

1:10

1pBA1. Contrast-enhanced ultrasound for monitoring locoregional tumor therapy: Recent clinical guidelines and emerging applications. John Eisenbrey (Radiology, Thomas Jefferson Univ., 132 South 10th St., 796E Main Bldg., Philadelphia, PA 19107, john.eisenbrey@jefferson.edu)

Limitations on the sensitivity of blood flow detection on ultrasound can be overcome using microbubble-based ultrasound contrast agents. At appropriate transmit frequencies, these intravascular agents produce significant nonlinear components, which can enhance the vasculature by over 30 dB. Locoregional therapies (in which minimally invasive, image guided approaches are taken to destroy or disrupt tumor blood supply) are essential for the clinical management of a variety of solid tumors. However, assessment of treatment response using traditional cross-sectional imaging suffers from a variety of limitations including suboptimal sensitivity to residual blood flow, patient contraindications, and required delays between treatment and imaging follow-up. Contrast-enhanced ultrasound (CEUS) of tumor vasculature has been shown to overcome many of these key limitations in assessing locoregional therapy treatment response. Both 2-D and 3-D CEUS have been shown to provide statistically significant improvements in sensitivity for detecting residual hepatocellular carcinoma vascularity following transarterial chemoembolization. Ongoing work also suggests that CEUS may be a useful imaging tool for guiding chemoembolization within the operating room by proving immediate assessment of vascular changes during treatment. Work from parallel clinical trials has also demonstrated similar advantages of using CEUS to monitor renal cell carcinoma ablation, particularly when fused to pre-treatment cross-sectional imaging.

1:30

1pBA2. Non-invasive ICP monitoring and ischemia detection *via* transcranial and ocular ultrasound. Sophie Haddad (Radiology, Children's Hospital of Philadelphia, Philadelphia, PA, haddads@chop.edu), Yibo Wu (Johns Hopkins Univ., Baltimore, MD), Ladan Yazdani, Aref Ghaderi, Katherine Luo (Radiology, Children's Hospital of Philadelphia, Philadelphia, PA), Todd J. Kilbaugh (Children's Hospital of Philadelphia, Philadelphia, PA), Joseph Katz (Johns Hopkins Univ., Baltimore, MD), and Misun Hwang (Radiology, Children's Hospital of Philadelphia, Philadelphia, PA)

Hydrocephalus causes elevated intracranial pressure (ICP) that can lead to permanent neurologic disability. While current diagnosis relies on invasive monitoring with risks of infection and hemorrhage, the critical goal is not only measuring ICP but identifying which ICP triggers ischemia. Our research indicates that microvessels are highly sensitive to pressure and can serve as biomarkers for ICP and ischemia. We hypothesize that quantifying microvascular changes in the brain and eye could provide crucial insights into ICP and brain health in the neonatal porcine hydrocephalus model. In a neonatal porcine model, ICP was incrementally increased (10–50 mmHg) while performing contrast enhanced ultrasound *via* transcranial and ocular windows. Using Particle Tracking Velocimetry (PTV), we derived two biomarkers: Cerebral Microcirculation (CMC) and Retinal Microcirculation (RMC), defined as time-averaged velocity in microvessels. Our findings reveal that while macrovascular flow often persists under stress, CMC and RMC show a highly predictable linear decrease as ICP rises ($R2 = 0.91$ and $R2 = 0.84$, respectively). Moreover, drastic reduction in flow correlated with the onset of ischemia as measured *via* microdialysis based lactate to pyruvate levels. As such, our study suggests the potential utility of advanced ultrasound imaging of the brain and ocular microvessels to infer important physiologic information in neonatal hydrocephalus.

1:50

1pBA3. Contrast-enhanced ultrasound for the risk assessment of carotid plaques. Kibo Nam (Radiology, Thomas Jefferson Univ., 132 South 10th St., Philadelphia, PA 19107, kibo.nam@jefferson.edu)

Carotid atherosclerosis, mainly characterized by identifying plaque in the carotid artery, is evaluated to predict an individual's cardiovascular risk. However, clinical practice is focused on carotid stenosis severity not on plaque vulnerability. This is partly due to a lack of clinical consensus on how to evaluate carotid plaque vulnerability. This study aims to determine if contrast-enhanced ultrasound (CEUS) with the subharmonic-aided pressure estimation (SHAPE) can help with assessing the carotid plaque vulnerability. Twenty patients scheduled for carotid endarterectomy were imaged. Following the B-mode, 2-D and 3-D CEUS were performed with an infusion of Definity (Lantheus Medical Imaging). The collected ultrasound data were analyzed offline. The surrogate predictors for cardiovascular events were obtained from histological analysis of the plaque specimens collected after the endarterectomy. The absolute and relative contrast-enhanced plaque areas as well as the mean intensity of contrast-enhanced plaque areas correlated moderately with the histological vascularity. The pressure gradient between carotid artery and plaque had negative correlations with plaque vascularity, cap thickness, and intraplaque hemorrhage. The plaque calcification showed negative correlation with cap thickness. The intraplaque pressure by 3-D SHAPE showed weak correlations with plaque vascularity. This study demonstrated the feasibility of risk assessment of carotid plaques using CEUS.

2:10

1pBA4. Identifying hemodynamically significant carotid atherosclerosis: Something old, something new, and some opportunities for research and technology translation. Siddhartha Sikdar (Bioengineering, George Mason Univ., 4400 University Dr., MS 1J7, Peterson Hall, Rm. 3303, Fairfax, VA 22030, ssikdar@gmu.edu) and Kirk W. Beach (Seattle, WA)

The lifetime risk of stroke globally is 25%, and nearly two in three Americans will experience some level of cognitive impairment by age 70. The causal role of carotid atherosclerosis in ischemic stroke is well documented, and emerging evidence suggests that carotid disease is also associated with cognitive and mobility dysfunction. A long-standing challenge is that only a small subset of individuals with carotid atherosclerosis experience neurological events. Current methods for risk stratification, including recent advances to characterize vulnerable carotid atheromas, do not account for abnormalities in the downstream cerebral circulation that impacts hemodynamic forces on the atheroma. This talk will review findings from a series of recent studies of carotid hemodynamics and the downstream cerebral circulation to differentiate between high-risk "pressure-reducing" stenoses and lower-risk "flow-reducing" stenoses. These findings will be interpreted in the context of studies using oculoplethysmography and carotid phonoangiography five decades ago. The talk will also discuss new opportunities to develop novel screening tests using wearable ultrasound technology to identify high-risk populations in a primary prevention setting outside traditional diagnostic imaging centers.

Contributed Papers

2:30

1pBA5. Chaotic behaviors of an encapsulated bubble of ultrasonic contrast agents and its control. Tianye Zhang (Mech. Eng., Univ. of Michigan, 2350 Hayward, Ann Arbor, MI 48105, tianyezh@umich.edu)

Encapsulated ultrasound contrast-agent (UCA) microbubbles exhibit strong nonlinear dynamics that can transition from stable periodic oscillations to period-doubling and chaos as excitation pressure increases, degrading spectral specificity and contrast reliability. Here, we investigate a chaos-control strategy for UCA radial oscillations using an encapsulated-bubble model based on modified Herring dynamics. In the uncontrolled regime (control pressure Pc = 0P_c = 0Pc = 0), the simulated bubble response shows irregular, broadband behavior characteristic of chaos. We then apply a small periodic perturbation to the driving pressure and demonstrate deterministic stabilization: after control is activated, the chaotic oscillation is replaced by a repeatable low-frequency subharmonic response at f/4f/4f/4. Because subharmonic emissions are widely used to improve contrast-to-tissue ratios in UCA imaging, converting chaotic motion into a stable subharmonic regime provides a physics-based pathway to broaden the usable acoustic pressure window while preserving diagnostically valuable nonlinear signatures. The results motivate experimental validation of perturbation-based chaos control as a practical tool for improving the robustness of subharmonic imaging modes.

2:45

1pBA6. Acoustic micro-rheology *via* droplet cavitation dynamics. Sander Spiekhout (Univ. of Michigan, An Arbor, MI, spiekhou@umich.edu), Ze Qi Chan, Mario L. Fabiilli, and Mitra Aliabouzar (Univ. of Michigan, An Arbor, MI 48109)

Acoustic cavitation is increasingly explored for biomedical applications. Acoustic droplet vaporization (ADV) allows controllable cavitation by providing an easily nucleated source. The growth and collapse dynamics of ADV-generated bubbles are strongly affected by the mechanical properties of the surrounding microenvironment, offering potential as a rheometer. Here, we explore acoustic emissions of perfluorocarbon microdroplets encapsulated in fibrin hydrogels with varying elastic moduli(G). Droplets [$\varnothing 3 \pm 0.5$ μm, 0.01%(v/v)] with perfluorohexane (PFH) or perfluorooctane (PFO) were vaporized by a 3 cycle, 2.5 MHz pulse, with peak negative pressures swept from 2.2 to 5.9 MPa. Growth and collapse events were isolated from acoustic emissions detected by a perpendicularly placed 1 MHz receiver. For PFH droplets, both growth and collapse events were detectable in up to 80% of the measurements, maximizing at 4.3 MPa, while 99% of PFO events were detectable at 5.4/5.9 MPa[FM1] . Growth-collapse times for PFH were 16.3 ± 0.97 μs (G: $\sim$0.1 kPa, n = 12), 12.1 ± 2.02 μs (G: $\sim$0.8 kPa, n = 18), and 7.1 ± 0.88 μs (G: $\sim$3 kPa, n = 16) μs. For PFO droplets, growth-collapse times were 22.5 ± 0.53 μs (G: $\sim$0.1 kPa, n = 18), 16.9 ± 0.96 μs (G: $\sim$0.8 kPa, n = 30), and 9.2 ± 1.06 μs (G: $\sim$3 kPa, n = 16). These findings suggest that growth-collapse events from ADV can be acoustically detected to measure tissue stiffness.

3:00–3:15 Break

3:15

1pBA7. Advancing carotid plaque imaging: On-axis acoustic radiation force-based methods and emerging 3-D approaches. Caterina M. Gallippi (Lampe Joint Dept. of Biomedical Eng., The Univ. of North Carolina at Chapel Hill and North Carolina State Univ., 10010 Mary Ellen Jones Bldg., CB 7575, Chapel Hill, NC 27514, cmgallip@email.unc.edu), Shureed Qazi (Lampe Joint Dept. of Biomedical Eng., The Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill and Raleigh, NC 27514), Lian-xia Li, and Boyce E. Griffith (Mathematics, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC 27514)

Accurate, noninvasive characterization of carotid plaque structure and composition is essential for distinguishing patients at low risk of stroke from those who may benefit from carotid endarterectomy. Over the past several years, our group has advanced Acoustic Radiation Force Impulse (ARFI)-based imaging methods to assess key markers of plaque vulnerability, including fibrous cap thickness and regions of lipid-rich necrotic core (LRNC) and intraplaque hemorrhage (IPH). Notably, we introduced the Variance of Acceleration (VoA) parameter to enhance discrimination between LRNC and IPH and developed automated classifiers to streamline interpretation and analysis. While histological validation confirms the accuracy of VoA-derived features, correlations with patient symptomatology remain limited—likely due to the 2-D nature of VoA imaging and omission of other biomechanical factors such as wall shear stress and intraplaque pressure gradients. To address these gaps, our current efforts focus on enabling 3-D plaque interrogation using matrix array transducers and alternative volumetric imaging approaches, and integrating spatially resolved hemodynamic and mechanical parameters, particularly near the arterial wall where imaging clutter poses challenges. This lecture will highlight these innovations and their potential to transform risk stratification in carotid atherosclerosis.

3:35

1pBA8. Non-invasive mapping of arterial compliance and pulse pressure using physics-informed neural networks and ultrasound imaging: *In silico*, phantoms validation, and *in vivo* human feasibility. Tuhin Roy (Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, roytuhin22@gmail.com), Parth Gami (Biomedical Eng., Columbia Univ., New York, NY), and Elisa Konofagou (Columbia Univ., New York, NY)

Towards estimating arterial stiffness, this study presents a physics-informed neural network (PINN) framework for non-invasive, spatially resolved assessment of localized arterial compliance variation and pulse pressure using high-frame-rate ultrasound imaging. The PINN integrates Pulse Wave Imaging (PWI) and Vector Flow Imaging (VFI) data with 1-D differential equations of pulse wave propagation. Model effectiveness is shown in *in silico* simulations to physical plaque phantoms, and initial feasibility is demonstrated in human carotid arteries (five subjects: one healthy, four with stenosis) in a clinical setting. *In silico* and phantom studies demonstrated effective compliance and pulse pressure reconstruction. The robustness of the model with varied inlet-outlet boundary conditions and different sizes and stiffness of plaques is demonstrated in the *in silico* studies. The *in vivo* analysis showed uniform compliance in healthy subjects and focal compliance variations in stenotic patients, correlating with the corresponding ultrasound B-mode images. Wall motion reconstruction achieved $< 0.2\%$ error (healthy) and $< 0.48\%$ error (stenotic cases). Computation times (20 min for down-sampled data in uniform arteries to 1.5 hours for high-fidelity data in stenotic arteries) demonstrate future clinical applicability. The PINN framework enabled non-invasive, patient-specific mapping of localized arterial compliance and pulse pressure for atherosclerotic plaque risk evaluation and personalized cardiovascular assessment.

3:55

1pBA9. Evaluating capillary stiffness *in vivo* using microbubbles under ultrasound excitation as mechanical probes. Sae Jang (Medicine/Cardiology, Univ. of Pittsburgh, 200 Lothrop St., Scaife 969, Pittsburgh, PA 15206, jangsk@upmc.edu), Xucai Chen (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA), Spandan Maiti (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Brandon Helfield (Phys., Concordia Univ., Montreal, QC, Canada), and Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA)

We developed a new technique to study *in vivo* biomechanical properties of capillaries using microbubbles, ultrasound, and ultra-high-speed intravital microscopy. We hypothesized that diabetes causes capillary stiffening and that acoustic behaviors of microbubbles flowing within capillaries can quantify capillary stiffness. Type 1 diabetes was induced in rats using streptozotocin. Cremaster muscle was externalized in anesthetized rats. Definity® was intra-arterially injected. Capillaries were imaged microscopically at 8–12 Mfps, during delivery of one 6–15 cycle ultrasound pulse ($f = 1$ MHz, $PNP = 1.0$–2.0 MPa). Capillary and microbubble sizes during insonation were measured from the images. Fourier transform was used for frequency spectral analysis. Elastic modulus was derived using normal stress exerted by microbubble (linearized Euler's equation) and circumferential vessel Green's strain. Elastic modulus was higher in diabetic capillaries (2.3 MPa, $n = 4$) than in healthy capillaries (4.3 MPa, $n = 11$, $p = 0.01$). Microbubbles in diabetic capillaries displayed greater subharmonic/fundamental power ratio ($p = 0.02$) and ultraharmonic/fundamental power ratio ($p = 0.04$) compared with microbubbles in healthy capillaries. Capillaries stiffen in diabetic rats. The subharmonic/fundamental difference was significantly increased in diabetic capillaries, and correlated to elastic modulus, suggesting that distinct microbubble acoustic behaviors may provide a basis for a novel approach to diagnose microvascular disease.

4:15

1pBA10. Ultrasonic guided wave elastography for early detection of atherosclerosis: A novel three-layer k-wave simulation framework. Azin Nadi (Dept. of Mech. Eng., Texas Tech Univ., Box 41021, Lubbock, TX 79409, azin.nadi@ttu.edu) and Jingfei Liu (Mech. Eng., Texas Tech Univ., Lubbock, TX, azin.nadi@ttu.edu)

Early atherosclerosis produces subtle but significant mechanical changes in arterial walls that increase tissue stiffness through early structural remodeling. These alterations increase wall stiffness long before geometric narrowing becomes visible in standard ultrasound or MRI, limiting early diagnosis and increasing the risk of stroke. Guided wave elastography (GWE) is an emerging ultrasound-based technique that exploits dispersive elastic waves to infer arterial wall mechanical properties. Because experimental measurements in vascular systems are inherently complex at this early stage of development, physics-based simulation models are essential for identifying stiffness effects and establishing the fundamental wave–structure relationships required for future clinical implementations and interpretation. This study introduces a three-layer k-wave simulation framework to model guided wave propagation in a realistic vessel system comprising blood, the arterial wall, and surrounding soft tissue. To our knowledge, this is the first k-wave-based model designed to capture stiffness-driven changes in wave speed and dispersion associated with early atherosclerotic remodeling. Simulation results demonstrate that small increases in wall stiffness produce measurable shifts in phase velocity, with sensitivity observed in surface and circumferential modes. These results provide the physical insight needed to interpret complex experimental measurements and support the translation of vascular GWE toward clinically deployable systems.

4:30

1pBA11. Towards local viscoelasticity imaging in arteries *via* high-resolution wavenumber analysis of flexural waves. max Gattin (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, 1 Rue Jussieu, 75238 Paris Cedex 05, Paris 75238, France, max.gattin@espci.fr) and Julien de Rosny (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, Paris, France)

Shear wave elastography is widely used to image elastic properties in soft tissues. However, living tissues are known to be viscoelastic, and imaging this viscous component could provide richer diagnostic information. Yet, mapping viscoelasticity remains challenging: elastic waves are only weakly affected by attenuation but are strongly influenced by scattering and geometric spreading. Large blood vessels, such as the carotid artery, behave as homogeneous 1D waveguides acoustically, offering a simplified framework for wave analysis. Recent observations have shown that pulse waves from the heart generate not only symmetric compressional waves but also antisymmetric flexural waves, whose attenuation could potentially be used to infer the viscous component. In this work, we adapt a high-resolution wavenumber signal analysis method, known for its sub-wavelength resolution and noise resilience, to extract both real and imaginary parts of the wave number (elasticity + viscosity) from the flexural mode in custom-fabricated flexible tubes filled with water, designed to mimic the carotid artery. This high-resolution approach enables localized mapping of viscoelastic properties at sub-wavelength scales. This approach could lay the groundwork for local viscoelastic imaging of arterial walls, offering a complementary tool to existing biomarkers for assessing arterial health and cardiovascular risk.

4:45

1pBA12. Quantitative ultrasound characterization of vessel density: Effects of microbubble concentration. Parniyan Norouzzadeh (Mech. and Aerosp. Eng., North Carolina State Univ., 1840 Entrepreneur Dr., Raleigh, NC 27606, pnorouz@ncsu.edu), François Legrand (MAE, North Carolina State Univ., Raleigh, NC, pnorouz@ncsu.edu), Jadyn Cook (Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC, pnorouz@ncsu.edu), Mohamed Irfan (NC State Univ., Raleigh, NC, pnorouz@ncsu.edu), Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC, pnorouz@ncsu.edu), Michael Daniele (NC State Univ., Raleigh, NC, pnorouz@ncsu.edu), and Marie Muller (MAE, North Carolina State Univ., Raleigh, NC, pnorouz@ncsu.edu)

This work investigates quantitative ultrasound methods for assessing tumor-related angiogenesis through vessel density analysis. Because neovascular density correlates with tumor aggressiveness, reliable noninvasive metrics are clinically valuable. We analyze ultrasound backscattered signals from microbubbles within vascular networks to evaluate whether signal-derived parameters can differentiate vessel density in tissue-mimicking phantoms. Three cellulose tube phantom configurations were used to model controlled changes in vessel density: two with parallel tubes at different spacings and one with parallel and crossed tubes. Gelatin phantoms with embedded channels were also used to better mimic tissue, with spacing adjusted to represent different vessel densities. Phantoms were tested under five bubble concentrations. Ultrasound data were acquired with a Verasonics L11-5v linear array transducer and a 256 Vantage NXT scanner using full synthetic aperture, producing a 3-D inter-element response matrix from which random matrix theory parameters, including R^2 and Δ_{QC}, quantifying deviation from the quarter-circle law, were calculated. Significant differences in R^2 and Δ_{QC} were observed with varying bubble concentrations in gelatin phantoms ($p = 0.0073$ and 0.0214) and parallel-tube cellulose phantoms ($p = 1.1 \times 10^{-4}$ and 0.0017). R^2 increased with bubble concentration in parallel-tube phantoms but not in crossed-tube configurations, indicating sensitivity to phantom configuration and microbubble concentration.

Session 1pEAa

Engineering Acoustics and Physical Acoustics: 3-D Printing-Enabled Acoustic Transducers and Sensors

Ahmed Allam, Cochair
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Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

1:00

1pEAa1. Additive manufacturing of structurally integrated lead-free piezoelectric transducers. Satya K. Ammu (Dept. of Aerosp. Structures and Mater., Delft Univ. of Technol., Delft, Netherlands), Xianfeng Chen (Dept. of Precision and Microsystems Eng., Delft Univ. of Technol., Delft, Netherlands), Derin Ulcay (Dept. of Aerosp. Structures and Mater., Delft Univ. of Technol., Delft, Netherlands), Saurav Sharma (Dept. of Aerosp. Structures and Mater., Delft Univ. of Technol., Pierre van Hauwelaan 224, Delft, South Holland 2625 WL, Netherlands, sauravsharma227@gmail.com), Farbod Alijani, Peter G. Steeneken (Dept. of Precision and Microsystems Eng., Delft Univ. of Technol., Delft, Netherlands), and Kunal Masania (Dept. of Aerosp. Structures and Mater., Delft Univ. of Technol., Delft, Netherlands)

Multi-material additive manufacturing methods alleviate geometric limitations and enable the structural integration of smart materials with embedded sensing and actuation capabilities. Here, we present a direct ink writing (DIW) method that enables the manufacturing of piezoelectric actuators integrated with structural components, without requiring high-temperature sintering. By optimizing the particle size and rheology of the printing ink, we achieve tunability of the piezoelectric properties and demonstrate a superior d33·g33 coefficient compared to other DIW-manufactured actuators. Utilizing this process, we prototype fully integrated ultrasound transducers comprising piezoelectric vibrating membranes and structural supports printed in a single workflow. We analyze the impact of membrane dimensions on resonance behavior and out-of-plane displacement, demonstrating a direct correlation between scaled dimensions and increased output pressure. To validate the utility of these printed devices, we demonstrate a functional proximity sensor based on time-of-flight measurements, highlighting the potential of this fabrication method for customized biomedical and aerospace sensing applications.

1:20

1pEAa2. Transducers incorporating additive manufactured components for power ultrasonic applications. Andrew Feeney (James Watt School of Eng., Univ. of Glasgow, James Watt North Bldg., Glasgow G12 8QQ, United Kingdom, andrew.feeney@glasgow.ac.uk), Weihuan Kong, Ehsan Malekipour, Alicia Gardiner (James Watt School of Eng., Univ. of Glasgow, Glasgow, United Kingdom), Yuchen Liu (Nami Surgical Ltd., Glasgow, United Kingdom), and Margaret Lucas (James Watt School of Eng., Univ. of Glasgow, Glasgow, United Kingdom)

Additive manufacturing (AM) enables incorporation of complex metal structures into ultrasonic transducers that are unachievable by subtractive manufacturing. Hence it is possible to control their vibrational behaviour in new ways and deliver miniaturisation and light-weighting solutions. AM also offers opportunities to explore alloy compositions that improve transducer performance, for example, shape memory alloys that can deliver switchable or tuneable dynamics. We present some of the benefits and challenges associated with AM created transducers, focusing on bolted Langevin transducers, which used in a wide range of power ultrasonic applications. We introduce lattices, manufactured from Ti_6Al_4V feedstock, in the form of triply periodic minimal surface structures, into the front mass of BLTs, achieving large increases in displacement amplitude and new ways to implement hybrid-mode operation by controlling the geometry of the lattice unit cell. A major challenge is structural integrity. We show how ultrasonic fatigue and fracture behavior can be characterized using digital image correlation, and how structural integrity can be improved by post-processing. Preliminary results are also presented on incorporating AM Nitinol into ultrasonic surgical devices, aiming to create a device adaptable to different tissues by exploiting the transformation properties of Nitinol and the geometry freedoms enabled by AM.

1:40

1pEAa3. From nature to technology: Mosquito-inspired acoustic signal amplification. José Daniel P. Sar (Electron. and Elec. Eng., Univ. of Strathclyde, 15 Moir St., Glasgow, Scotland G15AE, United Kingdom, jose.pastor-sar@strath.ac.uk), Andrew Reid (Elec. and Electron. Eng., Univ. of Strathclyde, Glasgow, United Kingdom), and James Windmill (Univ. of Strathclyde, Glasgow, United Kingdom)

Over the last decade, sensors capable of capturing physical quantities from the environment have become fundamental across disciplines ranging from medicine to environmental engineering. Their development has advanced significantly due to technological improvements that have made devices smaller, more efficient, and more accessible. As a result, researchers have increasingly drawn inspiration from nature to integrate unique functionalities into emerging technologies. A notable example is active hearing in insect antennal systems. In mosquito antennae, Johnston's organ detects vibrations in the surrounding fluid caused by deflections of the flagellum and can generate self-oscillations that nonlinearly amplify incoming signals. In addition, mechanical features, such as antenna segmentation and feathery structures, enhance sensitivity and frequency selectivity. This study addresses the challenge of designing acoustic sensors capable of passively amplifying signals without relying on electronic amplifiers or digital signal processing. The results demonstrate that significant passive amplification can be achieved at the first eigenmode of the system, a capability previously considered attainable only through electronic devices or feedback algorithms. These findings show how biological mechanics can improve sensitivity and selectivity in low-power acoustic sensors, enabling bio-inspired designs with potential applications in biomedical technologies and ultrasensitive sensing systems.

Contributed Papers

2:00

1pEAa4. Rapid prototyping of thickness-mode ultrasonic transducers for high-efficiency broadband applications. Osama S. Tayel (Mech. and Mater. Eng., Univ. of Cincinnati, 201 Klotter Ave. C3, Cincinnati, OH, tayelos@mail.uc.edu), Yehia Zakaria, and Ahmed Allam (Mech. & Mater. Eng., Univ. of Cincinnati, Cincinnati, OH)

Modern ultrasonic applications, like histotripsy and wireless power/data transfer, require transducers with high efficiency and large bandwidth. Achieving both requires precise design of transducer layer stacks to maximize transmitted power and minimize reflections and loss. While numerical tools provide baseline designs, realizing high-performance transducers requires experimental iteration to overcome parasitic losses and material property uncertainty. This work explores additive manufacturing to accelerate experimental optimization of air-backed thickness-mode piezoelectric transducers in the MHz range. It compares additive manufacturing technologies, like FDM and SLA, for fabricating housing and establishing layer geometry—weighing trade-offs between accuracy and rapid iteration. Filament material and printer settings for FDM workflows are optimized for waterproofing and repeatability, allowing underwater characterization. This workflow evaluates a high-efficiency broadband link for simultaneous underwater power and data transmission. Link efficiency and bandwidth are characterized experimentally for different matching approaches; results are compared to high-fidelity simulations to assess uncertainty and maximize performance. The link serves as the basis for an ultrasonic wireless docking station for underwater autonomous vehicles.

2:15

1pEAa5. Enabling *in situ* crack detection *via* acoustic emission for blown powder laser directed energy deposition additive manufacturing. Emmeline Evans (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30308, eevans70@gatech.edu), Jin Yeon Kim (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Laurence Jacobs (Civil Eng., Georgia Inst. of Technol., Atlanta, GA), and Aaron Stebner (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Blown powder laser directed energy deposition (DED) is an additive manufacturing technique in which metal powders are blown into a melt pool created by a laser beam in order to deposit layers of metal on a substrate. Parts manufactured *via* powder DED are prone to stress cracking due to cyclic cooling and remelting of the part as layers are deposited. The objective of this research is to employ acoustic emission to detect and localize cracking in powder DED molybdenum test samples. This work presents the results of several enabling experiments. A study of the change in wave speed in the molybdenum substrate with changes in strain and temperature was conducted to improve time-of-flight calculations. A search study of printing parameters determined the set that will organically induce cracking, while a similar parameter search for data collection resulted in a framework for hardware selection and data acquisition parameters. Finally, the results of preliminary printing experiments using single-track wall specimens are presented. [Work supported by the NASA Space Technology Graduate Research Opportunities Grant 80NSSC23K1213.]

Session 1pEAb

Engineering Acoustics, Physical Acoustics and Structural Acoustics and Vibration: Piezoelectric Metamaterials

Michael R. Haberman, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Randall P. Williams, Cochair
Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

3:00

1pEAb1. Tunable anisotropy in piezoelectric truss metamaterials *via* data-driven design and 3-D Printing. Saurav Sharma (Aerosp. Eng., Delft Univ. of Technol., Pierre van Hauwelaan 224, Delft, South Holland 2625 WL, Netherlands, sauravsharma227@gmail.com), Satya K. Ammu (Aerosp. Eng., Delft Univ. of Technol., Delft, Netherlands), Prakash Thakolkaran, Jovana Jovanova (Mech. Eng., Delft Univ. of Technol., Delft, Netherlands), Kunal Masania (Aerosp. Eng., Delft Univ. of Technol., Delft, Netherlands), and Siddhant Kumar (Mech. Eng., Delft Univ. of Technol., Delft, Netherlands)

Piezoelectric materials are limited in the anisotropies they can exhibit due to their intrinsic crystal structure. Such tuneability is highly desirable for developing custom transducers with properties tailored for specific acoustic applications. Here, we present an integrated framework for designing and 3-D printing piezoelectric metamaterials with tunable electromechanical responses. To explore the vast design space and optimize the geometry and topology of the truss unit cell—the fundamental building block of these metamaterials—we employ a generative machine learning framework. For practical realization, we utilize a developed in-gel direct-ink-writing (DIW) method to print the structures using a photo-curable resin slurry infused with piezoelectric microparticles. Our optimized metamaterials demonstrate a 48% improvement in the specific hydrostatic figure of merit compared to bulk lead zirconate titanate (PZT), as well as the unconventional phenomenon of a transverse piezoelectric coefficient exceeding its longitudinal counterpart. This approach paves the way for next-generation customized transducers in micro-robotics, biomedical devices, and structural health monitoring.

3:20

1pEAb2. Nonreciprocity in spatially modulated nonlocal piezoelectric metamaterials. Muhammad Bilal Khan, Anastasia Quinn, Jasper Kern (Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ), and Christopher Sugino (Mech. Eng., Stevens Inst. of Technol., Castle Point, Hoboken, NJ 07030, csugino@stevens.edu)

Acoustic metamaterials can achieve effective properties that are impossible to obtain in conventional materials. Nonlocal interactions between unit cells further expand this design space by introducing wavenumber-dependent effective properties. Still, it is challenging to engineer nonlocal interactions in conventional metamaterial architectures. Piezoelectric metamaterials provide a simpler, more general, and more tunable way to implement nonlocal interactions *via* nonlocal shunt circuitry. Here, we theoretically and experimentally investigate the dynamics of a nonlocal piezoelectric metamaterial with spatially modulated nonlocal interactions. First, we derive the dispersion relation of a nonlocal metamaterial with periodic, spatially modulated nonlocal shunt circuits. We show that the hybridization between the electrical dispersion bands and the underlying elastic dispersion results in strong nonreciprocity, especially when the nonlocal gains are close, but unequal. In such cases, nonreciprocal destructive interference occurs over a larger section of the waveguide, resulting in stronger nonreciprocity. Through numerical studies, we show how the nonlocal spacing, gain, and modulation strength can be used to optimize nonreciprocity without increasing the number of unit cells in the system, increasing performance without introducing additional complexity. Finally, we experimentally validate our model predictions using a piezoelectric metamaterial with nonlocal voltage-controlled current sources.

3:40

1pEAb3. Spatially and spectrally programmable topological interface modes in a locally resonant piezoelectric metamaterial. Joseph Shedleski (Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30318, jshedleski3@gatech.edu), Sai Aditya Raman Kuchibhatla, Prabhakaran Manogharan, and Alper Erturk (Georgia Inst. of Technol., Atlanta, GA)

In this work, we experimentally demonstrate topological interface modes for bending waves in a locally resonant piezoelectric metamaterial domain, guided by numerical simulations from a high-fidelity electromechanical finite-element model. Using digitally controlled synthetic impedance shunt circuits, we break the translational symmetry of a periodic piezoelectric beam and induce robust energy localization consistent with a Su–Schrieffer–Heeger-like (SSH-like) arrangement, namely a locally resonant counterpart of the SSH chain enabled by and programmed *via* shunt inductors. Band folding introduced by resonant shunts generates sub-Bragg Dirac degeneracies, which are opened by alternating the inductance values of the shunt circuits. We experimentally validate subwavelength topological bandgaps and the resulting interface localization, as well as programmable spatial control of the localized modes through

reconfiguration of the electrical boundary conditions of the piezoelectric unit cells. Moreover, tuning the local circuit resonances enables frequency programmability of the interface modes. The resulting reconfigurable topological platform can enable performance enhancement for a wide range of applications, including acoustic communication, targeted energy transfer, and energy harvesting, which is further illustrated through a case study demonstrating enhanced energy extraction using a resistive electrical load placed near the interface with comparison for the absence and presence of the interface mode.

4:00

1pEAb4. Experimental and numerical investigation of tunable vibro-acoustic attenuation performance in an A3-sized piezoelectric metamaterial panel. Stefan Janssen (Dept. of Mech. Eng., Campus Diepenbeek, KU Leuven, Wetenschapspark 27, Diepenbeek 3590, Belgium, stefan.janssen@kuleuven.be), Camila S. Schmidt (Dept. of Aeronautical Eng., São Carlos School of Eng., Univ. of São Paulo, São Carlos, Brazil), Leopoldo P. De Oliveira (Dept. of Mech. Eng., São Carlos School of Eng., Univ. of São Paulo, São Carlos, Brazil), Carlos De Marqui Jr (Dept. of Aeronautical Eng., São Carlos School of Eng., Univ. of São Paulo, São Carlos, Brazil), Claus Claeys (Dept. of Mech. Eng., KU Leuven, Leuven, Belgium), and Elke Deckers (Dept. of Mech. Eng., Campus Diepenbeek, KU Leuven, Leuven, Belgium)

Piezoelectric locally resonant metamaterials (PLRMs) show promise as a tunable and lightweight approach for low-frequency noise and vibration control, overcoming the narrow bandwidth limitations of mechanical resonator-based LRMs. This study presents a combined numerical and experimental investigation of an A3-sized PLRM panel designed for tunable vibro-acoustic attenuation. The metastructure comprises an aluminum plate with a 9-by-9 array of piezoelectric elements, each individually connected to a tunable inductive-resistive shunt circuit. A unit-cell-based finite element framework, integrating Bloch mode synthesis model-order reduction, enables the efficient prediction of vibration attenuation and stop band behavior. The model is validated experimentally through transverse plate velocity measurements using a scanning laser Doppler vibrometer for multiple tuning configurations up to 1 kHz, showing excellent agreement with simulations. Acoustic radiation measurements further demonstrate that this vibration reduction translates into strong, targeted, and tunable structure-borne noise attenuation of 6–10 dB in targeted frequency bands. These results validate the proposed modeling strategy and demonstrate the practical potential of PLRM panels for adaptive noise and vibration control in medium-sized panel structures.

4:20

1pEAb5. Artificial piezoelectric metamaterials and actuation devices. Jikun Yang (Univ. of Sci. and Technol. Beijing, No. 30 Xueyuan Rd., Haidian District, Beijing 100083, China, jkyang@ustb.edu.cn)

Continuously enhancing the electromechanical characteristics of piezoelectric materials is essential for optimizing the performance of piezoelectric devices. However, due to the intrinsic ∞mm point-group symmetry of ceramics, the piezoelectric strain matrix of natural piezoelectric ceramics and corresponding electromechanical coupling modes are very limited, which further constrains the design and fabrication of piezoelectric devices. In addition to traditional modulation methods, such as doping, domain engineering, and defect engineering, which focus on the intrinsic properties of materials, the emerging field of piezoelectric metamaterials emphasizes the artificial design of piezoelectric structures. This novel approach has proven to be a powerful pathway for achieving counterintuitive and enhanced piezoelectric responses. The construction of arrayed piezoelectric strain units with programmed polarization and external electric fields can surpass their individual properties, and enable the excitation of arbitrarily controllable artificial vibration modes and fully nonzero piezoelectric coefficients. As a result, this innovative design methodology holds great potential for advancing the development of piezoelectric devices, including precise actuation and ambient energy harvesting.

4:40

1pEAb6. Intelligent acousto-electrical metamaterials (IAM) for sound source localization. Victor Couedel (Mater. Sci., Univ. of California, Berkeley, Hearst Memorial Mining Bldg., Rm. 150, Berkeley, CA 94720, vcouedel@berkeley.edu), Haotian Lu (Mater. Sci., Univ. of California, Berkeley, Berkeley, CA 94720), Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332), and Xiaoyu Zheng (Mater. Sci., Univ. of California, Berkeley, Berkeley, CA)

Conventional acoustic transducers rely on piezoelectric crystals with fixed responses, necessitating bulky, high-power arrays for directional sensing. We introduce a new class of Intelligent Acousto-Electrical Metamaterials (IAM) featuring 3-D micro-architected topologies that enable structure-programmed piezoelectricity. Unlike static crystals, these metamaterials leverage geometry-driven charge transport to selectively amplify or reverse charge flow based on an incident wave's frequency and direction. This architecture allows a single transducer to perform frequency-dependent beam-pattern tunning and directional sensing without mechanical adjustments or complex wiring through local unit-cell sensitivity variations. Also, by integrating machine learning, these transducers can be trained to automatically perform real-time localization of multiple moving sound sources. This paradigm shift from chemical-defined to geometry-defined coupling enables compact, adaptive sensing systems for autonomous vehicles, underwater robotics, and medical imaging.

Session 1pED

Education in Acoustics: Sound Science in Informal Learning Settings

Keeta Jones, Cochair
Acoust. Society of America, 1305 Walt Whitman Rd., Ste. 110, Melville, NY 11747-4300

Donnelley Hayde, Cochair
COSI, 333 W Broad St., Columbus, OH 43215

Takayuki Arai, Cochair
Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan

Chair's Introduction—1:00

Invited Papers

1:05

1pED1. Modeling participant responses to environmental audio recordings. Daniel Shanahan (Bienen School of Music, Northwestern Univ., 70 Arts Circle Dr., Evanston, IL 60208, daniel.shanahan@northwestern.edu)

This presentation discusses a platform that trains a series of machine-learning models (specifically neural networks and random forest-based approaches) to predict perceived emotional qualities (derived from Russell's Circumplex Model of Emotion) in audio recordings of informal learning spaces, such as museums, zoos, and nature centers. In this framework, participants listen to a wide array of audio samples (from approximately 80 000 field recordings of similar informal learning spaces) and rate the perceived emotional content across multiple dimensions, including cepstral coefficients, amplitude, pitch, and event density. These ratings then inform a predictive model that maps acoustic features onto reported ratings of specific emotions. This tool will hopefully inform decisions made by exhibit designers seeking to incorporate insights into how visitors respond to the auditory environment of their learning spaces. In addition to discussing the specifics of the model, its performance, and its accuracy, this talk will also address how such approaches can address methodological challenges in auditory perception research while producing actionable tools for practitioners and creating a sustainable research infrastructure that bridges academic investigation and real-world application.

1:25

1pED2. Perceptions about sound in museums, zoos, and parks, and their effect on visitor experiences. Justin R. Meyer (Ctr. for Res. and Evaluation, Ctr. of Sci. and Industry, 333 West Broad St., Columbus, OH 43215, jmeyer@cosi.org), Donnelley Hayde, Laura Weiss, and Joe Heimlich (Ctr. for Res. and Evaluation, Ctr. of Sci. and Industry, Columbus, OH)

This talk will present results from a collaborative, NSF-funded project on the effects of sound in informal learning settings, from the perspectives of the project's visitor and informal learning research team. Using a during-visit interactive recording activity and post-visit intercept surveys, we collected self-reported data from over 1000 visitors from eight different areas (two each from a science museum, a zoo, a botanical garden, and a nature park in Columbus, Ohio). As part of this data collection, we asked visitors how they felt about the sounds in the areas they visited. We found that while positive-coded attitudes about sounds appear to have a strong, positive effect on overall attitudes about a visitor's experience, negative-coded attitudes do not appear to have a strong effect either way. In keeping with our project's intention to advance knowledge in ways that support informal learning practitioners, this talk will invite discussion of both theoretical implications and professional applications for our findings.

Contributed Paper

1:45

1pED3. Soundscapes and self-directed learning in outdoor settings. Hanbing Liang (Dept. of Planning and Landscape Architecture, Univ. of Wisconsin–Madison, 925 Bascom Mall, Madison, WI 53706, hliang8@wisc.edu)

Outdoor environments, such as university campuses, school grounds, and parks, are increasingly used for learning, yet sound is often treated as a problem rather than a factor shaping cognition and well-being. This study examines how soundscapes influence self-directed learning in outdoor settings. Semi-structured interviews with 51 students at a large U.S. University were analyzed using grounded theory. Participants engaged in activities ranging from routine coursework to creative and reflective work, with natural sounds and low-level human activity perceived as supportive, whereas intelligible conversation and mechanical noise were distracting. Thematic analysis revealed a process model describing how soundscape conditions mediate motivation, engagement, and withdrawal in self-directed outdoor learning. Findings further identified two mechanisms

through which soundscapes shaped learning experiences. The utility of sound supported learning by masking distractions, priming activities, and pacing tasks, while its effects fostered presence, safety, and restoration. Participants emphasized that outdoor soundscapes often provided a productive balance between quiet and stimulation, supporting concentration and reflection more effectively than many indoor environments. These findings demonstrate that outdoor soundscapes actively shape learning processes rather than serving as passive background conditions, highlighting opportunities for campus planning and urban design to support cognition, restoration, and lifelong learning.

Invited Papers

2:00

1pED4. Abstract withdrawn.

2:20–2:35 Break

2:35

1pED5. Towards fun activities in speech science: A few examples of museum exhibitions and workshops. Takayuki Arai (Information and Commun. Sci., Sophia Univ., 7-1Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

Some of our museum exhibitions and workshops are discussed, especially our vocal-tract models and digital pattern playback, or DPP. Our vocal-tract models were originally designed for formal teaching settings, and they have been widely used in science classes on speech and hearing and acoustic phonetics. The models tell us in an intuitive way how speech sounds are produced. Soon after the release of a commercial set of the models, one was installed at a science museum, and its success prompted several other museums to utilize several different versions as parts of their permanent or temporary exhibitions. They have also been used in outreach activities, such as science fairs and workshops, especially for children. DPP is an effective educational tool that can convert a printed spectrogram into sound. Activities based on vocal-tract models and DPP help to draw people's interest to speech science. [Work supported by the JSPS KAKENHI Grant No. 24K06423 and the Sophia University Special Grant for Academic Research (Research in Priority Areas).]

2:55

1pED6. Linguistics at schools and public events: University of Arizona. Natasha Warner (Univ. of Arizona, Dept. of Linguist., Box 210025, Tucson, AZ 85721-0025, nwarner@email.arizona.edu) and Eric Jackson (Univ. of Arizona, Tucson, AZ)

The Department of Linguistics at the University of Arizona has developed an unusually active public outreach program. The focus of the program is primarily on middle and high schools and on public events, such as science/book festivals. Although our outreach program is for Linguistics as a whole, we have found that acoustics and phonetics lend themselves especially well to hands-on, participatory activities that work for a wide age range, even in chaotic environments with high background noise. As a result, the majority of our activities provide acoustics education, and linguists in other subfields in our department also conduct acoustics education. In this talk, we will present activities we use (*e.g.*, ultrasound, waveform nametags) and discuss what has worked well in various venues. We will also discuss how we have built up opportunities to participate in outreach events, including STEM-themed events that do not always seek out social sciences topics. Informal observation suggests that such outreach activities can be very useful for spreading knowledge of acoustics and linguistics in the general public.

Contributed Paper

3:15

1pED7. Vibration playground: Learning sound science through multisensory interaction. Nicolas Sowers (Acoust., Penn State Univ., 1424 11th Ave. Ste. 400, Seattle, WA 98122, nicolas@timbrearchitects.com)

Acoustic phenomena are often difficult to grasp in informal learning settings because sound waves are invisible, time-based, and typically explained through abstract representations. Vibration playground explores how interactive, multisensory visualization can support intuitive engagement with sound science in public spaces, such as museums, science centers, and libraries. The project is a web-based platform featuring interactive demonstrations of acoustic wave phenomena, ranging from simple harmonic oscillators to vibrating strings and solid materials. Unlike static exhibits or looped animations, vibration playground enables real-time exploration: users can pluck or strike a virtual string with a touch on a mobile device, while visual motion and synthesized sound respond

immediately. This direct coupling of action, motion, and sound reinforces physical relationships underlying wave behavior without requiring formal mathematical training. Each demonstration includes optional technical references, supporting learners with varying levels of prior knowledge. All demonstrations are open source and developed using p5.js, a JavaScript library used by artists, making the platform adaptable to diverse, informal learning contexts. Designed to be extensible by acousticians, educators, and sound artists, vibration playground offers a flexible framework for making sound science engaging, accessible, and memorable beyond the classroom.

Invited Paper

3:30

1pED8. Calling all ages: How the Penn State student chapter uses demonstrations to engage the Happy Valley community. Erin Nguyen (ASA Student Chapter, The Penn State Univ., 414 Eng. Collaborative Res. and Education Bldg., 556 White Course Dr., University Park, PA 16802, een5106@psu.edu), Kourtney Libenow, Ruth Willet, Laura G. Brownstead (ASA Student Chapter, The Penn State Univ., University Park, PA 16802), Colin Flavin (ASA Student Chapter, The Penn State Univ., Crane, IN), and Joshua Langlois (ASA Student Chapter, The Penn State Univ., State College, PA 16802)

The Penn State University student chapter of the Acoustical Society of America (ASA) participates in many outreach programs throughout the year. A large part of the mission as an organization is to share acoustics with the community and promote science to people of all ages. To do this, the ASA students travel to a variety of events in the State College area and showcase numerous engaging and interactive acoustic demonstrations. Typical events include Centre County public school STEM/STEAM nights and Exploration-U as well as Penn State's Haunted-U. At each event, volunteers from PSU ASA set up and share the physics behind several different acoustics demos to attendees, specifically catering to whichever age group is there at that moment. Last December at ASA Honolulu, PSU ASA had the pleasure of participating in the Acoustics Demonstration Session where just a fraction of the fun demos were displayed. This presentation will go more into detail about how the organization promotes acoustics in the community and display even more of the educational and exciting demos used to inspire children and adults to study science and acoustics.

Contributed Paper

3:50

1pED9. Introducing the public to acoustics and amateur radio through a "science on tap" style presentation with interactive and visual demonstrations. Ruth Willet (Penn State, 14 Eng. Collaborative Res. and Education Bldg., 556 White Course Dr., University Park, PA 16802, raw5930@psu.edu), Kourtney Libenow, and Grace M. Wood (Penn State, University Park, PA)

As acousticians, we are always looking for creative ways to introduce our field of study to the general public. In the fall of 2025, the PSU ASA Student Chapter was invited to cohost a Pub Chat ("Science on Tap" style demonstration event) at The Dead Canary brewery in our community with the PSU Amateur Radio Club. Amateur radio is a hobby that allows licensed operators to communicate short and long distances around the world using radio waves, and it has many overlapping technical concepts with acoustics. This Pub Chat was given by members of PSU ASA and blended the two lesser-known studies into one seamless talk that was engaging for all ages. The audience ranged from elementary school students to retired professionals. This work will show how we introduced these two fields by grouping technical concepts into categories that were inspired by the ASA Technical Committees and how we involved the local community by integrating interactive and visual demonstrations into the presentation. Audience engagement indicated that the talk was particularly impactful during the interactive demonstrations. PSU ASA plans to pursue more of this style of outreach in the future to continue bringing acoustics to people in our community.

Session 1pID

Interdisciplinary and Student Council: Undergraduate and Graduate Program in Acoustics (Poster Session)

Caner Baydur, Cochair
College of Design and Innovation, Tongji Univ., Siping Rd. No. 1239, Shanghai 200092, China

Robert W. Drinnan, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6785

Jade F. Lopez Case, Cochair
Ocean Eng., Univ. of Rhode Island, 215 S Ferry Rd., Narragansett, RI 32837

Benjamin Masters, Cochair
Systems Design Engineering, University of Waterloo, 200 University Ave. W, Waterloo N2L 3G1, Canada

All posters will be on display from 1:00 p.m. to 4:00 p.m. Authors of odd numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even numbers papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Invited Papers

1pID1. Making waves: Inside Penn State's Graduate Program in acoustics. Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Established in 1965, the Penn State Graduate Program in Acoustics is the only civilian institution granting graduate degrees in Acoustics in the United States. Students have the option of earning their degrees in-person at the University Park campus (M. Eng., M.S., or Ph.D.) or entirely online through the distance education program (M. Eng. degree only; launched in 1987). Almost all classes are offered simultaneously to resident and distance education students, with recordings available for those who cannot attend the live classes. We have over 40 faculty from multiple disciplines that conduct world-renowned research and teach courses in architectural acoustics, structural acoustics, nonlinear acoustics, biomedical ultrasound, signal processing, acoustic metamaterials, underwater acoustics, noise and vibration control, transducers, aeroacoustics, computational acoustics, structural acoustics, *etc.* Our ∼900 alumni are employed worldwide in government labs, academic institutions, consulting firms, consumer audio, and other acoustics-related industries. Please visit our poster to learn more about opportunities to learn or collaborate with the Graduate Program in Acoustics at Penn State!

1pID2. Graduate acoustics education in the Cockrell School of Engineering at The University of Texas at Austin. Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman (Walker Dept. of Mech. Eng. & Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX, haberman@utexas.edu), Neal A. Hall, Mark F. Hamilton, and Preston S. Wilson (Univ. of Texas at Austin, Austin, TX)

While graduate study in acoustics takes place in several colleges and schools at The University of Texas at Austin (UT Austin), including Communication, Fine Arts, Geosciences, and Natural Sciences, this poster focuses on the acoustics program in Engineering. The core of this program resides in the Walker Department of Mechanical Engineering (ME) and the Chandra Family of Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and nine graduate acoustics courses are taught in ME and ECE. Course instructors include staff at Applied Research Laboratories at UT Austin, where many of the graduate students have research assistantships. The undergraduate course, which is taught every fall, begins with basic physical acoustics and proceeds to draw examples from different areas of engineering acoustics. Three graduate courses are taught every year: a two-course sequence on physical acoustics and an acoustic transducers course. The remaining six graduate acoustics courses, nonlinear acoustics, underwater acoustics, ultrasonics, architectural acoustics, wave phenomena, and acoustic metamaterials, are taught every two years. An acoustics seminar is held most Fridays during the long semesters, averaging over ten per semester since 1984. More information about the program can be found at https://www.texasacoustics.org/.

1pID3. Graduate education and research in architectural acoustics at Rensselaer Polytechnic Institute. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu), Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), and David Kahn (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Stamford, CT)

The Graduate Program in Architectural Acoustics has been constantly advancing from its inception in 1998 with an ambitious mission of educating future experts and leaders in architectural acoustics. These advancements are to stay on top of the rapid pace of change in the fields of architectural-, physical-, and psycho-acoustics, and noise-control engineering. For many years the program's pedagogy

use of "STEAM" (science, technology, engineering, art, and mathematics) methods has proven to be effective and productive, including intensive, integrative hands-on experiments that integrate architectural acoustics theory and practice. The graduate program has recruited graduate students from a variety of disciplines including individuals with B.S., B. Arch. or B.A. degrees in Mathematics, Physics, Engineering, Architecture, Electronic Media, Sound Recording, Music and related fields. RPI's graduate students have gone on to lead successful careers in this rapidly changing field. RPI's Graduate Program in Architectural Acoustics has graduated more than 180 graduates with M.S./ Ph.D. degrees. Under the guidance of the faculty members they have also actively contributed to the program's research in architectural acoustics, communication acoustics, psycho-acoustics, signal processing in acoustics as well as scientific exploration at the intersection of cutting-edge research and traditional architecture/music culture. This paper illuminates the evolution and growth of the Graduate Program.

1pID4. Opportunities in physical acoustics at the University of Mississippi: Graduate studies and undergraduate research in physics and engineering. Joel Mobley (Dept. of Phys. and Astronomy and National Ctr. for Physical Acoust., Univ. of Mississippi, P.O. Box 1848, 108 Lewis Hall, University, MS 38677, jmobley@olemiss.edu), Cecille Labuda (Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS), Likun Zhang, Ashoka Karunarathne (Dept. of Phys. and Astronomy and National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS), and Nathan E. Murray (National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

The University of Mississippi is a Ph.D. granting institution with an R1 Carnegie designation placing it among schools with the highest level of research activity. The Department of Physics and Astronomy and the School of Engineering offer a broad range of research and educational opportunities in Physical Acoustics. Our acoustics program is affiliated with the National Center for Physical Acoustics (NCPA), which is an 85000 square foot standalone facility on our campus solely dedicated to the physics and engineering applications of acoustics. NCPA has research groups dedicated to ultrasound, infrasound, aeroacoustics, atmospheric propagation, porous media, materials characterization, and ocean acoustics. Graduate students in both physics and engineering are pursuing Ph.D. and M.S. degrees at NCPA, and many of our faculty maintain research laboratories in the facility. In addition to graduate programs, NCPA provides a broad range of research opportunities for undergraduate STEM majors, as well as summer research open to external participants.

Contributed Paper

1pID5. Acoustics education and research opportunities at University of Massachusetts Dartmouth. David A. Brown (ECE, Univ. Massachusetts Dartmouth, 151 Martine St., Ste. 123, Fall River, MA 02723, dbrown@umassd.edu), John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), and Paul J. Gendron (ECE, Univ. of Masschusetts Dartmouth, Dartmouth, MA)

Acoustics graduate education and research opportunities at the University of Massachusetts Dartmouth are primarily in the College of Engineering with a strong focus in Electrical Engineering and its applications. Faculty offer graduate and upper level undergraduate courses associated with the Bachelors, Master and Ph.D. degrees in Electrical Engineering, capstone projects, theses, and dissertation in acoustic related studies. Areas of expertise and courses include signal and array processing, sonar, communications, detection and estimation, transduction, sensors and arrays, underwater acoustics and medical ultrasonics. Complimentary offerings in the areas of electromagnetics, radar, RF phonics, optics, and electronics offer students a broad education opportunity in applied science. Much of the research at UMass Dartmouth has been marine focused for over three decades, and the campus includes unique facilities including a large Underwater Acoustic Calibration Test Tank, School for Marine Science and Technology, ocean water access, and Unmanned Underwater Vehicles that support our projects. The Office of Naval Research is our largest sponsor of research and many of the alumni go on to work at US Navy laboratories, non-profit research centers, industry and local companies focusing on acoustic technology.

Invited Papers

1pID6. Acoustics research led by undergraduates and faculty at the University of Hartford. Heui Young Park (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, hpark@hartford.edu) and Christopher M. Jasinski (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, West Hartford, CT 06117)

The University of Hartford's Programs in Acoustics allow for undergraduate students to conduct research in a wide range of topical areas. Students may determine their own research objectives or seek the direction of a faculty member as independent study. Students in the B.S.M.E. Acoustics program and in the B.S.E. Acoustics and Music program regularly participate in research. Recent work has been completed in the technical committee areas of Architectural Acoustics, Musical Acoustics, Computational Acoustics, Education in Acoustics, and Noise. Research has also included advanced methods of data analysis, designing and prototyping, measurement, simulation, and subjective testing, with several research projects either interanally or externally funded. Every student conducting research also meets in a team setting on a regular basis to engage in interactive group dynamics, encouraging learning from others and being aware of varied research focuses. There are no graduate students in acoustics at the University of Hartford, meaning that every research project is led by undergraduate students and faculty. This presentation will provide an overview of several current research projects, how the research aligns with the acoustics curriculum, and a description of how the Acoustics programs at the University of Hartford provide a unique setting and experience for students.

1pID7. MIT music technology. Mark Rau (Music and Theater Arts & Elec. Eng. and Comput. Sci., Massachusetts Inst. of Technol., 77 Massachusetts Ave. Bldg. W18-Rm., Cambridge, MA 02139, mrau@mit.edu)

The Massachusetts Institute of Technology recently launched a graduate program in Music Technology. Available degrees include a thesis-based Master of Science and a course-based Master of Applied Science, offered through the Music Department, as well as a Ph.D. through the Department of Electrical Engineering and Computer Science. Postdoc opportunities exist as well, but are not consistent

year-to-year. Course and research areas include music software, audio signal processing, music and AI, music and audio hardware, digital instruments, human-computer interaction, acoustics, and creative design. For more information, please visit https://musictech.mit.edu/.

1pID8. The Nebraska acoustics group: Graduate studies in architectural acoustics and noise at the University of Nebraska–Lincoln. Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska–Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu) and Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska—Lincoln, Omaha, NE)

At the University of Nebraska–Lincoln (UNL) within the College of Engineering, students can pursue undergraduate and graduate (Master's or Doctoral) degrees in Architectural Engineering with a focus in acoustics. The Nebraska Acoustics Group, led by Dr. Lily Wang and Dr. Sam Underwood (https://go.unl.edu/acoustics), resides within the Durham School of Architectural Engineering and Construction and is based at UNL's Scott Campus in Omaha. This poster reviews the graduate-level acoustics courses, lab facilities, research interests, and achievements of our faculty, graduates, and students. There is currently both an Acoustical Society of America (ASA) student chapter based on the Scott Campus, as well as a new Nebraska regional chapter of the ASA. The Nebraska Acoustics Group also regularly collaborates with colleagues at Boys Town National Research Hospital, also located in Omaha.

1pID9. Physics and acoustics opportunities for undergraduates at High Point University. Eric Rokni (Phys., High Point Univ., One University Parkway, High Point, NC 27268, erokni@highpoint.edu)

High Point University (HPU) is a primarily undergraduate institution with a physics program that integrates acoustics into both coursework and research. Students can take courses such as *Physics of Sound and Music* and *Physics of Medical Imaging*, which explores ultrasound principles. These classes are accompanied by research spanning both musical and biomedical acoustics. Physics students engage in research projects from their first year and can participate in funded summer research through the Summer Research Program in the Sciences (SuRPS). Our modern labs and advanced equipment provide hands-on opportunities, giving students a strong foundation for graduate studies and careers in physics, acoustics, and related fields.

1pID10. Undergraduate and graduate education in ocean acoustics at the Scripps Institution of Oceanography, University of California San Diego. Julien Bonnel (Scripps Inst. of Oceanogr., 9500 Gilman Dr., # 0238, San Diego, CA 92093, jubonnel@ucsd.edu) and Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA)

Scripps Institution of Oceanography (SIO) at UC San Diego offers comprehensive education in ocean acoustics at the undergraduate, M.S., and Ph.D. levels. Instruction and research span the physics of sound propagation and scattering, acoustical oceanography, marine bioacoustics, acoustic signal processing and data science, and the development of innovative acoustic technologies. Students receive rigorous training and research experience, including unique opportunities for at-sea fieldwork, through close collaboration with world-renowned faculty in a highly interdisciplinary environment, contributing to a deeper understanding of the oceans. Ocean acoustics education at SIO is embedded within its undergraduate majors and graduate curricular groups, such as Applied Ocean Sciences and Biological Oceanography. Strong links also exist with the UC San Diego Jacobs School of Engineering, particularly with the Mechanical and Aerospace Engineering (MAE) and the Electrical and Computer Engineering (ECE) departments. Each year, SIO runs a week-long "acoustic bootcamp" to introduce students, postdocs and early-career scientists to hands-on experience with underwater acoustics technology in laboratory and field settings.

1pID11. Education and workforce development in acoustics at the Center for Acoustics Research and Education at the University of New Hampshire: Graduate certificate in acoustics and microcredentials. Laura Kloepper (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, 230 Spaulding Hall, Durham, NH 03824, laura.kloepper@unh.edu), Jennifer Miksis-Olds, Anthony P. Lyons, and Ian T. Jones (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, Durham, NH)

The Center for Acoustics Research and Education (CARE) and the University of New Hampshire (UNH) have invested in initiatives to strengthen acoustics education and workforce development. Over the past five years, CARE has expanded its educational mission in response to increasing demand for formal training in basic and applied acoustics across the physical and life sciences. The Graduate Certificate in Acoustics (GCA) was developed to provide graduate students and working professionals with a rigorous foundation in acoustics while allowing flexibility for disciplinary specialization. The GCA consists of a curated set of four semester-based, graduate-level courses with elective courses drawn from signal processing, biology, underwater acoustics, machine learning, and seabed mapping. Our microcredentials program offers short, competency-based learning pathways resulting in one of four digital badges: Introduction to Ocean Acoustics, Bioacoustics, Digital Signal Processing, or Underwater Acoustics. Both of these programs have options for in-person or remote completion. Together, the UNH CARE Graduate Certificate and Microcredentials programs offer a flexible pathway for graduate students and professionals seeking formal training in acoustics.

1pID12. Underwater acoustics and ocean engineering at the University of Rhode Island. Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu), James H. Miller, Gopu R. Potty, and Jennifer Amaral (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

The University of Rhode Island (URI) is home to the nation's first ocean engineering program and offers graduate and ABET-accredited undergraduate degrees with a strong emphasis on underwater acoustics. All undergraduate students complete a course in underwater acoustics, and many pursue acoustics-focused capstone projects and research. Graduate programs include both thesis and non-thesis masters degree options in addition to the Ph.D. program. A Graduate Certificate in Underwater Acoustics is also available. Research areas span acoustical oceanography, propagation modeling, geo-acoustic inversion, marine mammal acoustics, ocean acoustic instrumentation, transducers, and signal processing. The program is based at URI's Narragansett Bay Campus, providing direct access to Narragansett Bay, and is undergoing significant renovation to expand research capabilities. New facilities include a state-of-the-art Ocean Robotics Lab featuring a robotics tank, scheduled to open in Spring 2026, and an advanced acoustics tank currently in development. These additions will enhance hands-on learning and support cutting-edge research in ocean engineering and acoustics.

1pID13. Ocean acoustics education in the Massachusetts Institute of Technology/Woods Hole Oceanographic Institution Joint Program. John Colosi (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., WHOI, MS 11, Woods Hole, MA 02543, jcolosi@whoi.edu), Gil Averbuch, Kenneth G. Foote, Xavier Mouy, Yongsung Park, and Daniel P. Zitterbart (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA)

The Massachusetts Institute of Technology (MIT) and the Woods Hole Oceanographic Institution offer a Joint Program (JP) in ocean sciences. This is postgraduate only, leading to masters and doctoral degrees awarded jointly by both institutions. Represented disciplines are Applied Ocean Science and Engineering (AOSE), Biological Oceanography, Chemical Oceanography, Marine Geology and Geophysics, and Physical Oceanography. Because of the prominence of acoustics in both operational applications and research, there is a defined acoustics curriculum. This is mostly but not exclusively conducted within AOSE. Students take classes at MIT and WHOI, pursuing research in both traditional and novel areas, further facilitated by the seagoing resources of WHOI. Research subjects span propagation and scattering, navigation, communication, remote sensing, bioacoustics, measurement and inference of environmental properties from the air–water or ice–water interface to the sub-bottom, robotics, and soundscape, among others. Seawater environments span coastal and deep waters; tropics, mid-latitudes, and polar regions; hydrothermal and other vents; with both active and passive sensing. It is the explicit ambition of the JP to enable understanding of underwater sound for career application or research.

1pID14. The acoustics program at Georgia Tech. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

The acoustics education program at the Georgia Institute of Technology (Georgia Tech) is run by members of the Acoustics and Dynamics research area group from the School of Mechanical Engineering. We will present the scope of this program in terms of education at the undergraduate and graduate level as well its associated research activities and core facilities.

MONDAY AFTERNOON, 11 MAY 2026 FRANKLIN HALL 10, 1:00 P.M. TO 5:05 P.M.

Session 1pPA

Physical Acoustics and Noise: Atmospheric Turbulence Effects on Sound Propagation

Alexandra Loubeau, Cochair
NASA Langley Res. Ctr., MS 463, Hampton, VA 23681

Kent L. Gee, Cochair
Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602

Chair's Introduction—1:00

Invited Papers

1:05

1pPA1. Current status of sound propagation in a turbulent atmosphere. Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), D. Keith Wilson (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), and Michael Muhlestein (U.S. Army Engineer Res. and Development Ctr., Hanover, NH 03755)

The current status of sound propagation and scattering in a turbulent atmosphere is briefly reviewed. The review begins with the spectra of temperature and wind velocity fluctuations in the atmospheric boundary layer. It moves on to the second-order statistical moments of acoustic signals such as: the sound scattering cross section; the variance and correlation functions of the log-amplitude and phase fluctuations; the angle-of-arrival fluctuations; and the spatial, temporal, and cross-frequency coherences. The statistical moments are expressed in terms of the temperature and velocity spectra, and then are calculated for the von Karman and Kolmogorov spectral models and compared with the experimental data for vertical and slanted propagation. Correspondence between propagation in a turbulent atmosphere and a discrete medium (such as a forest or shrubland) is emphasized. The classical and recently derived probability density functions of acoustic signals are presented. Lastly, recent developments in numerical simulation of atmospheric turbulence and sound propagation through it are described.

1:25

1pPA2. Lessons learned in refining atmospheric profile and turbulence modeling for predicting supersonic en route noise. Joshua L. Kapcsos (Grad. Prog. in Acoust., Penn State Univ., 411 ECoRE Bldg., 556 White Course Dr., University Park, PA 16802, jlk642@psu.edu) and Victor W. Sparrow (Grad. Prog. in Acoust., Penn State Univ., University Park, PA)

Turbulence affects the propagation of supersonic en route noise in the atmospheric boundary layer and may be considered in international certification schemes for upcoming civil supersonic aircraft. Since its original public release in 2018, Penn State has been refining the realism of the atmosphere used in the turbulent propagation code KZKFourier. This presentation gives an overview of the code changes thus far. While the original version only took homogeneous atmospheres into account, modules have been modified to support the input of humidity and temperature profiles. Further, both subtypes of vector turbulence were revised for realism: a ground-blocking methodology has been applied to the buoyancy-induced vector turbulence model, and a revision to the behavior of friction velocity has been made to the shear-induced vector turbulence model. While these refinements were implemented individually, a near-term goal is to include all the new features in tandem. [Work supported by the FAA through ASCENT Project 57 under the Office of Environment and Energy (AEE). Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

1:45

1pPA3. Tools for simulating atmospheric turbulence effects on supersonic X-59 acoustic pressure signatures. Alexandra Loubeau (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov), Robin S. Downs, Joseph D. Fletcher (Environ. Measurement & Modeling, Volpe National Transportation Systems Ctr., Cambridge, MA), Jacob Klos, and William Doebler (NASA Langley Res. Ctr., Hampton, VA 23681)

NASA has continued to develop its capabilities to simulate atmospheric turbulence effects on sonic boom propagation. New KZKFourier simulations have been conducted of nonlinear propagation through the atmospheric boundary layer under a variety of expected weather conditions. Updated input conditions, including source waveform, azimuth angle, and extended propagation distances have been investigated. Ensembles of FIR filters have been designed using these results to provide a more computationally efficient estimate of turbulence effects on ground waveforms. These filters have been implemented into PCBoom, a software package maintained by NASA and available to the general public upon request. These filters require only a small fraction of the KZKFourier run time and exhibit low errors when results are compared to the original KZKFourier simulations, using model-to-model verification test cases and calculation of human perception noise metrics. Challenges with applying the filters to new cases have been addressed by designing longer filters and adjusting processing methods to increase efficiency. The methods will be used to estimate noise metric variability due to atmospheric turbulence during the NASA Quesst mission community tests with the supersonic X-59 aircraft. This variability will be used to quantify uncertainty in noise exposure of survey respondents.

2:05

1pPA4. Turbulence effects on shaped sonic booms using finite impulse response filter banks. Joseph D. Fletcher (US DOT Volpe Ctr., 220 Binney St., Cambridge, MA 02142-1093, joseph.fletcher@dot.gov), Robin S. Downs (US DOT Volpe Ctr., Cambridge, MA 02142), Jacob Klos, William Doebler, Joel B. Lonzaga, and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA 23681)

Numerical simulations of sonic boom propagation through the atmosphere can quantify turbulence effects on ground waveforms, but such simulations are computationally expensive. Finite Impulse Response (FIR) filters were previously developed for estimating turbulence effects on *N*-waves using the KZKFourier nonlinear turbulence propagation code as part of the NASA Sonic Booms in Atmospheric Turbulence test (SonicBAT, 2020). Downs *et al.* (ASA Spring 2023 meeting) modified that method for compatibility with shaped booms. In this work, large numbers of turbulized waveforms were randomly selected from an existing database of shaped-boom propagation results and banks of FIR filters were developed for each simulation condition. This approach centers on calculating ensembles of loudness metrics and statistics of interest from filtered waveforms resulting from convolution with the FIR filter banks. One benefit of this approach is that specific loudness metrics and statistics of interest need not be decided upon prior to generation of ensemble filter banks. Additionally, the ensemble method metrics are approximately twice as accurate as the previous FIR filter approach relative to the benchmark KZKFourier simulations, while still saving several orders of magnitude of computation time over KZKFourier simulations. These new shaped-boom filter banks were implemented in version 7.5 of NASA's PCBoom software.

2:25

1pPA5. Using finite impulse response filters to model turbulence effects across a range of propagation distances. Robin S. Downs (US DOT Volpe Ctr., 220 Binney St., Cambridge, MA 02142-1093, robin.downs@dot.gov), Joseph D. Fletcher (US DOT Volpe Ctr., Cambridge, MA), Jacob Klos, William Doebler, and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA 23681)

Ground waveform loudness metrics, including atmospheric turbulence effects, were simulated for an X-59 shaped boom carpet in two ways. The first method used the KZKFourier nonlinear turbulence propagation modeling code developed at The Pennsylvania State University as part of the NASA Sonic Booms in Atmospheric Turbulence test (SonicBAT, 2020), and the second used NASA's PCBoom version 7.5 with built-in shaped boom finite impulse response (FIR) filters. PCBoom results compared with the KZKFourier benchmark showed reasonable agreement across the inner carpet region but diverged near the lateral edges of the carpet. Rays comprising those regions propagate long distances through the atmospheric boundary layer, which challenges the design space of the FIR turbulence filters. Supplemental KZKFourier simulations were conducted at long propagation distances, extending the design space from 6.7 km up to 11.4 km. This presentation concentrates on efforts to reconcile the differences in turbulence effects modeled with KZKFourier versus those estimated using FIR turbulence filters. Examining turbulized waveforms propagated through long distances at higher turbulence conditions shows longer-duration low-frequency effects that are not accurately reproduced by the 230 ms FIR filters developed previously. Increasing filter lengths and modifying the previous approach shows promising results.

2:45–3:05 Break

3:05

1pPA6. Examining the relationship between atmospheric turbulence and metric variability at NASA Quesst Mission Phase 2 Dry Run. Kaylee Nyborg (Dept. of Phys. and Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, kaylee.nyborg@byu.net), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

In preparation for the NASA Quesst Mission Phase 2 acoustic validation flights of the X-59 aircraft, NASA conducted a dry run flight test with F-15 aircraft. Complementary to NASA instrumentation deployed to measure the sonic booms, a ground-based, 2-D microphone array, ARray Instrumentation for Sonic Thump Observations in TurbuLEnce (ARISTOTLE), was used to measure 19 of the sonic booms in the Mojave Desert, CA during summer 2025. ARISTOTLE included a total of 61 microphones arranged as a cross with 50 ft spacing between microphones and a denser grid of microphones in the center with 25 ft spacing. Turbulence parameters have been estimated from measured meteorological data. In this paper, the turbulence parameters are correlated with physical properties of the waveforms, such as spatial correlation length, rise time, and high-frequency spectral slopes. With a deeper understanding of the sonic boom waveforms' variability, the turbulence conditions are then related to variation in human perception-based metrics across the array. [Work supported through a NASA Graduate Research Fellowship.]

3:25

1pPA7. Solver for complex-valued scattering wavenumber for sonic booms in turbulence. William Doebler (NASA Langley Res. Ctr., NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov) and Joel B. Lonzaga (Flight Vehicle Acoust. Branch, National Aeronautics and Space Administration, Hampton, VA)

Multiple scattering theory (MST) has recently been applied to sonic boom propagation through atmospheric turbulence to predict the ensemble-averaged waveform [J. Acoust. Soc. Am. **154**, 3078–3088 (2023)]. In that work, the mean acoustic wavefield was expressed in terms of the acousto-turbulent self-energy, whose contribution modifies the effective wavenumber, with the dispersion relation defined by the zeros of the mean-field denominator. The present work focuses on obtaining these scattering wavenumbers numerically in the presence of both velocity and sound-speed fluctuations, a combination that cannot be treated analytically within the original formulation. A Newton–Raphson iteration scheme is used to solve the complex dispersion relation for exponential turbulence, yielding the turbulence-modified wavenumber predicted by MST. This work also extends the formulation to von Kármán turbulence, which was not treated in the original study, enabling evaluation of scattering behavior for an additional turbulence spectrum. The resulting scattering wavenumbers provide a basis for comparing theoretical predictions with previous N-wave sonic boom waveforms obtained in a flight test, and for quantifying how turbulence influences the mean field and the associated uncertainty in sonic boom loudness.

3:45

1pPA8. Turbulence-driven losses in sonic booms: A multiple scattering theory-based mean energy framework. Joel B. Lonzaga (Flight Vehicle Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St. (MS 463), Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

This paper discusses how the existing nonlinear parabolic equation used to account for turbulence effects conserves the mean acoustic energy of sonic booms when averaged over an ensemble of turbulence realizations. Because, the equation preserves the mean energy, it also preserves the mean loudness. In contrast, flight-test measurements consistently show that the mean loudness of sonic booms decreases during propagation through atmospheric turbulence, with the reduction increasing with both propagation distance and turbulence strength. These observations indicate that atmospheric turbulence scatters acoustic energy in ways not captured by the parabolic-equation framework. To address this discrepancy, the paper introduces a new method based on the multiple scattering theory for computing the mean energy spectral density that incorporates turbulence-induced backscattering, providing a more complete description of energy loss. Using this formulation, the predicted mean energy spectral density and loudness level are compared directly with those obtained from sonic booms measured during flight testing. The comparison demonstrates the extent to which backscattering must be represented to achieve accurate predictions. Developing a predictive capability that captures these turbulence effects is essential for reliable loudness estimation of NASA's X-59 low-boom flight demonstrator and for improving future sonic-boom propagation models.

4:05

1pPA9. Using a multiple-scales approximation approach to sound propagation in small-scale turbulent flows. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., US Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil)

Atmospheric turbulence can influence outdoor sound propagation in many ways. Since most of the energy of turbulence is accounted for by large-scale eddies and very little in the small-scale eddies, it is common to assume that the paraxial approximation (*e.g.*, geometrical acoustics) is applicable. However, since the smallest eddies in the atmosphere are frequently on the order of millimeters and wavelengths of interest can be on the order of centimeters or meters, this assumption is not always strictly valid. We are interested in quantifying the impact of the low-energy, high-wavenumber components of the turbulent spectrum on outdoor sound propagation. In this presentation we use a multiple-scales approximation of the dynamic equations and develop an effective-medium theory to estimate the effective mass density and compressibility of a plane wave propagating through a turbulent atmosphere, with an emphasis on the impact of small-scale turbulent flows.

4:25

1pPA10. Correlating atmospheric turbulence with acoustic camera beam wander. Michele L. Eggleston (Phys. and Astronomy, Brigham Young Univ., Brigham Young University, Provo, UT 84604, megglest@byu.edu), Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Matthew J. Kamrath (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Sergey N. Vecherin, D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH), and Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

This presentation discusses a follow-up experiment of a previous measurement, where improvements were implemented to get better data. These experiments were conducted to determine a correlation between atmospheric turbulence and beam wander. Measurements were conducted at the Bonneville Salt Flats in Utah with a Nor848B acoustic camera and a dual-diaphragm compression driver serving as the sound source emitting simultaneous frequencies at 500, 1000, 2000, 4000, and 8000 Hz. The source and acoustic camera were positioned approximately 64 m apart, with the source approximately elevated 7.5 m and the camera 1.8 m above ground level. A weather station, situated 8 m above the ground, was placed to the side of the propagation path to monitor atmospheric conditions. Measurements were carried out at various times of the day to capture variability in different weather conditions. Observations using the acoustic camera have revealed changes in the standard deviation of the angle of arrival in different turbulence conditions. This paper presents a statistical analysis of the acoustic image wander over time, and correlations between these variations and weather conditions. These statistics are then compared with previously developed theoretical predictions.

4:45–5:05 Discussion

MONDAY AFTERNOON, 11 MAY 2026 FRANKLIN HALL 8, 1:00 P.M. TO 4:25 P.M.

Session 1pPP

Psychological and Physiological Acoustics: Processing of Affective Sounds: Intersection of Audition and Emotion

Fatima T. Husain, Cochair
Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, 901 South Sixth, Champaign, IL 61820

Monita Chatterjee, Cochair
Commun. Sci. & Disorders, Northwestern Univ., Frances Searle Bldg., 2240 N. Campus Dr., Evanston, IL 60208

Chair's Introduction—1:00

Invited Papers

1:05

1pPP1. Neural bases of affective sound processing in individuals with tinnitus and hearing loss. Fatima T. Husain (Speech and Hearing Sci. and Beckman Inst., Univ. of Illinois Urbana–Champaign, 901 South Sixth, Champaign, IL 61820, husainf@illinois.edu)

In a series of studies, we have investigated the processing of affective sounds in the conditions of tinnitus and hearing loss when compared to matched controls using active, task-based (*e.g.*, Carpenter-Thompson *et al.*, 2014; Husain *et al.*, 2014) and passive listening fMRI (Shahsavarani *et al.*, 2021). We have used innovative functional connectivity analysis to delineate the functional coupling of regions within a network and across different networks in healthy controls and compared them to those with hearing loss and/or tinnitus. The primary findings from these studies illustrates the interconnected networks meditating emotion processing of sounds—attention networks, salience networks, the limbic system (particularly the amygdala and the hippocampus) and the auditory cortices. The studies underscore the importance of extra-auditory networks in sound processing in not only those with age-associated hearing loss or tinnitus and in developing potential treatments for these conditions. The studies also broaden our understanding of sensory perception in general and the importance of considering neural networks outside the canonical auditory system to understand auditory perception.

1:20

1pPP2. Single- and dual-task listening effort for word and emotion recognition using emotional and non-emotional stimuli. Shae D. Morgan (Otolaryngol., Univ. of Louisville, 400 S. Preston St., 4th Fl., Louisville, KY 40202, shae.morgan@louisville.edu)

Everyday speech perception is rarely restricted to a single domain but involves simultaneous processing of multiple cues available in the speech signal. For example, in addition to extracting the meaning of words from lexical cues, listeners also process indexical cues, such as the talker's age and emotional state. Many studies of speech perception focus on a single domain of perception, such as word recognition or emotion recognition in isolation. Fewer studies investigate how listeners process multiple cues simultaneously. In this presentation, we will discuss how cognitive resource allocation or "listening effort" (measured using pupillometry) varies during speech perception depending on whether listeners focus on what is said (word recognition), how it is said (emotion recognition), or both simultaneously. In the present study, young adults with normal hearing listened to emotionally expressive or emotionally neutral sentences and completed three tasks in separate blocks: Identifying target words from the sentence, identifying the talker's emotion, or identifying both. Using pupillometry, we measured pupil dilation in response to these tasks. This presentation will discuss the results within the context of emotional versus neutral and single-task (word recognition, emotion recognition) versus dual-task (both) speech processing. [Work supported by ASHFoundation New Investigator Research Grant.]

1:35

1pPP3. Amplitude modulation and perceived valence of nonspeech sounds among adults with and without hearing loss. Haiping Huang (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, haiping.huang.aud@gmail.com), William L. Martens (Macquarie Univ., Sydney, New South Wales, Australia), and Erin M. Picou (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN)

Amplitude modulation was previously associated with emotion perception of nonspeech sounds. Both short (1.5 s) and long (6 s) versions of nonspeech sounds from the International Affective Digitized Sound corpus have been used in the literature. Yet, it is uncertain whether sound duration affects rated valence in response to these sounds or if it affects the association between amplitude modulation and valence ratings. Therefore, for both short and long versions of nonspeech sounds, we extracted and compared amplitude modulation differences across pleasant, unpleasant, and neutral categories. We then correlated these acoustic findings with valence ratings, which were provided by adult listeners with and without hearing loss. Results showed similar patterns of acoustic differences for short and long versions of sounds. Pleasant sounds are associated with weaker amplitude modulation at lower frequency bands, while the opposite for higher frequency bands. Unpleasant sounds showed the inverse pattern. Interestingly, the association between amplitude modulation and valence rating showed that adults with normal hearing tend to provide lower valence ratings when amplitude modulation at the lower-frequency bands was high, but the opposite was true for listeners with hearing loss. This trend was generally consistent for both short and long versions of sounds.

1:50–2:00 Break

2:00

1pPP4. Effects of cochlear implant spectral degradation on affective sound processing. Kelly N. Jahn (Speech, Lang., and Hearing, The Univ. of Texas at Dallas, 800 W Campell Rd., Richardson, TX 75080, kelly.jahn@utdallas.edu) and Prabuddha Bhatarai (Speech, Lang., and Hearing, The Univ. of Texas at Dallas, Richardson, TX)

Individuals who use cochlear implants (CIs) rely on a spectrally degraded electric signal to perceive and process acoustic stimuli. In a series of experiments, we evaluated how CI processing affects emotional responses to sound in listeners with normal hearing (NH) and in those with CIs. First, 25 young adults with NH listened to environmental sounds that were processed through a noise band vocoder to simulate CI spectral degradation. They perceived the spectrally degraded (vocoded) sounds as significantly less extreme (*i.e.*, more neutral) and less arousing than the full-spectrum stimuli. A follow-up study investigated sound-induced emotional responses in 18 adult CI listeners and 17 adults with NH using a battery of self-report, behavioral, and autonomic measures. Like the vocoder study, CI listeners exhibited a narrower range of emotional responses to the sounds, perceiving both pleasant and unpleasant sounds as significantly less pleasant and less unpleasant, respectively, compared to the NH listeners. The CI listeners also exhibited significantly slower sound-evoked pupil dilations than the NH listeners, suggesting they were slower to process emotionally evocative stimuli. These findings can inform clinicians in identifying targets for counseling and rehabilitation to improve emotional experiences and quality of life for adults with CIs.

2:15

1pPP5. Cochlear implant users' categorical perception of vocal emotions with various prosodic cues. Kimia Habibnejad Arabi (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ), Monita Chatterjee (Commun. Sci. & Disord., Northwestern Univ., Evanston, IL), and Xin Luo (Speech and Hearing Sci., Arizona State Univ., Arizona State University, Tempe, AZ 85287, xinluo@asu.edu)

In theory, categorical perception involves two features: (1) a sharp transition in the identification of evenly spaced stimuli along a speech continuum, and (2) better discrimination for stimulus pairs that cross the category boundary than for pairs within the same category. This study employed a categorical perception paradigm to investigate how cochlear implant (CI) users use various prosodic cues to identify and discriminate emotional speech. The stimuli were happy–sad continua of the sentence "Time to go" spoken by an adult female talker, with five evenly spaced steps in voice pitch contour, overall intensity, and duration, either individually or in congruent combination. Results of 15 adult CI users tested with a single CI showed that emotion identification functions were significantly steeper with combined cues than with pitch or intensity cues, whereas functions with combined cues were similar to those with duration cues. Emotion discrimination d' values were significantly higher for across-boundary than within-boundary stimulus pairs. Combined cues yielded significantly higher d' values than duration and intensity cues, but not pitch cues. These results suggest that CI users employ different cue-weighting strategies, relying more on duration cues for vocal emotion identification and more on pitch cues for vocal emotion discrimination.

2:30

1pPP6. Acoustic cues utilized by cochlear implant patients for the perception of emotional prosody. Monita Chatterjee (Commun. Sci. & Disord., Northwestern Univ., Frances Searle Bldg., 2240 N. Campus Dr., Evanston, IL 60208, monita.chatterjee@northwestern.edu), Aditya M. Kulkarni (Boys Town National Res. Hospital, Omaha, NE), Parker Hagemann (Boys Town National Res. Hospital, Iowa City, IA), Abby Pitts (Boys Town National Res. Hospital, Omaha, NE), Jessica Combs (Boys Town National Res. Hospital, Tampa, FL), Denis Fitzpatrick, Dawna Lewis (Boys Town National Res. Hospital, Omaha, NE), John J. Galvin (House Inst. Foundation, Los Angeles, CA), and Xin Luo (Arizona State Univ., Tempe, AZ)

In this study, we explored acoustic cue weighting for emotional prosody perception by prelingually deaf school-age children and young adults with cochlear implants (CIs). Stimuli were derived from an original sentence "Time to go" spoken in a happy and a sad way by an adult female talker. The stimulus set consisted of 125 versions of "Time to go" with five levels of voice pitch contour cues, five levels of duration cues, and five levels of overall intensity cues combined orthogonally and spanning the happy-sad range in each dimension. Logistic regression analysis with 28 participants' data showed significant utilization of voice pitch contour and duration cues but not overall intensity cues. Earlier age at implantation and longer duration of device experience increased the utilization of voice pitch contour cues in the cue-weighting study, underscoring the importance of both early auditory input *via* CIs and experience-driven plasticity during development. Individual coefficients for voice pitch contour and duration cues showed considerable variability across participants, and both predicted emotional prosody identification accuracy in a task with 12 sentences recorded in five emotions by a different talker. Ongoing analyses of results, including additional participants, will be presented.

2:45–2:55 Break

Contributed Paper

2:55

1pPP7. Acoustic analyses of prosodic features in speech stimuli with shifted fundamental frequency and vocal tract length. Alexander Goodman (Speech-Lang. Pathol. & Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, agoodma4@students.towson.edu), Alexandra Z. Durfee, and William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD 21252)

Fundamental frequency (F0) and vocal tract length (VTL) determine the pitch and spectral shape of a talker's voice and are used for stream segregation of speech from multiple concurrent talkers. These same acoustic cues also provide information on the emotional content of the talker's message. In this study, a signal processing approach was implemented in Praat to shift F0 and spectral shape of a single male voice to generate perceptually distinct stimuli that would be perceived as a female talker and child talker. Acoustic analyses will be presented to describe the voice and emotional prosody features of these stimuli, which will be used in experimental research that explores the effect of voice features on recognition of emotional prosody and stream segregation of multiple talkers. The signal processing approach provides a controlled method for generating distinct talkers that carry identical prosodic features of rate and relative intonation contours. Results of the acoustic analyses will be discussed in terms of the advantages and disadvantages of this stimulus generation approach, as compared to recording stimuli from separate talkers.

Invited Papers

3:05

1pPP8. Amplitude modulation in non-speech sounds below 430 Hz predicts emotion perception with and without visual context. Erin M. Picou (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, erin.picou@vanderbilt.edu), Haiping Huang (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN), and William L. Martens (Dept. of Linguist., Macquarie Univ., Sydney, New South Wales, Australia)

Research has shown that amplitude modulation within auditory bands below 430 Hz predicts emotion perception of non-speech sounds. However, the extent to which this relationship generalizes to multimodal stimuli remains unclear. This study aimed to evaluate the potential relationship between emotional responses to non-speech sounds and the amplitude modulation energy in the auditory components of the stimuli presented with and without semantically related pictures. Adults with and without hearing loss were presented with auditory-only and auditory-visual stimuli to evaluate the extent to which hearing loss changes the modulation-emotion relationships. Analyses revealed that, although ratings of emotion perception were generally more extreme in response to auditory-visual stimuli than auditory-only stimuli, the relationship between modulation energy and emotion perception was similar across modalities. In addition, the relationship between modulation energy and emotion perception was generally weaker for the adults with hearing loss than for the adults with normal hearing. These findings highlight the importance of amplitude modulation in emotion perception, even in the presence of visual cues. They also suggest a potential mechanistic contributor to the reduced range of emotional responses to non-speech sounds previously observed in adults with hearing loss.

3:15

1pPP9. Acoustic and listener predictors of emotional responses to non-speech sounds. Eric M. Johnson (Div. of Commun. Sci. and Disord., West Virginia Univ., 375 Birch St., Morgantown, WV 26506, eric.johnson5@hsc.wvu.edu) and Erin M. Picou (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN)

Emotional responses to sounds are often described along two dimensions: arousal (exciting–calming) and valence (pleasant–unpleasant). Whereas sound level is a robust cue for arousal, the acoustic cues supporting valence are less well defined, particularly across listeners with different hearing abilities. Valence ratings were obtained from 236 adults (with and without hearing loss) for 75 sounds from the International Affective Digitized Sounds corpus (IADS-2). These data were pooled from several studies. For each sound, acoustic

characteristics reflecting spectral shape and its time-varying changes were extracted. For each listener, age, gender, pure-tone average hearing threshold (0.5–2 kHz), and anxiety/depression symptom scores were collected. Valence ratings for previously unseen listeners were predicted using an interpretable machine-learning model (gradient-boosted trees with a mixed-effects structure). Out-of-fold performance was moderate ($R^2 = 0.458$; Spearman $\rho = 0.679$). The strongest predictors included variability in spectral "peakiness" (spectral crest), mid-frequency spectral contrast near 1–2 kHz (mean and variability), and loudness variability. Hearing loss also contributed, suggesting audibility and/or altered sound representation affects pleasantness judgments. Overall, listeners appear to rely on time-varying spectral structure, particularly in mid frequencies, when forming pleasant–unpleasant judgments, providing candidate acoustic cues for affective sound processing in listeners with and without hearing loss.

Contributed Papers

3:25

1pPP10. Stream segregation and the identification of talker and emotional in school-aged children. William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, wbologna@towson.edu), Alexander Goodman, and Alexandra Z. Durfee (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD)

The abilities to recognize a talker's voice and identify their emotion are critical for social development in children. In many real-world situations, these abilities are performed concurrently as listeners segregate a target voice from other background talkers. Acoustic features of a talker's voice, such as spectral shape, fundamental frequency, and intonation contour, contribute of all three of these perceptual abilities: talker recognition, emotion recognition, and stream segregation. A novel perceptual task has been developed to study these perceptual abilities in isolation and in concert with each other in the same stimulus set. Data on children aged 5–12 with normal hearing will be described and compared to existing data from this task on adult listeners with normal hearing. Results will be discussed in terms of the contributions of voice features to stream segregation, voice identification, emotional recognition, and concurrent performance of these perceptual abilities in children and adults. These data and future work will be used to determine the extent to which these perceptual abilities develop independently or scaffold each other's development throughout childhood and adolescence. Implications for how these developmental trajectories may be affected in special populations, such as cochlear implant recipients and autistic children, will also be discussed.

3:35–3:45 Break

3:45

1pPP11. How street soundscapes shape emotional experience and well-being: Insights from expert focus groups. zeynep s. ozturk (Univ. College London, Central House, 14 Upper Woburn Pl., London WC1H 0NN, United Kingdom, zeynep.ozturk.22@ucl.ac.uk), Jian Kang, and Francesco Aletta (Univ. College London, London, United Kingdom)

High street atmospheres play an important role in the daily lives of communities and can evoke a wide range of emotional and sensory responses. Previous soundscape research has largely focused on physical and environmental characteristics, while the emotional dimensions of street soundscapes have received comparatively little attention. This study aims to explore how people perceive street soundscapes through the lens of emotional responses. The research uses online focus groups with soundscape experts, policymakers and urban planners. A deductive thematic analysis showed that street soundscapes are shaped not only by the type of sounds present, but also by the meanings people assign to them and the emotions they evoke. The findings suggest that street sound environments influence emotions in several ways: they can affect mood and well-being, shape perceptions of safety, contribute to feelings of belonging and place attachment, support restorative–positive experiences, and also trigger negative emotional responses. Future studies should examine the relationship between street soundscapes and emotional responses from a broader perspective. They should also consider how soundscape characteristics interact to shape emotional experience.

3:55

1pPP12. Atypical brain activation and functional connectivity during affective sounds processing in misophonia: An fMRI study. Namitha Jain (Dept. of Speech and Hearing Sci., Univ. of Illinois–Urbana Champaign, 901 S Sixth, Champaign, IL 61820, USA, namitha3@illinois.edu), Shagun Ajmera (Beckman Inst. for Adv. Sci. & Technol., Univ. of Illinois Urbana Champaign, Urbana, IL), Somayeh Shahsavarani (Dept. of Audiol., San Jose State Univ., San Jose, CA), Rafay A. Khan (Beckman Inst. for Adv. Sci. & Technol., Univ. of Illinois Urbana Champaign, Urbana, IL), Gibbeum Kim (Speech and Hearing Sci., Univ. of Illinois at Urbana–Champaign, Champaign, IL), Howard Berenbaum (Dept. of Psych., Univ. of Illinois Urbana Champaign, Champaign, IL), and Fatima T. Husain (Speech and Hearing Sci., Univ. of Illinois Urbana–Champaign, Champaign, IL)

Sound is a powerful driver of emotion, with the brain rapidly processing acoustic features and attributing emotional significance to everyday sounds. In misophonia, this sound–emotion interaction is exaggerated, provoking intense negative emotional and behavioral reactions to otherwise benign "trigger" sounds like chewing or breathing. While misophonia is typically linked to these specific triggers, growing evidence suggests that misophonia-related affective responses may extend beyond classical triggers. To explore this, we investigated brain responses to generally unpleasant and neutral sounds using task-based fMRI. Ninety-one young adults with misophonia and matched controls listened to 90 emotionally valenced sounds and provided valence ratings during scanning. Compared to controls, individuals with misophonia exhibited hyperactivation in visual association areas and reduced functional connectivity between salience and visual networks when processing unpleasant relative to neutral sounds. These findings indicate atypical cross-modal sensory engagement and suggest that altered brain activation and connectivity in misophonia extend beyond trigger sounds to a broader range of emotionally relevant auditory stimuli. Building on these fMRI results, our ongoing behavioral study explores whether similar emotional response patterns generalize across cultures. Together, these studies aim to advance understanding of misophonia's neural and behavioral mechanisms and inform culturally sensitive clinical assessment and intervention.

4:05

1pPP13. Auditory emotional attention in anxiety disorders. Deanna V. Sharpe (Emory Univ., 201 Dowman Dr., Atlanta, GA 30322, deannas0730@gmail.com), Lauren O'Rourke, and Mishaela DiNino (Univ. at Buffalo, Buffalo, NY)

Individuals with anxiety disorders exhibit attentional bias to negative visual stimuli. They may also demonstrate a bias toward emotional sounds, which could disrupt the auditory attentional processes vital for hearing in background noise. This study examined whether anxiety disorder symptoms predicted young adults' ability to attend to and ignore emotionally prosodic sentences. Participants with normal hearing completed a task with two conditions: one in which the target sentence contained emotional prosody, but the distractor was neutral, and one in which the target sentence was neutral, and the distractor contained emotional prosody. Participants also completed questionnaires of generalized and social anxiety symptoms. Results indicated that individuals with higher levels of generalized anxiety, but not

social anxiety, performed significantly better on both conditions in the emotional attention task than did individuals with low symptom levels. These results suggest that individuals with generalized anxiety disorder, which is characterized by clinically significant worry that negative events will occur, experience greater auditory salience to emotionally prosodic stimuli. This means they are better able to both attend to and ignore emotional stimuli than are individuals without this disorder. Further research on auditory emotional attention in anxiety is critical to understanding attentional bias and ameliorating anxiety symptomatology.

4:15

1pPP14. Perceiving trusting and doubting prosody with typical hearing and with cochlear implants. Abbey L. Thomas (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., Minneapolis, MN 55455, abbeyt@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN, abbeyt@umn.edu)

Prosody, including speaking rate and intonation, is key for communicating trust and doubt. These attitudes are crucial in healthcare settings, as a provider's trust in the patient encourages continuity of care and effective communication, while a provider's doubt diminishes patient satisfaction and willingness to follow recommendations. The present study extends previous research on young typical hearing listeners, comparing perception of doubting and trusting prosody by young typical hearing listeners to older typical hearing listeners and listeners across the adult age span with cochlear implants (CIs). Using a continuous slider scale with endpoints of "doubt" and "trust," listeners rated attitude in tokens of the word "okay," in which F0 contour, vowel duration, and closure duration and intensity of /k/ were manipulated independently. Across groups, listeners rated longer duration as more doubting. Young typical hearing listeners consistently identified high-falling F0 as trusting and low-rising F0 as doubting. CI listeners followed the same pattern, but only for tokens with longer durations. Intonation had smaller effects on older listeners' responses, with some giving responses opposite the expected patterns. These results suggest caution in relying solely on intonation to convey trust and in speaking slowly for intelligibility, as slowed speech is consistently interpreted as doubting.

MONDAY AFTERNOON, 11 MAY 2026 SALON B, 1:15 P.M. TO 2:45 P.M.

Session 1pSA

Structural Acoustics and Vibration: Laser-Based Measurement and Wave Control

Brian E. Anderson, Cochair
Phys. & Astronomy, Brigham Young Univ., Dept. of Phys. & Astron., N245 ESC, Provo, UT 84602

Colby W. Cushing, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

1:15

1pSA1. Laser Doppler vibrometer measurements through multi-layered media. Jeremy Keirn (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, Steam Dr., State College, PA 16804, jdk5633@psu.edu) and Megan B. Ewers (Appl. Res. Lab., The Penn State Univ., State College, PA)

Laser Doppler vibrometers (LDVs) can be used for a wide range of acoustic vibration measurements in a variety of applications. LDVs use the phase difference between transmitted and reflected beams of light in an interferometer to determine an object's velocity in the direction of the incident beam using the Doppler effect. The ideal test configuration is an uninterrupted beam through a homogeneous medium towards a highly retroreflective object; however, such conditions are often infeasible in a practical setting. A physical measurement can be impacted by environmental test conditions like multiple inhomogeneous media with different indices of refraction and reflectivities, which makes retroreflection from the target specimen challenging. For example, the target beam might need to travel through an air-to-water interface, which can be difficult due to reflections from the interface and mismatched indices of refraction. A tabletop test setup, including a small acrylic tank, water, and a shaker, was used to determine parameters to achieve favorable test conditions to measure the in-plane velocity of the acoustic vibrations of a test article. Different retroreflective surface finishes, standoff distances, incident angles, and shaker-induced acoustic vibration frequencies are explored to provide recommendations for experimentalists attempting similar measurements.

1:30

1pSA2. Time reversal focusing of vibrations in a metal plate: Determining the best method for spatially-extended, multi-point focusing. Kadyn T. Wright (Phys. & Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, ktw33@student.byu.edu) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) is a technique that can be used to focus vibrations to a desired location on a plate. An impulse response is obtained between a shaker and a scanning laser Doppler vibrometer receiver which records the timing of various reflections as they arrive. The TR technique involves broadcasting the time reversed impulse response from the shaker, which results in broadcasting of the various reflections in reversed timing order so that a portion of the energy from each of these emissions retraces their original paths and arrives at the laser location at the same time. The experiment is conducted on a finite-dimension, thin aluminum plate for proof of concept. In this research, TR is used to focus signals at multiple locations along a line. A comparison of different methods used to create these foci is done using time delays between each focus event (all in phase, applying the same time delay between adjacent foci, and random starting phases). The goal is

to determine which method leads to the highest time-averaged amplitude in a target area compared to everywhere else in the plate. Focusing of impulses, white noise, and other signals will be explored.

1:45

1pSA3. Time reversal focusing of specific signals used for vibration excitation of structures. Spencer T. Neu (Phys. & Astronomy, Brigham Young Univ., N284 ESC, Provo, UT, stneu73@student.byu.edu), Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Washington DeLima (Adv. Manufacturing Eng., Kansas City National Security Campus, Kansas City, MO)

Time reversal focusing uses an impulse response to create a focus of energy at one or more locations on a structure by exploiting the reciprocal nature of the source to receiver path propagation. The focused signal can be an impulsive spike or another signal as desired. The ability to create a customized signal at a target location is applicable to structural vibration testing. Testing structures usually requires exciting the whole structure. Because time reversal can excite specific areas of a structure, it may allow for a more efficient delivery of high-amplitude vibration excitation to target locations. The goal of this work is to characterize the time reversal technique's ability to accurately reproduce a desired excitation signal at a target location, starting with a rectangular plate for proof of concept. The use of deconvolution with time reversal will also be analyzed.

2:00

1pSA4. Comparison of surface fitting basis functions for calculating structural power flow on curved shells from noisy data. Tysum Ruchti (Mech. Eng., Brigham Young Univ., 155 East 1230 North, Provo, UT 84602, tysumruchti@byu.edu) and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Structural power flow provides a critical metric for identifying energy sources, sinks, and transmission paths within vibrating structures. However, computing spatially continuous power from experimental data remains challenging due to the requirement of high-order spatial derivatives to determine force and moment resultants. These numerical derivatives amplify measurement noise, making the calculation an ill-posed problem. This study evaluates the efficacy of using various basis functions to smooth noisy velocity data prior to differentiation. We perform a comparative analysis of six functions: Polynomial, Chebyshev, Fourier, quintic B-spline, Mexican Hat wavelet, and Morlet wavelet. To test these methods, we utilize a finite element model of a curved shell, as a reference model. Noise is added to the nodal displacements based on typical speckle noise and spatial uncertainty from a scanning laser Doppler vibrometer. Results indicate that the Mexican Hat wavelet is the most robust method, exhibiting the lowest sensitivity to basis function configuration. Furthermore, it demonstrates superior accuracy in computing power near free and simply supported boundary conditions, where other methods often fail.

2:15

1pSA5. Bending wave characterization of viscoelastic panels using a laser Doppler vibrometer. Mengyi Yang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, yangm11@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Viscoelastic materials are widely used in noise and vibration controls. They exhibit strong damping for bending waves and resulting in significant transmission loss. The dispersive nature of viscoelastic materials intrigues the efforts of experimental characterization of it, which will benefit the prediction of bending wave propagation in the viscoelastic panels. Utilizing the laser doppler vibrometer (LDV), a non-contacting measurement method has been developed to determine the bending wave properties of the viscoelastic panels. Mechanical characterization including phase speed, loss factor, and Young's modulus can be calculated from experimentally measured transfer functions of bending velocities. This work discusses the recent experiments using the LDV-based transfer function method and associated challenges when improving robustness and accuracy over broad frequency ranges.

2:30

1pSA6. Non-destructive vibroacoustic evaluation of material properties in ceramic components. Jadwiga E. Hyla (Mech. and Vibroacoustics, AGH Univ. of Krakow, Mickiewicza av. 30, AGH Bldg. D1, Kraków 30-059, Poland, hylajadzia@gmail.com) and Michał Kozupa (Hitachi Energy Res., Kraków, Poland)

Accurate material properties are essential for reliable finite element method (FEM) dynamic analysis. Those taken from literature or standards can sometimes be unsatisfactory. Conventional methods for determining material properties of ceramic components often rely on destructive mechanical tests, such as tensile testing, that permanently damage the sample and compromise its integrity. As a non-destructive alternative, experimental modal analysis (EMA) enables the estimation of dynamic characteristics. In this study, EMA was employed to identify resonance frequencies and associated modal parameters of ceramic components using a laser Doppler vibrometer. The Modal Assurance Criterion (MAC) was applied to assess the consistency of experimental mode shapes and their correlation with the finite element model predictions. The resulting modal data were incorporated into an iterative numerical optimization framework, facilitating accurate estimation of material properties. This methodology provides a non-destructive and highly precise solution for material characterization, making it particularly advantageous for engineering and industrial applications.

Session 1pSC

Speech Communication: Production & Articulation (Poster Sesson)

Margaret E. Renwick, Chair
Linguist., Univ. of Georgia, Dept. of Cognitive Sci., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218

All posters will be on display from 1:00 p.m. to 4:00 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbers papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

1pSC1. Voice onset time in medial syllable stop consonants produced by typically developing children. Michelle Swartz (Commun. Sci. and Disord., Stockton Univ., 101 Vera King Farris Dr., Galloway, NJ 08205, michelle.swartz@stockton.edu), Elaine R. Hitchcock (Commun. Sci. & Disord., Montclair State Univ., Bloomfield, NJ 08205), and Laura Koenig (Haskins Labs, New Haven, NY 06511)

Voice onset time (VOT), an acoustic measure indexing the temporal coordination between stop release and voicing onset, is widely used as an indicator of speech motor control. Developmental research shows that children produce longer and more variable VOTs than adults, with systematic effects of age and phonetic context. Multisyllabic contexts, in particular, tend to yield shorter but more variable VOTs than monosyllables. However, VOT production in non-initial, less prosodically prominent syllable positions remains underexplored. This study examines VOT in second-syllable stop consonants embedded in nonwords varying in length and stress context (*e.g.*, /bidə/, /pəki/, /bidəgo/) produced by ten typically developing children aged 9–15 years. Nonwords were used to minimize lexical influences, while medial syllable contexts were selected to increase demands on intergestural timing and coordination. Second-syllable voiced and voiceless stop consonants were followed by either tense vowels (*e.g.*, /i, u, a/) or unstressed schwa, allowing examination of potential vowel-related effects on preceding stop consonant VOT. Results will extend current developmental accounts of stop consonant timing by characterizing VOT in medial syllable positions and will provide preliminary reference data for future comparisons with children who present with speech sound disorders.

1pSC2. An articulatory and acoustic investigation of Lutuv voiceless liquids. Grayson Ziegler (Linguist., Indiana Univ., 504 Ballantine Hall, 1020 E. Kirkwood Ave., Bloomington, IN 47405, grzieg@iu.edu)

Voiceless liquids are typologically uncommon phones and are underdocumented both articulatorily and acoustically. Numerous questions persist about voiceless liquids, including whether voiceless liquids are articulated with a similar lingual setting as their voiced counterparts, and whether impressionistic comparisons to voiceless fricatives, especially sibilant fricatives in the case of voiceless rhotics, are well-founded. The present work provides a preliminary analysis of voiceless liquid consonants in Lutuv (also Lautu), an under-resourced South-Central Tibeto-Burman (or Chin) language spoken by 18 000+ people in Chin State of western Burma and in diaspora communities worldwide (Eberhard *et al.* 2025). Lutuv contrasts voiced and voiceless laterals and rhotics /r l ɾ̥ l̥/, with /r ɾ̥ l̥/ attested as fricative-like productions (Berkson *et al.* 2022). This investigation observes Lutuv liquid production from one speaker using 3D ultrasonography for lingual articulatory comparison and spectral moments and voice quality (e.g., F0, spectral tilt, and harmonics-to-noise ratio) measures using VoiceSauce (Shue *et al.* 2011) for acoustic comparison. Additionally, articulatory and acoustic comparisons are drawn between voiceless liquids and voiceless fricatives /s s^h∼ʃ/. Results suggest that voiceless liquids are articulatorily like their voiced counterparts while occupying an acoustic space between voiceless fricatives and other voiceless sonorants, such as voiceless nasals.

1pSC3. Rhotics in Matu Chin. Amanda Bohnert (Linguist., Indiana Univ., 1020 E. Kirkwood Ave., Ballantine Hall 504, Bloomington, IN 47405, abohnert@iu.edu), Kelly H. Berkson, Grayson Ziegler (Linguist., Indiana Univ., Bloomington, IN), and Kenneth Van Bik (English, Comparative Lit., and Linguist., California State Univ., Fullerton, CA)

This work presents an acoustic investigation of rhotics in Matu, an under-resourced and under-documented South Central Tibeto-Burman (or Chin) language spoken by approximately 40 000 people in Chin State in western Burma/Myanmar and Mizoram in eastern India (Eberhard *et al.*, 2025). Existing works on Matu present very little rhotic data (often only a few words), and make contrasting claims regarding Matu rhotics: two scholars identify two rhotics common across Chin languages, /r, ɾ̥/, with no explanatory commentary or acoustic information (Shintani 2016; Khoi Lam 2001); two others identify a single rhotic /x/ (Hoffman 2018; So-Hartmann 1988), with Hoffman (2018) reporting inter- and intra-speaker free variation between [x, ɻ̊, z̠, r]. Uvular rhotics have never been reported for Matu, but were immediately evident to us in fieldwork with a Matu speaker in Maryland. Moreover, Matu has a well-established and relatively transparent orthography which includes symbols for four distinct rhotics: <r, hr, rh, rhr>. We present an investigation of the Matu rhotic system to explicate why our data diverge from the literature, counteract the paucity of extant acoustic documentation and rhotic data, and probe the rhotic diversity suggested by the orthography. We identify three clear rhotic phones with a possible marginal fourth phone.

1pSC4. Contrastive hyperarticulation in Breton and French vowels. Brandon Gauthier (Linguist., Univ. of Arizona, 10583 E Geyer Willow Rd., Tucson, AZ 85747, bgauthier@arizona.edu)

This study investigates contrastive hyperarticulation in the vowel systems of Breton and French, examining whether speakers enhance vowel distinctiveness in the presence of lexical competitors. While contrastive hyperarticulation has been well documented in majority languages, its status in minoritized languages under sustained contact remains understudied. Using crowdsourced speech from the Common Voice corpus, productions of /i/, /e/, and /ɛ/ were analyzed for 119 Breton and 506 French speakers. Vowel articulation was assessed using two complementary acoustic measures: (1) Euclidean distance from a speaker's average production of the same vowel, and (2) distance from the average production of a competing vowel. Results from the first measure show clear evidence of contrastive hyperarticulation in both languages: vowels were produced with greater deviation from baseline in the presence of within-language minimal pairs, with phoneme-specific effects. Breton vowels also showed sensitivity to French minimal pairs, suggesting bilingualism effects. However, the second measure yielded inconsistent and often opposing patterns, including

decreased contrast. These findings suggest that evidence for contrastive hyperarticulation depends critically on how distinctiveness is operationalized. The results underscore both the presence of listener-oriented articulatory enhancement and the methodological challenges of detecting it in large, automatically processed speech corpora.

1pSC5. Glottalization and hyper-articulation of /t/ in the Wasatch Front variety of English. Richard A. Wright (Linguist., Univ. of Washington, Dept. of Linguist., Box 352425, Seattle, WA 98195-2425, rawright@uw.edu) and Mykel L. Brinkerhoff (Linguist., Univ. of Washington, Seattle, WA)

Glottalization of /t/ before syllabic -n, as in words like "button" and "mountain," is common many dialects of North American English. In the English spoken along the Wasatch Front in Utah it is often realized as glottalization with a reduced vowel prior to a non-syllabic-n often transcribed as [ʔɨn] or [ʔən]. While this feature is widespread, it is also highly stigmatized resulting in frequent hyper-articulation to what is referred to as a "hard-t" which is typically described as aspirated. Although the glottalized and hypercorrected variants are widely described in the sociolinguistic literature there has been very little acoustic study of them. In this study, we examine the acoustic variation of t-glottalization and the hyperarticulated variants. We recorded 20 speakers (10 men) of the Wasatch Front variety of English reading sentences designed to elicit reduced (glottalized) and hyper-articulated forms. The sample includes a variety of ages between 18 and 80 from the Salt Lake metropolitan area. Acoustic measures of glottalization include spectral tilt, harmonics to noise ratio, strength of excitation, and qualitative spectrographic analysis. Measures of aspiration include VOT and burst duration.

1pSC6. Speech-based sleep deprivation detection based on changes in articulatory coordination. Chigozie Udeogu (Dept. of Elec. and Comput. Eng., Univ. of Maryland, College Park, MD 20742, cudeogu@umd.edu) and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, College Park, MD)

Sleep deprivation poses a growing burden on individuals and society, as insufficient sleep impairs daily functioning, elevates accident risk, and may result in severe health outcomes, including mortality, highlighting the importance of early and reliable detection. Sleep-deprived individuals exhibit measurable changes like reduced articulatory precision in speech production thus making speech a rich and informative attribute for detecting sleep deprivation. Articulatory coordination features (ACFs) derived from vocal tract variables (TVs), predicted using a speech inversion system, provide a principled means of quantifying changes in the coordination of articulatory gestures. Such features have previously demonstrated strong effectiveness in the classification of mental health conditions, including depression and schizophrenia. In this study, we plan to leverage data from the DCIEM/HCRC and Sleepy Language Corpus (SLC) datasets the ACFs can be used to capture changes associated with sleep deprivation and support its detection. We analyze articulatory coordination patterns using eigenspectra derived from channel delay correlation matrices computed from speech when someone is sleep deprived and when they are not. A detailed interpretation of the differences in eigenspectra will be discussed.

1pSC7. Exploring regularities in respiratory and supralaryngeal aspects of speech breathing. Laura Koenig (Commun. Sci. and Disord., Adelphi Univ., 300 George St., New Haven, NY 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Leibniz Ctr. for General Linguist, Berlin, Germany)

Speech inhalation may occur orally, nasally, or through a combination of both, yet remarkably little is known about the consistency of these patterns across speakers or over time. Inhalation modes may be speaker-specific, situation-dependent, or variable. Understanding this behavior is important for both speaker identification and clinical assessment. Previous work relied largely on pneumotachometer masks and constrained speech materials (*e.g.*, sustained vowels), which may limit ecological validity by restricting jaw motion, altering the acoustic signal, or failing to reflect natural speech. We recorded six speakers reading a multi-paragraph German passage on four occasions across several weeks. Alongside acoustics, we measured thoracic and abdominal movements using RespTrack and supraglottal movements using optical stomatography to capture oral aperture. Inspiratory noise was identified acoustically and auditorily classified as oral, nasal, or mixed. Using lip sensor data, we test whether inspirations labeled as oral, nasal or nasal–oral do in fact involve mouth opening, and if so, when opening occurs. This allows us to evaluate the accuracy of auditory classification. We further assess whether oral inspirations, confirmed articulatorily, exhibit steeper respiratory kinematic slopes than nasal inhalations. Using these analyses, we will discuss the consistency and speaker specificity of inhalation strategies in read speech.

1pSC8. Assessing vowel hyperarticulation in Korean infant-directed speech: A token-level approach. Eon-Suk Ko (Dept. of English Lang. and Lit., Chosun Univ., Gwangju 61452, Korea, eonsuk@gmail.com) and Sunghye Cho (Univ. of Pennsylvania, Philadelphia, PA)

Research on vowel hyperarticulation in infant-directed speech (IDS) has long been marked by empirical inconsistency, with prior studies reporting both expanded and non-expanded vowel spaces relative to adult-directed speech (ADS). Such divergence has been attributed to methodological variability and analytic choices. The present study examines Korean IDS using the vowel hypo- and hyper-articulation (VHH) index, a token-level metric designed to mitigate limitations inherent in aggregate vowel space measures. Data were drawn from a word-teaching task involving 22 Korean mother–infant dyads, including prelexical and early-lexical infants. Mothers produced nonce words containing /a/, /i/, and /u/ in IDS and ADS contexts across reading and spontaneous speech. A total of 2883 vowel tokens were analyzed, with speaker- and vowel-specific optimization of formant ceilings. Hyperarticulation was assessed using both traditional formant-based comparisons and linear mixed-effects models incorporating the VHH index. Results indicate that Korean mothers systematically hyperarticulated vowels in IDS when evaluated via the VHH index, with this effect largely mediated by vowel duration. This pattern suggests that hyperarticulation emerges as a consequence of temporal expansion rather than independent articulatory enhancement. These findings highlight the importance of methodological refinement in capturing dimensions of IDS accommodation that may be obscured under conventional analytic approaches.

1pSC9. Temporal and spectral cues for voicing in children's stop consonants. 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, CT 06511)

This study examined stop consonants produced by typically developing children to examine patterns of voice onset time (VOT) and secondary voicing cues. Thirteen monolingual English-speaking children (40–98 months) with typical speech, language, and hearing were recorded over 18 sessions across 9–11 months. Each session included 10 repetitions of picture-elicited monosyllabic minimal pairs (*e.g.*, beach–peach, dock–tock, gay–kay) spanning bilabial, alveolar, and velar stops in two vowel contexts, yielding 28 860 usable productions. VOT was measured manually from waveform and spectrogram displays; the current presentation will add data on secondary cues: f0 and F1 at voicing onset, burst intensity, and vowel duration. For VOT, children demonstrated high accuracy; ranges varied widely depending on child and consonant, being heavily influenced by the use of prevoicing. Voicing categories were generally discrete (separated in VOT values) based on averaged data, but could show mild–moderate overlap when all tokens were included. Overshoot (exaggerated VOT lags) was also observed, more frequently using a token-by-token analysis than using means. Evaluation of secondary cues may lend insight into children's voicing contrasts in cases where VOT values are ambiguous or inappropriate to the target category.

1pSC10. Laryngeal devoicing gestures observed using transverse-oriented ultrasound. Mark Tiede (Dept. of Psychiatry, Yale Univ., New Haven, CT 06511, mark.tiede@yale.edu), Fengyue Lisa Zhao (Dept. of Linguist., Cornell Univ., Ithaca, NY), Sam Tilsen (Cornell Univ., Ithaca, NY), Laura Koenig (Adelphi Univ., New York, NY), and D. H. Whalen (CUNY Graduate Ctr., New York, NY)

Ultrasound imaging of midsagittal tongue posture is well established due to its benign qualities and ease of use, but its application to laryngeal imaging has been limited. Exploratory work reported here demonstrates that transverse-oriented ultrasound at the level of the laryngeal protuberance successfully tracks devoicing gestures of the ventricular (false) folds, known to be reliably correlated with abductory movements of the true folds. Native speakers of American English produced VCV sequences while maintaining consistent fundamental frequency to minimize changes in laryngeal height, recorded with synchronized speech audio. C included [h], [ʔ], [p], and [b], with context vowels [a] and [i]. For quantification, a sector mask encompassing the abducted folds was used to obtain the log-sum of pixel intensities within the mask as a proxy for the extent of abduction for a given ultrasound video frame, with lower values (showing less included reflective tissue) indicating greater abduction. As expected [h] shows the greatest extent of abduction, but systematic differences in the timing and degree of ab/adduction are seen across all C types. Measurements of abductory gestural timing obtained through this approach have the potential to improve our understanding of consonant devoicing and glottal stop realization.

1pSC11. Phonological awareness and coarticulation revisited. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN 55455, munso005@umn.edu) and Eugene Wong (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Recent research showed that increases in children's phonological awareness is associated with decreases in coarticulation (Noiray *et al*., 2019). This relationship might occur because both skills index broadly the degree of segmental detail in phonological representations. Noiray *et al*. assessed the relationship using articulatory measures of stop production in a cohort of 4- to 7-year-old children speaking German. The current study extends this work by examining whether phonological awareness predicts the degree of stop-vowel and fricative-vowel coarticulation a large ($n = 147$) cohort of typically developing 5-year-old children who had participated in a longitudinal study of phonological development and vocabulary growth (Erskine *et al*., 2020; Wong *et al*., 2025). The analysis focuses on whether the degree of stop-vowel and fricative-vowel coarticulation, as measured acoustically, predict scores on a widely-used standardized measure of phonological awareness, while controlling for other factors that might affect coarticulation (*e.g*., vocabulary size, speech perception acuity). Additionally, this work allows us to examine the consistency of consonant-vowel coarticulation across four sequence types: /s/ + vowel, /ʃ/ + vowel, /t/ + vowel, and /k/ + vowel, and whether all four types of coarticulation are equally strongly related to phonological awareness. [Work supported by NIH Grant R01DC002932 to J. Edwards, M. Beckman, and B. Munson].

1pSC12. Examining the effect of speaking tasks on speech rate in dysarthria secondary to ALS. Leila Ghayedlou (Commun. Sci. and Disord., The Penn State Univ., University Park, 308E Ford Bldg., University Park, PA 16802, lqg5490@psu.edu), Navin Viswanathan, Anne J. Olmstead, Jimin Lee, and Janice Chen (Commun. Sci. and Disord., The Penn State Univ., University Park, PA)

Dysarthria caused by amyotrophic lateral sclerosis (ALS) involves disrupted speech timing and decreased speech rate (Turner and Weismer, 1993). One common intervention to improve intelligibility is clear speech, which leads to systematic slowing of speech (Tjaden *et al*., 2014). Although temporal changes in clear speech are well established, it remains unknown whether different speaking tasks elicit similar temporal adjustments in people with ALS (PALS) compared with age-matched typical speakers. Here, we examine whether PALS demonstrate uniform global slowing or task-dependent temporal adjustments across communicative contexts. We studied 40 PALS and age-matched speakers who produced phrases across four speaking conditions: habitual speech and clear speech (non-interactive tasks), matching and clear-speech matching (interactive tasks with an interlocutor). Our results showed task effects on speech rate, with clear speech being associated with longer durations and slower rates compared to habitual speech. PALS exhibited overall slower speech timing than age-matched typical speakers; however, task-related temporal adjustments were similar across the groups, with no additional temporal changes associated with the presence of an interlocutor in the interactive tasks (matching and clear-speech matching). Together, these results indicate that speech timing reflects task-dependent temporal adjustments rather than global slowing.

1pSC13. The impacts of asymmetric acoustic conditions on speech production and communication dynamics during interactive conversation. Benjamin Masters (Systems Design Eng., Univ. of Waterloo, 200 University Ave. W, Waterloo, ON N2L 3G1, Canada, bpmasters@uwaterloo.ca), Polina Stepanenko, and Ewen MacDonald (Systems Design Eng., Univ. of Waterloo, Waterloo, ON, Canada)

The Lombard effect describes the adaptation of speech production based on the acoustic environment (*e.g*., increased speech level in noise). Although it's been suggested that the manifestation of this effect during conversation may be partially driven by the monitoring of a partner's voice, we expect that the effect remains primarily driven by the monitoring of one's own voice in the environment. This study investigates this by examining whether receiving an attenuated version of a partner's voice during a conversation elicits adjustments in speech or communication behavior. Pairs of university-aged, typical hearing talkers held conversations in adjacent sound booths in either a quiet reference condition, or in the presence of a 70 dBA speech-shaped background noise. During the noise condition, one of three gain settings was applied to the speech signals: unity gain or a 3 dB attenuation of either talker's voice. Talkers always heard their own voice at unity gain. We discuss results which evaluate differences in speech acoustics, speech content, and communication behavior across gain settings to determine if the received level of a partner's voice affects one's own speech production.

1pSC14. Effect of voicing of bilabial plosives on surrounding vowels: Acoustic and articulatory analyses of native Japanese speech. Tomomi Furukawa (Graduate School of Sci. and Technol., Sophia Univ., 7-1 Kioicho, Chiyoda-ku, Tokyo 102-8554, Japan, t-furukawa-7f7@eagle.sophia.ac.jp), Ai Mizoguchi (Maebashi Inst. of Technol., Gunma, Japan), Weiyu Li (Graduate School of Sci. and Technol., Sophia Univ., Tokyo, Japan), Maho Morimoto (Chuo Univ., Tokyo, Japan), and Takayuki Arai (Information and Commun. Sci., Sophia Univ., Chiyoda-ku, Tokyo, Japan)

Vowels preceding a voiced consonant have a lower first formant (F1) frequency than vowels preceding a voiceless consonant in English. In addition, a higher jaw position has been observed when producing vowels before voiced consonants than before voiceless consonants. In this study, we compared vowels surrounded by the voiced bilabial plosive /b/ and voiceless bilabial plosive /p/ in native Japanese speakers to examine whether the F1 differences and jaw positioning reflect language-specific patterns or are language-universal, physiologically driven phenomena. Native Japanese speakers participated in a production experiment, where we simultaneously recorded audio, midsagittal ultrasound images of the tongue, and frontal images of the lips during utterances. As a result, a liner mixed model analysis revealed that F1 showed the same tendency as that observed in native English speakers. There was also no difference between the vowels (/a/ and /o/). As for the result of the articulatory analysis, a generalized additive model analysis of the ultrasound images of the tongue revealed no differences in tongue height between the vowels preceding voiced and voiceless consonants. We discuss the relationship among the F1, tongue height, and jaw position. [Work supported by Sophia University Special Grant for Academic Research (Research in Priority Areas).]

1pSC15. Speech production variability and distinctness in children learning to read. Sandy Abu El Adas (Linguist., Stony Brook Univ., 100 Nicolls Rd., Stony Brook, NY 11794-4376, sandy.abuadas@gmail.com) and Marie Lallier (Basque Ctr. on Cognition, Brain and Lang., San-Sebastian (Donostia), Gipuzkoa, Spain)

Poor reading ability is widely understood to stem from broader deficits in phonological processing. Speech perception studies show that individuals with poor reading ability demonstrate reduced identification and discrimination of speech sounds. However, it remains unclear whether these deficits extend to speech production, despite evidence for strong links between perception and production. This longitudinal study examined the relationship between speech production variability, phonetic distinctness, and reading ability at two time points: before and after formal reading instruction. Twenty-nine Spanish-speaking children aged six to seven years completed a battery of reading assessments and a picture-naming task repeated five times. Results showed that children with lower reading scores exhibited a smaller vowel space and shorter voice onset times than children with higher reading scores at the first time point, prior to reading instruction. These differences were no longer observed after formal reading instruction. No significant relationships were found between reading ability and either production variability or phoneme overlap at either time point. These findings suggest that speech production measures are related to reading ability primarily before reading instruction begins, highlighting their potential utility for early clinical identification of reading difficulties.

1pSC16. The effect of rehearsal on filled pause production in short presentation speech. Oliver Niebuhr (Süddänische Universität, Odense, Denmark) and Benjamin V. Tucker (Commun. Sci. and Disord., Northern Arizona Univ., 208 E. Pine Knoll Dr. P.O. Box 15045, Flagstaff, AZ 86011, benjamin.tucker@nau.edu)

Spontaneous conversations and presentations often contain filled pauses like um and uh, yet their functional role and acoustic realization remain poorly understood. We investigated the effect of rehearsal on filled-pause production. Participants delivered a 5-minute slide-based presentation comparing coffee and tea consumption and repeated the task three times in succession. Speech recordings were automatically transcribed and force-aligned for acoustic analysis. Results show that rehearsal did not lead to a uniform reduction of filled pauses. Instead, hesitation behavior was redistributed across grammatical embedding sites: mid-phrase hesitations decreased substantially, while fillers increasingly clustered at phrase and predicate onsets, indicating a shift from local execution difficulties to higher-level structural planning. This redistribution was accompanied by a differential reduction of filler types, with uh occurring less frequently overall and disappearing more rapidly across rehearsal rounds than um. Acoustically, filled pauses became significantly shorter, lower in intensity, and exhibited higher spectral tilt. Their mean F0 remained stable across rounds, but F0 variability and range decreased significantly. At the level of the full presentations, rehearsal led to fewer and shorter silent pauses and an increased speech rate. Thus, rehearsal does not simply eliminate filled pauses but reshapes their distribution and acoustic realization, resulting in increased fluency.

1pSC17. Speech sequencing indicators of depression severity and psychomotor dysfunction in major depressive disorder. Erika L. Exton (Linguist., Northwestern Univ., 0100 LeFrak Hall, College Park, MD 20742, eexton@umd.edu), Lauren Grzelak (Psychiatry and Behavioral Sci., Northwestern Univ., Evanston, IL), David Klemballa (Psychiatry and Behavioral Sci., Northwestern Univ., Chicago, IL), Joseph Keshet (Andrew and Erna Viterbi Faculty of Elec. and Comput. Eng., Technion—Israel Inst. of Technol., Haifa, Israel), Vijay Mittal (Psych., Northwestern Univ., Evanston, IL), Stewart Shankman (Psychiatry and Behavioral Sci., Northwestern Univ., Chicago, IL), and Matthew Goldrick (Linguist., Northwestern Univ., Evanston, IL)

Psychomotor dysfunction, which can manifest as slowing or agitation or both, often occurs in individuals with major depressive disorder (MDD). The goal of this work is to investigate how psychomotor dysfunction manifests in speech production, particularly in tasks that involve rapid repetition of speech motor sequences. In previous research, we found that speed and rate consistency in a sequential diadochokinetic speech task ("pataka") correlated with manual motor performance among individuals with current MDD, suggesting that psychomotor dysfunction can manifest in motorically-demanding speech tasks. In the current study, we extend this work with oral articulators to examining speech motor sequences involving laryngeal control. We find that in a laryngeal DDK task (repeated production of "uh"), individuals with more severe depression symptoms used a faster speech rate and produced the syllables with a shallower slope of intensity change from trough (laryngeal closure) to peak (laryngeal opening). When controlling for speech rate differences, individuals with current MDD also have a shallower intensity slope than healthy controls in the oral DDK task. This pattern may indicate difficulty with speech motor control, resulting in weaker vocal tract closures during repetitive action sequences. Additional analyses examining error patterns in the sequential DDK task will be discussed.

1pSC18. Long-lag voice-onset time in English onsets is longer when codas are voiced, and listeners know it. Chandan Narayan (York Univ., 4700 Keele St., Toronto, ON M3J1P3, Canada, chandann@yorku.ca)

The relationship between coda voicing and the duration of a preceding vowel is well known: vowels are longer before voiced consonants than before voiceless ones (*e.g.*, Chen, 1970). Voice-onset time (VOT) in CVs is positively correlated with vowel duration and most evident in languages like English with long-lag oro-laryngeal timing (Narayan, 2023). Weismer's (1979) study of English speakers' ($n = 3$) VOT in CVCs found that coda voicing affected word-onset VOT for voiceless stops, which was longer ($\sim$3–6 ms) when codas were phonologically voiced. A larger study of onset VOT in English speakers' ($n = 20$) CVC productions showed results consistent with Weismer (1979), but with differences between VOT being slightly greater magnitude. A perception study was conducted to examine whether listeners are sensitive to the relationship between onset VOT and coda voicing. English listeners ($n = 26$) heard sentences where the final (target) word was gated such that only long lag aspiration and one glottal cycle was presented. In a 2AFC task listeners guessed whether the aspiration came from a voiced- or voiceless-coda nonword. Consistent with the production data, short aspiration (25–50 ms) onsets reduced the probability that listeners identified the target word as having a voiced coda relative to long aspiration (125–150 ms) onsets. Results are discussed in terms of onset long-lag VOT duration as a long-distance cue to coda voicing and listener knowledge of aero-articulatory mechanisms for the relationship between VOT and vowel duration.

1pSC19. The role of aerodynamic timing in stop voicing: Evidence from true voicing and non-voicing languages. Le Xuan Chan (Dept. of Linguist., Univ. of Pennsylvania, 3401-C Walnut St. Ste. 300, Philadelphia, PA 19104, lxchan@sas.upenn.edu) and Jianjing Kuang (Dept. of Linguist., Univ. of Pennsylvania, Philadelphia, PA)

While aerodynamic factors are well established as crucial to laryngeal contrasts, little is known about how aerodynamic correlates are timed to vocal fold movement. In this work, we examine the mechanisms behind *initiating*, *sustaining*, and *inhibiting* voicing in stop consonants, drawing on EGG and aerodynamic data from L1 Malay (true-voicing) and L1 Mandarin (non-voicing) speakers in Malaysia. Malay /b,p,m/ and Mandarin /p, p^h,m/ were elicited in phrase-initial and intervocalic (phrase-medial word-initial, word-internal) positions. We examine the trajectories of intraoral pressure and voicing amplitude (EGG) during the stop closure, and find that the timing and rate of pressure changes play a key role in manipulating voicing. To *initiate* (phrase-initial /b/) and *sustain* (intervocalic /b/) voicing, pressure rises slower and later to facilitate vocal fold vibration. Voicing does not cease but decreases in amplitude as pressure increases. To *inhibit* voicing (intervocalic /p, p^h/), pressure rises sharply and peaks early around 30% of the closure. While pressure rises, voicing drops sharply in amplitude and ceases completely also at the 30% mark. After voicing ceases, pressure remains high but constant until the release. Together, these findings show that distinct voicing outcomes across phonetic environments are achieved through systematic differences in aerodynamic timing and control.

1pSC20. Global articulatory entrainment and conversational success in spontaneous dyadic speech. Thanushi R. Withanage (Elec. and Comput. Eng., Univ. of Maryland, Rm. 3180, A.V. Williams Bldg., College Park, MD 20742, thanushi@umd.edu) and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, College Park, MD)

Speech accommodation is a fundamental mechanism underlying successful social interaction and has been widely studied across multiple modalities. However, articulatory entrainment during spontaneous conversation remains underexplored, with prior work focusing primarily on local alignment or phoneme-level precision rather than global, interaction-level trend. In this study, we analyzed 1650 spontaneous, task-free dyadic conversations from the CANDOR corpus involving previously unacquainted interlocutors. Speech signals are transformed into six vocal tract variables (TVs) using an acoustic-to-articulatory speech inversion model, eliminating the need for invasive measurement techniques, such as EMA. To characterize articulatory coordination, we computed channel-delay correlation matrices from the TVs and derived rank-ordered eigen spectra as compact representations of articulatory system correlation and complexity. Results show that articulatory coordination becomes complex over the course of interaction, consistent with Lindblom's Hypo-Hyper articulation theory, which posits a continuum of articulatory adjustment driven by listener's feedback. Concurrently, interlocutors exhibit convergence in eigenspectral separation, indicating global articulatory entrainment. Importantly, stronger eigenspectral convergence is associated with higher participant-rated conversational success, demonstrating that long-term convergence in articulatory control structure characterizes more successful interactions.

1pSC21. Vowel hyperarticulation in Mandarin foreigner-directed speech. Shang-Yune S. Tang (Dept. of Linguist & TESOL, The Univ. of Texas at Arlington, 601 S. Nedderman Dr., 5th Fl. University Hall, Arlington, TX 76013, shawn.tang@uta.edu)

When speakers modify their speech to accommodate the listener's needs, they use a special speaking style called clear speech (CS). If CS is a listener-oriented speech style, we might expect to see granularity in CS modifications based on the degree of accommodation listeners require. I tested this prediction by examining how L1 Mandarin speakers adjusted their corner vowel productions when addressing high- and low-proficiency non-native listeners. Nineteen L1 speakers completed a word-list reading task. For the analysis, speakers' productions were compared across blocks (native-directed speech, NDS; foreigner-directed speech high-proficiency, FDS-high; FDS-low) using linear mixed effects models, with a separate model for each vowel (/a/, /i/, /u/) and formant (F1, F2) pair. Across models, block was the strongest predictor of acoustic variation. From NDS to FDS, /a/ lowering and backing, /i/ fronting, and /u/ raising and backing were significant. Together, these adjustments created larger vowel spaces. Speakers also significantly reduced F1 of /u/ and F2 of /a/ and /u/ when addressing low-proficiency listeners relative to high-proficiency listeners. Analyses of two-way interactions with block revealed that prior experience interacting with non-native Mandarin interlocutors and block order influenced the degree of speech adjustments speakers made. Altogether, the findings suggest that speakers are sensitive to the different magnitudes of speech accommodation that different listeners require and adjust their speech accordingly.

MONDAY AFTERNOON, 11 MAY 2026 SALON D, 1:00 P.M. TO 3:45 P.M.

Session 1pSP

Signal Processing in Acoustics, Acoustical Oceanography and Animal Bioacoustics: Signal Processing and Modeling for Distributed Acoustic Sensing (DAS) II

Abner C. Barros, Cochair
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Léa Bouffaut, Cochair
K. Lisa Yang Ctr. for Conservation Bioacoust., Cornell Univ., Cornell Lab. of Ornithology, 159 Sapsucker Eoods Rd., Ithaca, NY 14850

Invited Papers

1:00

1pSP1. Toward physics-preserving Lossy compression for distributed acoustic sensing. Aleix Segui (Dept. of Eng. Sci., Univ. of Oxford, Linacre College, Oxford OX13JA, United Kingdom, aleix.seguiugalde@eng.ox.ac.uk), Andreas Fichtner (Dept. of Earth and Planetary Sci., ETH Zürich, Zürich, Switzerland), Arantza Ugalde (Inst. of Marine Sci. (ICM-CSIC), Barcelona, Spain), and Wes Armour (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Distributed acoustic sensing (DAS) produces spatially dense measurements of fiber-optic strain or strain-rate data at rates that challenge storage, telemetry, and scalable analysis. DASPack is an open-source framework for DAS data that implements error-bounded compression while exploiting spatio-temporal redundancy. It demonstrated that substantial data reduction is possible by combining traditional lossless compression with a controlled layer of loss while remaining compatible with established DAS processing workflows.

Building on this foundation, this work explores how lossy compression for DAS can be deliberately structured to preserve physically meaningful signal content, rather than optimizing generic waveform fidelity. Many DAS applications depend primarily on a restricted set of physical signal attributes, including band-limited wavefield energy, spatial coherence along the array, inter-channel correlations, and event-level features relevant to detection, classification, and picking. By explicitly accounting for these characteristics during compression design, fidelity can be preferentially maintained for coherent, physically relevant signal components, while greater loss is tolerated in noise-dominated or physically irrelevant parts of the data. This work provides a foundation for physics-aware, learned compression of DAS data that maintains scientific reliability.

1:20

1pSP2. Detection and characterization of marine acoustic signals using distributed acoustic sensing and deep learning. Chun Zhang (Earth and Planetary Sci., Univ. of California at Berkeley, Berkeley, CA 94720, chun.zhang@berkeley.edu), Weiqiang Zhu, Barbara Romanowicz, Richard Allen (Earth and Planetary Sci., Univ. of California at Berkeley, Berkeley, CA, chun.zhang@berkeley.edu), Kenichi Soga (Civil and Environ. Eng., Univ. of California at Berkeley, Berkeley, CA, chun.zhang@berkeley.edu), and Yuxin Wu (Lawrence Berkeley National Lab., Berkeley, CA, chun.zhang@berkeley.edu)

Understanding marine seismic and acoustic activity requires continuous observations over large spatial scales, which have traditionally been limited by sparse instrumentation and short deployment durations. Distributed acoustic sensing (DAS), which transforms fiber-optic cables into dense sensing arrays, offers a unique opportunity for large-scale and continuous ocean monitoring. To leverage this capability, we developed DASNet, a deep learning framework for automated detection and classification of diverse signal types in DAS data, and applied it to a continuous submarine DAS dataset (2021–present) from the SeaFOAM deployment in Monterey Bay. Over three years of monitoring, DASNet detected hundreds of thousands of signals, including seismic events, *T*-waves, marine mammal vocalizations, and anthropogenic activity. For geophysical signals, DASNet detects local earthquakes and *T*-waves associated with distant seismic and volcanic sources, enabling characterization of coherent long-range acoustic propagation. For marine bioacoustics, DASNet identifies and tracks over 300 000 blue and fin whale calls, revealing temporal variability consistent with hydrophone observations. In addition, vessel traffic detected near the cable is validated against AIS records, demonstrating the potential for monitoring human activity. These results show that combining DAS with deep learning enables long-duration monitoring of diverse marine acoustic phenomena and highlights the potential of submarine DAS arrays.

1:40

1pSP3. Real-time, commercial implementation of machine learning for distributed acoustic sensing over ∼100 000 km of cumulative optical fiber deployments. Giovanni Milione (Sintela, Inc., 935 Technol. Dr., Ann Arbor, MI 48108, gio@sintela.com) and David Hill (Sintela, Ltd., Dorchester, United Kingdom)

In distributed acoustic sensing (DAS), a 100 km-scale optical fiber is transformed into ∼100 000 independent, synchronized, and meter-spaced acoustic sensors, each having the combined bandwidth of geophones, accelerometers, and microphones. Recently, there has been significant interest in applying machine learning to DAS. Machine learning can extract actionable information from otherwise esoteric DAS data. However, real-time, commercial implementation remains a challenge due to the intensive computation required for ∼100 000 sensors sampling at kHz rates (Gbps-scale data). This work overviews the real-time, commercial implementation of machine learning for DAS. This includes novel, domain-knowledge-based representations of DAS data in multiple dimensions (*e.g.*, space, time, and frequency), their use in deep learning-based artificial neural networks, and real-time deployment *via* edge-based GPUs. We overview ∼100 000 km of cumulative commercial and academic DAS deployments that are disruptively innovating situational awareness in physical security, physical monitoring of the natural environment, and physical monitoring of critical infrastructure, such as telecommunications, transportation, and energy.

Contributed Papers

2:00

1pSP4. Low-frequency whale call detection in distributed acoustic sensing data. Eric R. Snyder (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Supsucker Woods Rd., Ithaca, NY 14850, esnyder@cornell.edu), Holger Klinck, and Léa Bouffaut (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Distributed acoustic sensing (DAS) adds a spatial component to conventional acoustic recordings, providing positional insights for baleen whale monitoring. Compared to traditional PAM methods, DAS drastically increases data volumes (many TBs/day), requiring real-time analysis solutions. Deep learning methods are particularly well-suited for efficient data analysis of these datasets and ultimately, the generation of actionable data products, but require labeled data for training and evaluation. Here, we present a dedicated and efficient workflow for creating strong labels from DAS data, building on the DAS4Whales Python package. Our process includes: filtering and compressing datasets using sparse matrices in the f–k domain (ranging from 66 to 99% compression); standardizing the data structure and storage format by interpolating datasets to a sampling rate of 200 Hz and an element spacing of 8 m; and a user-interface prototype which allows for fast and intuitive labeling. We use this workflow to strongly label data from Svalbard, Norway, and offshore of Oregon, USA, with fin and blue whale calls, in multiple linked representations that include time, space, and frequency. This dataset is then used to train and evaluate a deep learning model to detect baleen whale calls in DAS data.

2:15–2:30 Break

2:30

1pSP5. A roadmap to processing distributed acoustic sensing data for marine mammal monitoring. Léa Bouffaut (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Cornell Lab. of Onithology, 159 Sapsucker woods Rd., Ithaca, NY 14850, lea.bouffaut@cornell.edu) and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Distributed acoustic sensing (DAS) enables real-time data collection with high spatio-temporal resolution while reducing energy and data storage constraints compared to traditional real-time acoustic monitoring packages. DAS has demonstrated success in detecting low-frequency baleen whales (*e.g.*, blue, fin, Omura's whales), and ongoing research is investigating its extended ability to monitor higher-frequency baleen whales (*e.g.*, bowhead, and humpback whales) and low-frequency odontocetes (*e.g.*, killer and sperm whales). DAS represents a cost-effective and scalable alternative to traditional monitoring methods, providing comprehensive data on marine

mammal populations that enables adaptive, real-time management strategies for more effective species conservation. However, DAS technology presents significant challenges: lower signal to noise ratios than traditional hydrophones, multiple sources of variability between datasets (*e.g.*, optical probe and processing, optical properties of the instrumented fiber, and mechanical properties of the cable and surrounding media), and unprecedented data volumes. Therefore, DAS requires the development of dedicated analysis pipelines to extract relevant information (*e.g.*, recognition, localization) for multi-stakeholder data use. Here, we provide a roadmap for advancing DAS as a transformative technology for marine mammal monitoring.

2:45

1pSP6. Modeling background noise for distributed acoustic sensing using elastic lattice-driven scatterer dynamics. Masayoshi Sakakura (Univ. of Illinois at Urbana Champaign, 1003 West Clark St. 1, Urbana, IL 61801, ms4@illinois.edu), Diego A. Cuji (Dept. of Elec. and Comput. Eng., Stony Brook Univ., Boston, MA), and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

Statistical models of distributed acoustic sensing (DAS) signals commonly follow Hartog's Rayleigh backscattering framework, in which scatterer locations and scattering amplitudes are treated as independently and identically distributed and resampled between realizations. While this assumption is often justified by the stochastic nature of polarization and microscopic scattering, it neglects the possibility that effective scatterer motion is driven by physically causal mechanical dynamics. In this work, we hypothesize that a measurable component of DAS background noise statistics arises from mechanically induced, spatially correlated perturbations of scatterer locations. To investigate this, we couple Hartog's discrete scatterer model with an elastic lattice model (ELM) that governs the spatio-temporal evolution of strain and displacement along the fiber. Material points are defined along the sensing cable and evolve according to elastic dynamics; scatterer positions are then advected *via* linear interpolation of the local displacement field rather than resampled independently. Using this hybrid ELM–scatterer framework, we simulate the resulting complex backscattered electric field and analyze its statistical structure. Power spectral density, spatial covariance, and related statistics are compared against experimental DAS recordings acquired under nominally quiescent conditions. These findings suggest a physics-informed pathway for modeling and interpreting DAS noise beyond purely stochastically i.i.d. assumptions.

3:00

1pSP7. Characterizing spatially coherent clutter measured across distributed fiber networks in a hard-rock environment. Adrian K. Doran (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, adrian.k.doran@usace.army.mil), Matthew J. Kamrath, Scott M. Slone, Aaron Meyer, Meghan Quinn, and Constantine Coclin (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Distributed acoustic sensing (DAS) networks enable dense spatial sampling of an ambient seismic and acoustic soundscape by measuring strain-rate at discrete intervals along a fiber-optic cable. Clutter, defined here as ambient noise exhibiting spatial and temporal coherence across multiple sensing channels, poses a challenge to detection, classification, and localization algorithms applied to these networks because the coherent noise can appear similar to, or otherwise mask, signals of interest. The statistics of clutter are environment-specific, array-specific, and can be challenging to characterize due to the spatial variability of environmental conditions in which the sensing element is placed. We examined clutter-rich datasets collected on two phase-sensitive DAS arrays surface-deployed over a hard-rock environment in central Colorado. A classification scheme was developed to characterize clutter events in four dimensions: spatial length scales, temporal scales, frequency content, and relative amplitude. Geospatial dependencies were identified in the clutter and associated with deployment conditions, although insufficient supporting environmental information was available to link clutter parameters to specific geophysical or atmospheric conditions. Finally, we reproduced the observed clutter distributions by generating synthetic DAS data from our statistical models and from a deep-learning approach.

3:15

1pSP8. Sensitivity of distributed acoustic sensing of a trenched optical fiber to underwater sound. Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

Distributed acoustic sensing (DAS) of underwater fiber-optic cables has emerged as powerful means of characterizing wave processes in the ocean. While acoustic applications of DAS are now common, the physics-based understanding of the transfer function from acoustic pressure in water to DAS signal and its dependence on environmental parameters remains incomplete. This paper aims to extend to buried cables the recently developed theoretical model of DAS pressure sensitivity in fluids [O. A. Godin, *J. Acoust. Soc. Am.*, **158**(4), A279–A280 (2025)]. We consider an idealized model of DAS cable as a long, cylindrical, clad or unclad, single-mode optical fiber buried at a constant depth below horizontal seafloor. Diameter of the cable is small compared to its distance to the seafloor and the wavelengths of compressional and shear waves in the seabed. Strains and stresses in the fiber and the resulting DAS signal are found by solving analytically the problem of scattering of a plane acoustic wave incident from fluid on a solid half-space with embedded elastic cylinder. Occurrence of scattering resonances, which strongly affect the angular and frequency dependence of DAS sensitivity, is investigated. The theoretically predicted DAS transfer function is compared to the models proposed in the literature.

3:30

1pSP9. Abstract withdrawn.

Session 2aAAa

Architectural Acoustics: Student Design Competition (Poster Session)

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

Molly Smallcomb, Cochair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, IL 60604

This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significant importance.

Design Scenario: A city orchestra wishes to construct a new 13,000-seat music pavilion with covered and lawn audience seating to serve as its summer home. The music pavilion will have a variety of programing acts, with the orchestra as a resident company during the summer season. Additionally, the pavilion will host touring acts, like pop and rock shows to balance out revenue stream in support of the orchestral program. The pavilion should be sized to accommodate 3,000 seated audience members under a roof and 10,000 audience members in an open lawn. The anticipated number of audience members is 7,500 people for orchestra performances and 13,000 people for touring acts.

Entries to the 2026 competition will be posted in this session for viewing. Awards are made possible through a generous donation from the Wenger Corporation to the Newman Student Award Fund.

Session 2aAAb

Architectural Acoustics, Noise and ASA Committee on Standards: Acoustics in Public Accommodations Spaces Part I

Kenneth Good, Cochair
Architecture Acoust., Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17601

Keely M. Siebein, Cochair
Siebein Acoust., 625 NW 60th St., Suite C, Gainesville, FL 32607

Samuel H. Underwood, Cochair
Univ. of Nebraska-Lincoln, 1110 S 67th St., Omaha, NE 68182-0816

Chair's Introduction—8:15

Invited Papers

8:20

2aAAb1. Signal-to-noise ratio in restaurants: Fine lines between terrific and terrible dining experiences. Brandon Cudequest (Acentech, 33 Moulton St., Cambridge, MA 02138, bcudequest@acentech.com)

Speech intelligibility in restaurants can be predicted using the signal-to-noise ratio. The talker's level, or signal, will depend on the ambient noise level due to the Lombard effect. This presentation will review models for calculating both the noise and signal levels for different restaurant designs and highlights salient parameters. Acceptable signal-to-noise ratios are then derived from literature and crowdsourced data of patrons' subjective ratings for conversational ability. Finally, these signal-to-noise ratios are adjusted to different hearing conditions, such as individuals with age-related hearing loss (presbycusis) or autism. The outputs from these models reinforce practical ways to achieve acceptable signal-to-noise ratios if sufficient information is known about the age and hearing conditions of the diners.

8:40

2aAAb2. Challenges in application of signal-to-noise ratio to dining room acoustics. Noral D. Stewart (3000 RDU Ctr. Dr. Ste. 119, Morrisville, NC 27560, norals2020@sacnc.com), Joseph F. Bridger (Stewart Acoust. Consultants, Raleigh, NC), and Brandon Cudequest (Threshold Acoust., Chicago, IL)

Prior work (Stewart and Bridger, 2020) confirmed mathematically that the ratio of absorption per talker and the talker-to-listener distance are the most important factors determining the signal-to-noise ratio in dining rooms, but that additional factors are important and when considered can reduce that amount of absorption needed to achieve a given signal-to-noise ratio. This presentation will provide a more comprehensive discussion of the practical application of those results, the challenges involved in dining room acoustics, and the research still needed.

9:00

2aAAb3. The noisy picnic: A free-field analogy to determine recommended minimum table spacing in restaurants. Nick H. Lin (Durham School of Architectural Eng. and Construction, Univ. of Nebraska–Lincoln, 3024 Harney St., Omaha, NE 68131, nlin6@huskers.unl.edu) and Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska–Lincoln, Omaha, NE)

Predicting conversational noise requires coupling acoustic propagation with Lombard driven vocal changes. We present a hybrid simulation framework intended as the computational core of a fast restaurant space-planning tool for predicting conversational sound levels across realistic table layouts. The model consists of three parts: (i) layout and local party geometry, (ii) acoustic transmission based on distance attenuation, directivity, and ambient contributions, and (iii) a Lombard control law that treats each talker's vocal level as a state variable governed by first-order dynamics that compensate for intelligibility loss at the listener (e.g., via a speech-band proxy, such as band-limited SNR). Simultaneously, intermittent speech sources are modulated by a conversational discrete event simulation (DES) model. To focus on the effects of seating layout and establish minimum table spacing guidelines, we adopt a free-field assumption that resembles a thoroughly treated restaurant or an outdoor picnic. We compared model predictions to field measurements from ten occupied restaurants in Omaha, Nebraska, using a dataset of 5 s A-weighted equivalent sound pressure levels sampled across multiple locations. Across venues, predicted spatial level patterns and overall statistics were consistent with observations, supporting the framework's use as a fast-screening tool for table spacing and simple acoustic interventions.

9:20

2aAAb4. Correlating online user reviews and acoustical conditions in Omaha area restaurants. Brandon Cottle (Durham School of Architectural Eng., Univ. of Nebraska–Lincoln, 3024 Harney St., 106, Omaha, NE 68131, 98235801@nebraska.edu), Samuel H. Underwood, Ryan Yost, and Lily M. Wang (Durham School of Architectural Eng., Univ. of Nebraska–Lincoln, Omaha, NE)

Previous work has identified that many restaurants exhibit complex acoustical conditions that prove challenging for verbal communication. One method of assessing the occupant experience is the evaluation of online user reviews. In this study, occupied measurements of sound pressure level and occupancy count and unoccupied measurements of reverberation time and spatial decay rate of A-weighted speech were recorded in 25 restaurants in Omaha, Nebraska, and the surrounding area. Acoustical measurements are then correlated with experiences communicated in online user reviews for each venue. Reviews are manually annotated and flagged to identify occurrences of noise complaints, communication difficulties, and other acoustically relevant comments. Results from this study can be used to inform the development of design guidelines for favorable acoustical conditions in restaurants.

9:40

2aAAb5. Restaurant of the future: Experiential dining acoustics. Keely M. Siebein (Siebein Acoust., 625 NW 60th St., Ste. C, Gainesville, FL 32607, ksiebein@siebeinacoustic.com), Abigail Gulley, Jennifer R. Miller, Marylin Roa, Matthew Vetterick, Gary W. Siebein (Siebein Acoust., Gainesville, FL), and Gary Siebein Jr. (Consulting, Siebein Assoc. Inc., Gainesville, FL)

As dining environments become increasingly experiential, acoustics play a central role in shaping comfort, perception, and sensory response in public accommodation spaces. This study examines how intentional soundscape design can support differentiated dining experiences within restaurants and mixed-use hospitality projects. Case studies of specially designed restaurants and restaurant–hotel combinations explore strategies for creating distinct acoustic environments that align with different phases of the dining experience, such as aperitifs, main courses, and desserts. Topics include controlling sound transmission between vertically stacked and horizontally adjacent occupancies, managing open kitchens adjacent to dining areas, and using architectural elements—such as absorptive finishes, acoustical buffers and filters, such as corridors and transitional spaces, and room-to-room separation—to limit sound bleed while maintaining spatial continuity. The presentation also discusses the role of music selection and room acoustics in influencing diner behavior, pacing, and perceived taste, including instances where curated soundtracks are paired with culinary intent for purposeful listening and dining effects. These projects position sound as an active design parameter that contributes to experiential quality, comfortable socialization, and overall satisfaction in contemporary restaurant and hospitality spaces.

10:00–10:15 Break

10:15

2aAAb6. Intentional design for social connection: Creating effective public gathering spaces through strategic spatial and acoustic planning. Josh Gregoire (Aspen Group, 9645 Lincolnway Ln., Ste. 201, Frankfort, IL 60423, jgregoire@aspengroup.com) and Derek DeGroot (Aspen Group, Frankfort, IL)

Community gathering spaces serve critical social functions, yet their effectiveness depends heavily on intentional design decisions that facilitate or inhibit human connection. This paper examines how coordinated design elements, including acoustics, spatial arrangement, materials, and circulation patterns, encourage desired interactions. The coordination of sight lines, speech intelligibility, background noise control, furniture scale and arrangement, flooring materials, ceiling geometry, and lighting significantly impacts whether spaces function as intended. These elements create environments where people intuitively interact with space and each other in desired ways, respecting proxemic comfort zones while encouraging appropriate social engagement. Our work demonstrates these principles in functional areas: entry zones designed for confident wayfinding, hospitality spaces balancing conversational intimacy with community visibility, quiet retreat areas supporting contemplation through acoustic separation, and engagement zones facilitating participation. Observations reveal successful spaces share common characteristics: acoustic environments matched to activity requirements, spatial configurations supporting varying social distances, and layered design cues guiding intuitive navigation. These principles apply to restaurants, cultural institutions, civic centers, and other venues where physical environment shapes social outcomes and community formation.

10:35

2aAAb7. Exploring sonic environments in relation to the quality of life of older adults. Saleh Naseer (College of Design, Construction, and Planning, Univ. of Florida, 2510 NE 9th St., #711, Gainesville, FL 32609, saleh.naseer@ufl.edu) and Hassan Azad (College of Design, Construction, and Planning, Univ. of Florida, Gainesville, FL)

The sonic environment plays an important role in shaping quality of life (QL), particularly for older adults living in care facilities, where daily well-being is closely linked to environmental conditions. This study proposes investigating how sonic experiences relate to the physical, psychological, social, and environmental aspects of QL among older adults. An Integrated Sonic Experience with Quality of Life (SE-QL) subjective assessment tool, developed in a previous study, is applied, combined with objective acoustic measurements, to capture both perceptual and physical characteristics of the sonic environment. The study involved interview-based questionnaires administered to residents in a residential care facility, with the collection of demographic information and acoustic parameters. The analysis examined associations between the environmental QL domain and other QL domains, with particular attention to the analytical role of environmental factors. By integrating subjective and objective approaches, this research aims to provide a clearer understanding of how sonic environments contribute to the enhancement of older adults' QL. The findings are expected to inform future design and management strategies for acoustic quality in care settings, supporting healthier and more supportive living environments for older adults.

10:55

2aAAb8. The acoustical design of the Hudson Yards Subway Station. Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, STE 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

The Hudson Yards neighborhood, built above the commuter rail yard adjacent to the Hudson River on Manhattan's West Side, features luxury residences, high-end retail, corporate offices, restaurants, as well as public art and performance spaces. To serve this new neighborhood, the New York City subway system's 7 Line was extended from Times Square to the new 34th Street-Hudson Yards station. Ostergaard Acoustical Associates was engaged to consult on the acoustical design of the station. Concerns included overall room acoustics and acoustical comfort within the cavernous multi-level station, localized noise control at train tracks and above trains (to control train-mounted air conditioning noise), sound system intelligibility (critical for public address and NFPA requirements in emergencies), and control of HVAC system noise. This presentation will focus on the acoustical and logistical challenges of this ambitious project.

11:15

2aAAb9. Auralization of babble noise in a lobby. Ana M. Jaramillo (Olson Sound Design LLC, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu) and Bruce Olson (Olson Sound Design LLC, Brooklyn Park, MN)

During the design process of an art museum, a question arose about the acoustic characteristics of the lobby space, which would be used for exhibit openings, and other "cocktail style" gatherings. We were engaged to provide auralizations as a tool to demonstrate to the client the need for absorption in such a space. In order to set up a realistic aural environment, several anechoic voice files were selected and auralized individually, then combined as a single sound file for playback. Although the files were not presented to the client under ideal play-back conditions, we hoped that they were sufficient to convey the relative effect of absorption on background babble noise, when having a conversation at close distance.

TUESDAY MORNING, 12 MAY 2026 FRANKLIN HALL 5, 8:00 A.M. TO 11:00 A.M.

Session 2aAB

Animal Bioacoustics: Acoustics Ecology and Biological Soundscapes

Xavier Mouy, Chair
Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543

Contributed Papers

8:00

2aAB1. Synchronization parameters in fish choruses of weakfish (*Cynoscion regalis*). Joseph J. Luczkovich (Dept. of Biology, East Carolina Univ., Greenville, NC 27858, luczkovichj@ecu.edu), Mark W. Sprague, Martin Bier (Phys., East Carolina Univ., Greenville, NC), and Maciej Majka (Malopolska Ctr. of Biotechnology, Jagiellonian Univ., Krakow, Poland)

In many soniferous fishes, males use nocturnal mating calls to attract females, resulting in characteristic mating choruses dominating estuary and ocean soundscapes. Multiple fish calling in synchrony can occur if the males are coupled when they hear other males calling, changing the time of onset of a call. We analyzed synchronous calling in males of the weakfish, *Cynoscion regalis* (family Sciaenidae), using power spectral band (PSB) sums evaluated across the frequency range 300 to 600 Hz (characteristic frequency band for weakfish calls). If the fish form synchronized choruses, they should be visible as waves in the PSB sums. If synchronization were not occurring, a flat line would be observed in the PSB sum, not a wave. We computed an order parameter $r = A/B$, which is the ratio of the amplitude (A) of the PSB sum wave relative to the average PSB sum value (B), using pressure-squared units. If calling is completely in unison $r = 1.0$; $r = 0.0$ if individuals are calling at random. During the chorusing after sunset, $r = 0.7$. The synchronization of the chorus was partial because some individuals were not tightly coupled to the main chorus.

8:15

2aAB2. Patterns of co-occurrence: Vessel movements and endangered Southern Resident killer whales on Washington's outer coast. Alyssa Lopez (Fisheries, Wildlife, and Conservation Sci., Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, lopezaly@oregonstate.edu), Pablo Santiago Dominguez Sanchez (Fisheries, Wildlife, and Conservation Sci., Oregon State Univ., Newport, OR), Hannah J. Myers (College of Fisheries and Ocean Sci., Univ. of Alaska Fairbanks, Fairbanks, AK), David D. Huff (Northwest Fisheries Sci. Ctr., NOAA, East Seattle, WA), David K. Mellinger (Coop. Inst.for Marine Resources Studies, Oregon State Univ., Newport, OR), Joshua D. Stewart, and Kathleen Stafford (Fisheries, Wildlife, and Conservation Sci., Oregon State Univ., Newport, OR)

Understanding the distribution of endangered Southern Resident killer whales (SRKWs) is essential for conservation, as their presence reflects the interplay between prey availability, habitat characteristics, and anthropogenic activities. Although SRKWs have been extensively studied within Puget Sound, acoustic monitoring has revealed frequent utilization of Washington's outer-coast habitats. Increased time offshore highlights the need to

quantify SRKW outer-coast habitat use and vessel overlap, as vessel noise masks communication and echolocation, potentially reducing foraging efficiency and group cohesion. Fifteen hydrophones were deployed along Washington's outer-coast from October 2023 to December 2024 to assess spatial and temporal patterns of co-occurrence between vessels and SRKWs. Acoustic data were manually annotated for SRKW calls and processed through a trained, automated detector in BirdNET. Ship detections from each hydrophone were combined with AIS data to investigate the relationship between vessels and SRKW presence. An acoustic propagation model around each hydrophone estimated SRKW detection ranges of 2–5 km. Of 467 SRKW detections, 53 coincided with vessels, mainly during summer and earlyfall. Cargo, tanker, and fishing vessels most frequently overlapped with SRKWs, particularly near the Strait of Juan de Fuca. This framework reveals spatial patterns of SRKW-vessel co-occurrence and identifies prospective hotspots where vessel management could reduce overlap.

8:30

2aAB3. Geographic variation in echolocation clicks of Risso's dolphins (*Grampus griseus*) in the California Current. Sarah M. Moseley (San Francisco State Univ., 3150 Paradise Dr., Tiburon, CA 94920, sarah@shierscreative.com), Marie A. Roch (San Diego State Univ., San Diego, CA), Shannon Rankin (NOAA Southwest Fisheries Sci. Ctr., La Jolla, CA), Ellen Hines, and Anne Simonis (San Francisco State Univ., San Francisco, CA)

Risso's dolphins (*Grampus griseus*) produce broadband echolocation clicks that vary across regions, offering insight into population structure and habitat use. Two stocks are recognized in U.S. Pacific waters: the California–Oregon–Washington stock and the Hawaiian stock, but these designations may overlook finer-scale structure within the California Current Ecosystem (CCE). Building on evidence of geographic variation in Risso's dolphin echolocation click types across the U.S. East Coast, Gulf of Mexico, Southern California Bight, and Hawaiian Islands, spectral peak and notch frequencies were measured from Risso's dolphin echolocation clicks during three NOAA surveys between 2016 and 2023. Acoustic recordings were collected with drifting buoys, allowing broad coverage of the entire CCE region along the U.S. West Coast. Clustering of spectral features highlighted several different click types. We compared spatio-temporal variation in Risso's click types across Oregon, Northern California, and Central California, including seasonal and diel variation. These results provide the first comprehensive assessment of Risso's dolphin click characteristics across the CCE and enable comparison with other U.S. regions where click-type divergence has been observed. Fine-scale spatial, seasonal, and diel acoustic patterns can inform population assessment, ecological understanding, and ecosystem-based management.

8:45

2aAB4. Killer whale (*Orcinus orca*) seasonal acoustic presence in two widely separated regions of the Eastern Canadian Arctic. Jessica Sportelli (Conservation Biology, National Marine Mammal Foundation, 2240 Shelter Island Dr. Ste. 200, San Diego, CA 92106, Jjsportelli@gmail.com), Frances Agnes, Jack Alford (Mount Edgecumbe High School, Sitka, AK), Joey Angnatok (M/V What's Happening, Nain, Nunatsiavut, NF, Canada), Ikalyn Brower (Mount Edgecumbe High School, Sitka, AK), Keith Clarson, Dani Earls, Blake Eisbrenner (Ilwaco High School, Ilwaco, WA), Asha Foruria, Alexandra Fujioka (Mount Edgecumbe High School, Sitka, AK), Eva C. Hidalgo Pla (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), David Hill, Cecilia Kimzey (Mount Edgecumbe High School, Sitka, AK), Kasen Linthakhan, Michael Mahoney (Ilwaco High School, Ilwaco, WA), Ursula Mangrobang, Cadence Mesak, Chohla Moll (Mount Edgecumbe High School, Sitka, AK), Sarah M. Moseley (San Francisco State Univ., Tiburon, CA), Breana Nelson (Mount Edgecumbe High School, Sitka, AK), Alyssa Ortez (Ilwaco High School, Ilwaco, WA), Michelle Saunders (Dept. of Lands and Natural Resources, Nunatsiavut Government, Nain, Nunatsiavut, NF, Canada), Anne Simonis (San Francisco State Univ., San Francisco, CA), Kristin Westdal (Oceans North, Vancouver, BC, Canada), and Joshua M. Jones (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

Killer whale presence is well-documented in the Eastern Canadian Arctic (ECA). Their movements and habitat use are likely correlated with the distribution of prey species (e.g., narwhals and seals) and constrained by the seasonal advance and retreat of sea ice. However, questions remain about killer whale ecology in the Arctic, including their movements and subsequent impacts of predation on endemic cetacean populations. Killer whales are readily identified in underwater acoustic recordings from their stereotyped pulsed calls, enabling studies of movements and distribution in widely separated acoustic monitoring sites. Year-round acoustic recordings from Nunavut (2017–2024) and Nunatsiavut (2021–2024), Canada were analyzed for killer whale pulsed calls by students from two U.S. high schools. Killer whale calls were seasonally present during open water in both regions, with one call detected in both Nunatsiavut and Nunavut. These findings contribute to a growing understanding of the distribution and movement patterns of these apex predators in the ECA; a priority for Inuit communities where killer whales may compete for access to marine mammals hunted for subsistence. This collaborative research project includes youth, Inuit residents of the study region, and collaborators from multiple U.S. and Canadian institutions, reflecting a powerful, multigenerational approach to research.

9:00

2aAB5. Tusåven? Spatio-temporal presence of odontocetes in the waters of the Torngat Mountains Area of Interest (Nunatsiavut). Eva C. Hidalgo Pla (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, echidalg@ucsd.edu), Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Michelle Saunders (Nain, Nunatsiavut, NF, Canada), Joey Angnatok (Nain, Nunatsiavut, NF, Canada), Rodd Laing (Nunatsiavut Government, Nain, Nunatsiavut, NF, Canada), Sid Pain (Oceans North, Nain, Nunatsiavut, NF, Canada), John Hildebrand, and Joshua Jones (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Baseline information on marine mammal spatiotemporal presence is essential for understanding ecological responses to rapid environmental change, including increasing underwater noise, and for informing marine spatial planning. This study presents the first multi-year passive acoustic assessment of odontocete species richness and seasonal presence in the Torngat Area of Interest (TAOI) in Nunatsiavut, within the inner northern Labrador Shelf. Passive acoustic recordings from 2021–2024 at two year-round hydrophone moorings in the central and southern TAOI were analyzed for echolocation clicks. A semi-automated detection and classification pipeline clustered signals by spectral characteristics, inter-click intervals, and duration. A trained deep neural network assigned detections to dominant classes, with species identifications confirmed through expert review. Across the monitoring period, at least five odontocete species were detected, including harbour porpoises, white-beaked dolphins, sperm whales, and belugas. Most species were primarily detected between aujak (summer) and ukiak (fall). In contrast, belugas exhibited peak acoustic presence during the ice-covered season, underscoring the TAOI as a potential overwintering habitat. These findings provide new insight into habitat use in a region where visual surveys are limited. Led by the Nunatsiavut Government, this work supports the Imappivut Marine Plan and ongoing efforts toward designating the TAOI as an Inuit Protected Area.

9:15

2aAB6. Impacts of retreating sea ice on marine mammal detections in the Chukchi Sea. Miriam Hauer-Jensen (Oregon State Univ., 1500 SW Jefferson Way, Corvallis, OR 97331, hauerjem@oregonstate.edu), Kathleen Stafford (Oregon State Univ., Newport, OR), and Maria Kavanaugh (Oregon State Univ., Corvallis, OR)

Acoustic data from the Chukchi Ecosystem Observatory (2021–2025) were analyzed to examine how sea ice influences the habitat use of sea ice-obligate and ice-adapted marine mammals in the Chukchi Sea. Vocalizations were visually identified in Ishmael 3.0 to produce time series of species presence and seasonal occurrence. Daily, sea ice concentrations derived from satellite microwave brightness were averaged within 25 km of the mooring and compared with daily vocalization rates of bearded seals and Pacific walruses. Bearded seal calling peaked in spring, consistent with their mating season, and coincided with high sea ice cover. Walrus detections were scarce during peak ice in midwinter to spring but increased as ice retreated in late summer, suggesting seasonal habitat use during breakup.

Acoustic detections also revealed November as a key period for baleen whale presence, with bowhead and humpback whales dominating the late-fall soundscape. Belugas were rarely detected, while bowhead calls occurred across multiple years, and fin whale calls were identified in a subset of recordings. By documenting seasonal and interannual trends in vocal activity, this study provides insight into how changing sea ice conditions shape the distribution and behavior of marine mammals vital to Arctic ecosystems and coastal communities.

9:30–9:45 Break

9:45

2aAB7. Narwhal behavioral responses to underwater noise from ships before and after regional voluntary speed reduction measures in Eclipse Sound, Nunavut. Maxima Resano (La Costa Canyon High School, 1 Maverick Way, Carlsbad, CA 92009, maxima.resano@gmail.com), Peyton Steffek (Scripps Inst. of Oceanogr., La Jolla, CA), Jack P. Ewing (Dept. of Marine Resources, State of Maine, Boothbay Harbor, ME), Kristin Westdal (Oceans North Conservation Society, Vancouver, BC, Canada), Alex J. Ootoowak (Oceans North Conservation Society, Pond Inlet, NU, Canada), Rebecca Wu (Scripps Inst. of Oceanogr., San Diego, CA), and Joshua Jones (Scripps Inst. of Oceanogr., La Jolla, CA)

Narwhals, *Monodon monoceros*, rely on sound for navigation, foraging, and social communication, making them particularly sensitive to changes in the underwater acoustic environment. Economic factors, such as mineral extraction, cargo shipment, and marine tourism, have resulted in increasing ship presence in regions with historically low ship traffic, including Eclipse Sound in Nunavut. Bulk carriers have transported iron ore through the region since 2015, with a voluntary speed limit of 9 knots established in 2020. Previous studies have demonstrated narwhals' behavioral responses to relatively low levels of underwater noise from ships, raising concerns of noise impacts in Eclipse Sound. Underwater sound recordings and ship location information from Eclipse Sound were used to determine narwhal responses to ship proximity before and after the speed reduction. Narwhal presence was assessed through the detection of echolocation clicks and evaluated in relation to ship presence and speed within a 40 km radius of the recording location. The probability of narwhal acoustic detection was reduced at greater distances to ships prior to the speed reduction. These findings highlight the potential for incorporating vessel speed management into conservation strategies for ecologically and culturally important Arctic marine wildlife.

10:00

2aAB8. Kurtosis of air gun pulses in an extremely shallow-water environment in the Arctic. Shane Guan (Div. of Environ. Sci., Bureau of Ocean Energy Management, 45600 Woodland Rd., Ste. #455-C33, Sterling, VA 20166) and Tiffini Brookens (Marine Mammal Commission, Bethesda, MD)

Kurtosis (β), a statistical parameter that measures the presence of outliers in a dataset, has been increasingly used to quantify the impulsiveness of a sound field. For an impulsive sound source, such as a seismic air gun, impulsiveness is expected to decrease as the distance from the source increases due to multipath propagation. In this study, acoustic data from a shallow-water marine seismic survey in the Arctic were analyzed for kurtosis. The results indicated that in general kurtosis did not vary noticeably from 1 to 15 km from the source, which is the maximum range of the dataset. However, large differences in kurtosis values were observed between surveys conducted in waters less than 5 m in depth for a single air gun array ($\beta_{\text{median}} = 22.8$) and in water depths greater than 5 m for a double air gun array ($\beta_{\text{median}} = 154.8$). This difference is likely due to a strong low-frequency cutoff present in the extremely shallow-water environment reducing the impulsiveness of the sound. This observation highlights the importance of careful consideration of bathymetry when modeling sound fields for impact assessments in terms of impulsiveness.

10:15

2aAB9. Listening range reduction of predator and prey during shipping noise: A Gulf of Maine case study on grey seals and Atlantic cod. Ian T. Jones (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, 8 College Rd., Morse Hall, Durham, NH 03824, ian.t.jones@unh.edu) and Michelle Fournet (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

Shipping is a dominant underwater noise source globally, including throughout the Gulf of Maine (GOM), where noise is predicted to increase due to anticipated increases in vessel traffic to and from opening Arctic trade routes. Grey seals and Atlantic cod are protected and managed species whose populations in the GOM have rapidly increased and declined, respectively. Shipping noise can interfere with, or mask, these animals' ability to detect ecologically relevant sounds critical to navigation, foraging, predator avoidance, and reproduction. Harnessing data from the Atlantic Ecosystem Observatory Network in the GOM, this study investigated masking potential due to shipping noise of ambient soundscape detectability by Atlantic cod and migratory young-of-year grey seals that predate cod among other fishes. Masking potential was modeled via Listening Range Reduction (LRR), using sound pressure data for both species and additionally particle motion for cod. Increases in vessel-associated ambient noise in summer to early fall were associated with increased LRR for both species. This study highlights limitations vessel noise in the GOM may place on the ability of young-of-year pinnipeds to detect ecologically relevant soundscape cues including those of their prey, and the ability of mid-trophic fishes to detect predators.

10:30

2aAB10. Noise-induced hearing loss susceptibility in the short-tailed fruit bat (*Carollia perspicillata*). Keegan Eveland (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, kevelan1@jh.edu), Grace Capshaw (Johns Hopkins Univ., Baltimore, MD), Amanda M. Lauer (Otolaryngol.-HNS, Johns Hopkins Univ. School of Medicine, Balitmore, MD), and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Hearing supports behaviors critical for survival across vertebrates and is especially important for echolocating bats that depend on sound for navigation, foraging, and communication. Although most mammals experience hearing loss due to age or noise exposure, echolocating bats appear less susceptible to both age-related and noise-induced hearing loss. Most evidence for this resilience comes from *Eptesicus fuscus*, an insectivorous bat that relies on laryngeal echolocation to track and intercept insect prey. Notably, the Egyptian fruit bat, a lingual echolocator that relies on vision and olfaction to forage, exhibits age-related hearing loss. The factors underlying these species differences remain poorly understood. Here, we examined susceptibility to noise-induced hearing loss in *Carollia perspicillata*, a fruit bat that relies on laryngeal echolocation, along with vision and olfaction, to navigate and forage. We measured auditory sensitivity and outer hair cell functionality before and after noise exposure using auditory brainstem responses (ABRs) and distortion product otoacoustic emissions (DPOAEs). Substantial permanent threshold shifts were not observed in noise-exposed bats. These preliminary results suggest that *C. perspicillata* may possess mechanisms that protect against noise-induced hearing loss. These data add to the growing body of evidence that laryngeal echolocating bat species are resistant to noise exposure.

10:45

2aAB11. Like a moth to a flame: Effects of a fireworks show on big brown bat feeding behavior. Kerri D. Seger (Integral Consulting Inc., 719 2nd Ave., Seattle, WA 98104, kerriseger@re-1.net) and Greg Cotten (Integral Consulting Inc., Santa Cruz, CA)

For the October 2025 annual fireworks show in Capitola, CA, the city requested biological monitoring be done to document reactions of wildlife with a Before-After/Control-Impact (BACI) study. The monitoring program consisted of visual and acoustical monitoring, but this talk will focus on the acoustics. It will present the differences in big brown bat (*Eptesicus fuscus*) calling behavior for the half hour during the fireworks compared to the half hour after and the half hour before. Using terminal phases as a proxy for

prey consumption and search and approach phases as a proxy for feeding effort, the data tell a story about big brown bat feeding behavior during a disruptive noise event. Namely, searching for food occurred at a reduced effort during the fireworks, then rebounded quickly, with the most calling activity of the monitoring window occurring 12 min after the show ended. Despite this rebound peak, though, the overall search effort and success of prey capture for the half hour after the show ended were below (−52% and −80%, respectively) what they had been during the half hour before the show commenced. Interpretations of whether this is a "moths to a flame" effect will be discussed.

TUESDAY MORNING, 12 MAY 2026 SALON A, 8:30 A.M. TO 11:00 A.M.

Session 2aAO

Acoustical Oceanography and Underwater Acoustics: Geoacoustic Inversion and Sediment Characterization

Robert W. Drinnan, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6785

Julien Bonnel, Cochair
Scripps Inst. of Oceanogr., 9500 Gilman Drive, # 0238, San Diego, CA 92093

Contributed Papers

8:30

2aAO1. Geoacoustic inversion of modal dispersion data using randomized blocky Occam. Julien Bonnel (Scripps Inst. of Oceanogr., 9500 Gilman Dr., # 0238, San Diego, CA 92093, jubonnel@ucsd.edu), Matthias Morzfeld (Scripps Inst. of Oceanogr., La Jolla, CA), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This talk introduces a new nonlinear inversion method, randomized blocky Occam (RamBO), recently developed for geophysical inversion. In short, blocky Occam determines the blockiest model that fits the data using l1-norm regularization of the model derivative; it does so effectively using the split Bregman method. Uncertainties are then estimated using a modified randomize-then-optimize approach. Here, we study the application of RamBO inversion to time-frequency modal dispersion data, as recorded in ocean acoustics using a single hydrophone. The method is applied to simulated data and experimental data, recorded during the seabed characterization experiment (SBCEX) on the New England Mud Patch. Results are compared to those obtained using Bayesian trans-dimensional (TransD) inversion. While based on different hypotheses, RamBO and TransD both aim to describe the seabed with a relatively simple model (blocky or layered), with complexity driven by the data information content. The advantages and drawbacks of the two methods will be discussed. [Work supported by the Office of Naval Research.]

8:45

2aAO2. Geoacoustic inversion of dispersed modes using vertical reactive intensity. Robert W. Drinnan (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6785, drinnan@uw.edu), David Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Acoustic intensity, derived from coherent, collocated measurements of pressure and acoustic velocity, contains information about the sound field that is not accessible from pressure measurements alone. In shallow-water waveguides, the vertical component of reactive intensity is highly sensitive to changes in the local waveguide environment. For dispersed modal propagation, the modal vertical intensity can be normalized by the acoustic potential energy, removing any dependence on source location and enabling multiple measurements to be directly compared and averaged. Vector acoustic measurements of dispersed pulses from explosive SUS deployed during the 2022 seabed characterization experiment are analyzed for three receiver locations on the New England Mudpatch (NEMP). The underlying sediment at these locations exhibits distinct variations in the vertical intensity field whose acoustic properties are estimated using a Bayesian inversion approach. The inferred spatial variation in surficial sediment properties is consistent with previously reported measurements for the NEMP region.

9:00

2aAO3. Deep learning seabed classification using ship noise and a porosity-based catalog. Dallin T. Harwood (Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, dallinharwood@gmail.com), Alexandra M. Hopps-McDaniel (Elec. Eng. and Comput. Sci., Univ. of Wyoming, Laramie, WY), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and David P. Knobles (Platt Inst., Austin, TX)

Sound transmission in the ocean is highly dependent on the characteristics of the sea floor. An effective seabed, suitable for long-range transmission loss modeling, can be predicted with machine learning using passive sonar. Previously, both synthetic and measured spectrograms of passing ships of opportunity (SOO) from the seabed characterization experiment (SBCEX 2017 and 2022) were used in a ResNet-18 deep learning algorithm to identify seabed type from a catalog of 34 acousto-elastic seabeds. The current study reduces the 34-seabed catalog into 15 options defined by varying porosity. Porosity is hypothesized to be a more significant parameter of the seabed and allows for more intuitive organization within a smaller

catalog. This work compares and contrasts the seabed classifications from the two catalogs for the SBCEX data to evaluate the efficacy of using a porosity-based seabed catalog. Additionally, the sound speed profiles vary between the 2017 and 2022 SBCEX data, and any differences between the performance of models trained on either data set show the significance of understanding the water sound speed to accurately predict seabed type. [Work supported by Office of Naval Research.]

9:15

2aAO4. Acoustic group velocity estimations in shallow freshwater ice fracturing. John Case (Penn State Univ., 414 ECoRE Bldg., 556 White Course Dr., University Park, PA 16802, jac7175@psu.edu), Andrew Barnard (Michigan Technolog. Univ., Houghton, MI), and Daniel C. Brown (Penn State Univ., State College, PA)

Ice fracturing noise has been widely studied in deep-water, oceanic environments. In Arctic regions, acoustic energy generated by ice breakup can represent the dominant noise source, masking underwater communication, sonar, and other naval systems. Acoustic propagation in floating ice sheets has been modeled since the mid-20th century; however, these environments remain difficult to fully characterize due to fluid–structure interactions, ice anisotropy, layering, and spatial variability in thickness. Additionally, oceanic ice-covered environments pose substantial challenges in accessibility, logistics, and cost, limiting opportunities for controlled experimentation. In this work, a small-scale propagation experiment conducted in shallow freshwater ice is presented as a testbed. A synchronouslysampled array of hydrophones, microphones, and geophones was deployed below, above, and on the ice surface, respectively. Impact locations were marked across the ice and excited using an instrumented force hammer, with propagated responses recorded on each sensor type. Group velocity estimates for acoustic propagation within the coupled ice–water system were extracted from the measured data and compared to a simple dispersive model. In addition to controlled impacts, passively generated ice fracture events were recorded and are discussed in the context of the observed propagation characteristics.

9:30–9:45 Break

9:45

2aAO5. Range dependent sediment sound speed profile estimates in the New England Mud Patch using the image source method. Jason Philtron (The Penn State Univ., University Park, PA 16804), Chad M. Smith (The Penn State Univ., P.O. Box 30, State College, PA, chad.smith@psu.edu), and Charles W. Holland (Portland State Univ., Portland, OR)

The Image Source Method (ISM) shows promise as a method for rapidly calculating seabed sound speed estimates and probing horizontal seabed variability on scales of 10s of meters. This method uses a source and horizontal line array towed near the seafloor to measure acoustic reflections from sediment layering structure. The classic image source concept and nearfield beamforming are then used at each transmission location to create a range dependent profile of sediment layer thickness and sound speed independent of other seabed parameters. Data were collected along tracks spanning 27 km of the New England Mud Patch (NEMP) in the east-west direction using the Three-Octave Research Array at Penn State and an omnidirectional GeoSpectrum M18C source between 0.5 and 6 kHz. Point comparisons of ISM results at seabed depths less than 15 m compare favorably with previous data collection and analysis using other methods, such as the reflection coefficient method, while the 2-D nature of these measurements provides information about the horizontal variability in seabed structure. Advantages of the ISM include its relative computational efficiency, and the ability to achieve sound speed estimates from a single transmission with a smaller array aperture than required by wide-angle reflection coefficient methods.

10:00

2aAO6. Geoacoustic tomography using modal dispersion data. Brandyn Butchard (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, brandynbutchard@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Julien Bonnel (Scripps Inst. of Oceanogr., San Diego, CA)

This talk presents a tomographic approach to estimate a 3-D model of seabed geoacoustic parameters and their uncertainties from the inversion of modal time-frequency (TF) dispersion data collected using multiple acoustic sources and receivers. The model is obtained by a two-step inversion process similar to that applied in surface-wave seismic tomography. The first step is a linearized, regularized inversion of the TF dispersion data to define a gridded 2-D map of the mode group velocities at multiple frequencies (i.e., modal dispersion curves). The second step applies nonlinear Bayesian trans-dimensional inversion to the dispersion curves at each grid cell to produce a depth-dependent geoacoustic model for that cell. The results are then interpolated to create a full 3D model of the seabed. The Bayesian inversion provides quantitative uncertainty estimates of the results. The two-step inversion process can applied to data collected with unsynchronized/uncalibrated receivers and can account for uncertainty in the locations of all elements. The approach is benchmarked on realistic simulated data.

10:15

2aAO7. Relating sediment geoacoustic properties to organic carbon using the grain-shearing model. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 9767, Austin, TX 76766-9767, kevin.lee@arlut.utexas.edu), Megan Ballard, Colby W. Cushing, Andrew R. McNeese, Sally A. Walker (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kenneth H. Dunton (Marine Sci. Inst., The Univ. of Texas at Austin, Port Aransas, TX)

Seagrasses have an important role in the global carbon cycle because they are highly productive coastal communities that sequester carbon in the underlying sediment. When deposits of organic matter adsorb onto the mineral surfaces, they create a compliant layer between the grain contacts, leading to a reduction in sediment dynamic stiffness. As the organic matter increases beyond a saturation threshold, the grains become separated within a high-porosity sediment organic matter suspension. This leads to strong correlations between sediment geoacoustic properties and organic carbon, and previous work has demonstrated empirical relationships between sediment compressional wave modulus and total organic carbon (TOC), [Geophys. Res. Lett. **46**, 5968–5977 (2019)]. Here, we describe new measurements collected on 29 sediment cores collected in a seagrass meadow in Corpus Christi Bay, Texas. Sediment compressional wave speed (200–1000 kHz) was measured along the length of each core, then cores were extruded, sectioned, and analyzed for wet bulk density, porosity, grain size, and TOC. Rather than relying solely on an empirical approach, Buckingham's grain-shearing model was adapted to relate geoacoustic properties to TOC, with good data-model agreement providing a robust method for estimating sediment TOC from sediment acoustic measurements. [Work supported by ARPA-E and ONR.]

10:30

2aAO8. Modeling shear-wave dispersion in fine-grained marine sediments. Nickalaus H. Anderson (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St. ETC II 5.160, Austin, TX 78712, Nickalaus@utexas.edu), Megan Ballard (Univ. of Texas at Austin, Austin, TX), Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Frequency dispersion of sound speed in marine sediments directly affects acoustic propagation, reflection, and attenuation in the seabed, with important consequences for both interpretation of acoustic measurements and predictive modeling. As part of the Seabed Characterization Experiments in the New England Mud Patch (NEMP), compressional-wave sound speed dispersion in fine-grained sediments has been characterized over a broad frequency range (20 Hz to 1 MHz). In contrast, shear-wave dispersion within the mud layer remains poorly understood, in part because broadband

shear-wave measurements are difficult to obtain in soft, fine-grained sediments. Poor coupling and low signal-to-noise ratio prove obstacles in obtaining broadband data from an individual source receiver pair. Despite these challenges, shear-wave dispersion is a key diagnostic of sediment physics, providing constraints on grain-contact stiffness, effective stress, and consolidation state. This work advances understanding of shear-wave dispersion in fine-grained sediments by applying acoustic propagation models to published low-frequency (20–50 Hz) inference results and high-frequency (800 and 1200 Hz) direct measurements collected in the NEMP. Model results reconcile the apparent disparities in shear-wave speeds inferred from these disparate frequency bands, demonstrating that frequency-dependent dispersion provides a consistent physical explanation for the observed variability. [Work supported by ONR.]

10:45

2aAO9. Geoacoustic inversion approach assessment and resolution determination. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu) and Jason E. Gaudette (Raytheon Technologies, Portsmouth, RI)

Inversion results—even when produced by carefully crafted methods and applied to data from the same site—often diverge, suggesting different interpretations of the seabed's layered structure. Data modality, the sound-propagation model, and the environmental characterization impact the results. Prior information and methodological assumptions further shape the outcome. The question is then how to determine whether a given estimate reflects the "right" answer. More specifically, it is important to understand how we assess potential biases, quantify uncertainties, and understand the resolution that each method can achieve. The aim of this work is to compare inversion approaches under differing assumptions. To do so, we rely on simulations in which the true environmental properties are known, allowing us to draw clear conclusions about the influence of modeling choices and prior constraints. We focus on two approaches, matched field Iinversion and inversion exploiting group velocity information, and we compare them in terms of accuracy and uncertainty as well as resolution in seabed characterization. For the latter approach, we extract dispersion curves with a technique relying on the fractional Fourier transform (FrFT) to identify distinct modal components. This technique follows previous work in biosonar signal analysis, specifically for separating components of bat echolocation calls. [Work supported by ONR.]

TUESDAY MORNING, 12 MAY 2026 FRANKLIN HALL 9, 7:55 A.M. TO 10:30 A.M.

Session 2aBAa

Biomedical Acoustics: Biomedical Methods: Messy, Marvelous, and Meh I

Timothy J. Hall, Cochair
Univ. of Wisconsin, Madison, 1153 WIMR, UW-Med. Phys., Madison, WI 53705

Kevin J. Haworth, Cochair
Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

Virginie Papadopoulou, Cochair
Biomedical Eng., The Univ. of North Carolina at Chapel Hill, 116 Manning Drive, 9004 Mary Ellen Jones Building, CB 7575, Chapel Hill, NC 27599-7575

Chair's Introduction—7:55

Invited Paper

8:00

2aBAa1. The upside down and rightside up of acoustic droplet vaporization: Methods, measurements, and challenges. Sander Spiekhout (Radiology, Univ. of Michigan, Ann Arbor, MI 48109), Bachir A. Abeid (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Ze Qi Chan (Radiology, Univ. of Michigan, Ann Arbor, MI), Jon B. Estrada (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Mario L. Fabiilli (Radiology, Univ. of Michigan, Ann Arbor, MI), and Mitra Aliabouzar (Radiology, Univ. of Michigan, 1301 Catherine St. 3226A Medical Sci. Bldg. I, Ann Arbor, MI, aliabouz@umich.edu)

Ultrasound-responsive droplets undergo a liquid-to-gas phase transition when exposed to focused ultrasound, a phenomenon known as acoustic droplet vaporization (ADV). ADV is investigated in a growing range of biomedical applications. Despite its increasing use, experimentally resolving ADV dynamics and associated acoustic emissions across relevant spatial and temporal scales remains

challenging. Here, we present an integrated optical–acoustic framework for interrogating ADV that combines ultra-high-speed imaging (up to 10 million frames per second), slow-frame optical microscopy (50 frames per second), time-lapse confocal fluorescence imaging (1 frame per second), and passive cavitation detection. This approach captures ADV from single-droplet events to macroscale bubble clouds and their associated acoustic signatures. We discuss key experimental considerations, including optical and acoustic alignment, synchronization workflows, sample mounting, and acoustic coupling strategies. In addition, we share practical protocols and troubleshooting strategies. This work aims to provide a comprehensive and practical guide for researchers studying ADV.

Contributed Papers

8:20

2aBAa2. High-speed optical dynamics of repeated acoustic vaporization of a single droplet. Sander Spiekhout (Radiology, Univ. of Michigan, Ann Arbor, MI 48109, spiekhou@umich.edu), Bachir A. Abeid, Jon B. Estrada (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI), and Mitra Aliabouzar (Radiology, Univ. of Michigan, Ann Arbor, MI)

High-speed investigations of phase-shift droplet dynamics are often limited to single pulse excitations. However, single pulses are rarely used in *in vitro* and *in vivo* studies of acoustic droplet vaporization (ADV). Here, we study via high-speed optical imaging the response of individual droplets with various perfluorocarbon (PFC) cores to 20 repeated ultrasound excitations. Droplets (diameter: 12 ± 1 μm) composed of perfluoropentane (PFP), perfluorohexane (PFH) or perfluorooctane (PFO), surrounded by a fibrin matrix, were vaporized by a six-cycle, 2.5 MHz pulse with a peak-negative pressure of 6 MPa. Droplet dynamics were captured at 10 Mfps. During the first excitation, droplets exhibited strong growth and collapse events, irrespective of PFC core. From the second excitation onward, PFP and PFH droplets turned into bubbles that oscillated near their Minnaert resonance frequency. In contrast, PFO persisted as droplets and showed a characteristic ADV response across all 20 excitations, while exhibiting progressively larger maximum radii (from 44 ± 4 to 59 ± 5.3 μm) and longer growth-collapse times (7.6 ± 1.3 to 10.3 ± 1 μs). These findings provide insight into how droplet formulations can be tuned to provide either typical bubble responses or repeated vaporization events.

8:35

2aBAa3. Design and construction of a low-cost amplifier for focused ultrasound applications. Joshua Elbertson (Phys., High Point Univ., One University Parkway, High Point, NC 27268, jelberts@highpoint.edu) and Eric Rokni (Phys., High Point Univ., High Point, NC)

Histotripsy is a non-invasive therapy that can fractionate diseased tissue by inducing cavitation through high-pressure ultrasound waves. Histotripsy requires both a high-frequency and a high-voltage setup to generate the power necessary to cause cavitation. However, the high cost (>$15000) of conventional amplifier systems limits accessibility for research and education. In this study, we present a cost-effective radio-frequency (RF) power amplifier designed for focused ultrasound experimentation. Built for under $3000, the amplifier is designed to be a Class D system that uses MOSFET switching to achieve high efficiency while delivering up to 600 V from a DC power supply. We detail the design, construction, and function of each component and validate the performance through measurements of signal integrity and voltage gain. This low-cost, high-power apparatus allows institutions with limited resources to investigate cavitation and histotripsy, expanding opportunities for research into therapeutic ultrasound.

8:50

2aBAa4. Ultrasound-induced dynamics of nanodroplets and microbubbles in synthetic cystic fibrosis mucus: Toward pulmonary drug delivery and mucus rheology assessment. Hieu Doan (Radiology & Biomedical Eng., Univ. of North Carolina at Chapel Hill, 2000 Old Clinic, CB #7510, Chapel Hill, NC 27599, hieu_doan@med.unc.edu), Jyoti Jethe (Radiology & Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Mahmud Raihan (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), Kelly VanTreeck (Pharmaceutical Sci., Biomedical Eng., & Radiology, UNC Chapel Hill, Chapel Hill, NC), Kaymille Sherman (Radiology & Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Jacob A. Mattern (Pharmaceutical Sci. & Biomedical Eng., Univ. of North Carolina, Chapel Hill, NC), Phillip Durham (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Paul A. Dayton, David Hill (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), and Virginie Papadopoulou (Radiology & Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

The viscoelastic properties of airway mucus in cystic fibrosis (CF) limit particle transport, motivating non-invasive approaches to assess mucus rheology. Ultrasound-activated nanodroplets serve as acoustic probes, as their vaporization and cavitation responses depend on the surrounding medium. This work presents an experimental platform enabling concurrent optical and acoustic characterization of nanodroplet behavior in CF-like media. A focused 1.1 MHz transducer was integrated with an off-the-shelf DSLR camera to enable real-time visualization of acoustic droplet vaporization (ADV) and cavitation detection. A custom adapter on a motion stage enabled precise alignment of the acoustic focus with the optical focal plane, addressing depth-of-field limitations from earlier iterations. Lipid-shelled octafluoropropane (OFP) nanodroplets were studied in 1× PBS and 1%, 2%, and 3% polyethylene oxide (PEO) solutions, representing progressively viscoelastic CF-like environments. Vaporization thresholds were quantified using synchronized optical and acoustic measurements. Results show that vaporization thresholds increase systematically with increasing PEO concentration, consistent with elevated viscosity and elastic resistance, while cavitation activity was observed across all media. Ongoing experiments are extending this platform to investigate microbubble responses across ultrasound cycles and pressures. Together, this work contributes to ultrasound-based methods for probing fluid viscoelasticity in CF-like environments.

Invited Papers

9:05

2aBAa5. Development of improved liver steatosis phantoms for accurate histopathologic representation and multimodal imaging. Tara Movaghar, Ali Shariq, Esther Park, Adam Woodward, Nazia Hafeez, Anbarin Khalid, Liqiang Ren, Yue Zhang (Radiology, UTSouthwestern Medical Ctr., Dallas, TX 75390), I. Rosado-Mendez (Medical Phys. and Radiology, Univ. of Wisconsin–Madison, Mexico City, Mexico), David T. Fetzer (Radiology, UTSouthwestern Medical Ctr., Dallas, TX), and Caroline de Gracia Lux (Radiology, UTSouthwestern Medical Ctr., 5323 Harry Hines, Dallas, TX 75390-8514, Caroline.Lux@UTSouthwestern.edu)

Metabolic dysfunction-associated steatotic liver disease is now the leading cause of chronic liver disease. Microvesicular steatosis is specifically associated with progression to cirrhosis, whereas macrovesicular steatosis is typically indolent. Current noninvasive tools for liver fat quantification fail to differentiate these forms of steatosis, limiting patient-tailored management. Quantitative ultrasound

(QUS) is uniquely positioned overcome this limitation. The size and spatial distribution of lipid vacuoles is known to influence acoustic properties and, consequently, QUS-based liver fat measurements. However, no clinical studies or phantom models currently account for lipid microstructure, leaving the clinical potential of QUS unexplored. This work addresses this gap by developing novel, multimodal liver phantoms that replicate hepatic architecture, fat content, and the microstructural characteristics of microvesicular and macrovesicular steatosis. Phantom structure and composition were validated using microscopy and chemical analysis, and imaging performance was evaluated using ultrasound, magnetic resonance imaging, and computed tomography. These phantoms provide a robust platform for evaluating and standardizing QUS tools and for advancing noninvasive characterization of hepatic steatosis, with the potential to improve risk stratification and personalized disease management. [Work supported by the Society for Radiologists in Ultrasound, the Radiological Society of North America and the Siemens Partnership Grant.]

9:25–9:45 Break

Contributed Papers

9:45

2aBAa6. Reproducibility of ultrafast vector Doppler imaging in a murine model under free-hand cardiac flow measurements. Geraldi Wahyulaksana (Radiology, Weill Cornell Medicine, 416 E 55th St., New York, NY 10022, gew4002@med.cornell.edu), Colin K. L. Phoon (Div. of Pediatric Cardiology, NYU Langone Health, New York, NY), Ashmit Pal (New York Univ., New York, NY), Glenn I. Fishman (Leon H. Charney Div. of Cardiology, NYU Grossman School of Medicine, New York, NY), and Jeffrey A. Ketterling (Radiology, Weill Cornell Medicine, New York, NY)

Ultrafast vector Doppler imaging (VDI) enables high–spatiotemporal-resolution characterization of intracardiac blood flow and is sensitive to subtle hemodynamic alterations relevant to early cardiac dysfunction. However, cardiac flow is inherently 3-D, while VDI is typically acquired in 2-D imaging planes; thus, measurement limits arise from 2-D sampling of 3D cardiac flow. Here, we evaluated the reproducibility of ultrafast VDI-based cardiac flow measurements under experimentally and clinically realistic free-hand conditions. Experiments were performed in a murine model comprising three dilated cardiomyopathy (DCM) knockout mice and six control mice. For each mouse, flow was sampled over 8–14 consecutive cardiac cycles and repeated across six independent free-hand acquisitions, enabling separation of beat-to-beat physiological variability from acquisition- and slice-dependent uncertainty. Reproducibility was assessed for representative flow descriptors, including kinetic energy, vector concentration, and vorticity. Across repeated heartbeats within a single acquisition, variability was low to moderate, with coefficients of variation typically $\leq$0.30. Inter-acquisition variability was larger, reflecting sensitivity to imaging plane selection. Importantly, DCM and control mice exhibited comparable intra- and inter-acquisition variability, indicating that measurement reproducibility is largely independent of disease state. These findings characterize uncertainty in murine ultrafast cardiac flow imaging.

10:00

2aBAa7. Characterizing the bias in estimates of the shear wave speed with a simplified viscoelastic model. Nicholas A. Bannon (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824) and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, East Lansing, MI, mcgough@egr.msu.edu)

Estimation of the shear wave speed in soft tissues is often evaluated with cross-correlations. Unfortunately, cross-correlations fail to account for the shear viscosity, which can produce significant errors. To characterize these errors, cross-correlations are performed on simulated shear wave particle velocity waveforms that are generated with a far-field viscoelastic shear wave model for a point source. The viscoelastic shear wave propagation model is obtained from the shear term of an approximate time-domain Green's function [Bercoff *et al*., IEEE Trans. UFFC 51, 1523–1536 (2004)]. To obtain estimates of the shear wave speed, cross-correlations are performed on pairs of waveforms spaced 1.232 mm apart at distances ranging from 0.62 to 20 mm with a temporal sampling rate of 10 kHz. The shear wave speed is estimated for shear modulus values ranging from 1 to 4 kPa with an increment of 1 kPa and shear viscosity values varying from 1 to 5 Pas with an increment of 1 Pas. While cross-correlations provide effective estimates of the shear wave speed for smaller values of the shear viscosity and larger values of the shear modulus, larger input values of the shear viscosity produce significantly higher median percent errors.

10:15

2aBAa8. Quantitative cavitation imaging: Computational considerations and potential pitfalls. Abigail Collins (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7LD, United Kingdom, Botnar Inst. for Musculoskeletal Sci., Old Rd., Headington, Oxford OX3 7LD, United Kingdom), Darcy M. Dunn-Lawless, Michael Gray, and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford OX3 7LD, United Kingdom)

With cavitational therapies continuing to be adopted clinically, the ability to accurately quantify and compare treatments is essential to ensure consistent efficacy and patient safety. Passive Acoustic Mapping (PAM) can reconstruct quantitative maps of cavitational energy density in real time, providing a powerful platform for cavitation dosimetry. However, the quality and accuracy of PAM images can be significantly affected by the measurement setup, physical properties of the monitoring device, and choice of computational parameters. To obtain reliable cavitation doses that can be compared across indications and experiments, compensations for the device, position, and frequency dependent point spread function as well as attenuation and array element sensitivity and directivity must be applied. The selection of beamforming parameters, such as data acquisition length and windowing, voxel size, and array element decimation, can profoundly affect the accuracy of PAM-derived cavitation doses. This work will present a step-by-step guide to mapping cavitational signals, using practical examples of *in vitro*, *ex vivo*, and *in vivo* experimental data covering a range of device geometries and measurement setups to illustrate potential pitfalls. Trade-offs between image quality, accuracy, and computation time will be explored for each application, toward energy preserving, setup independent dosimetry of cavitational treatments.

Session 2aBAb

Biomedical Acoustics and Physical Acoustics: Ultrasound Beamforming and Its Applications I

Jian-yu Lu, Chair
Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606

Chair's Introduction—7:55

Invited Papers

8:00

2aBAb1. Real-time synthetic aperture dual-stage beamforming for volumetric row-column imaging. Jørgen A. Jensen (Dept. of Health Technol., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 349, Rm. 222, Kgs. Lyngby DK-2800, Denmark, jaje@dtu.dk), Lasse T. Jørgensen, and Sebastian K. Præsius (Dept. of Health Technol., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Volumetric imaging carries a very high calculation load when a full volume must be beamformed in real-time. The number of beamforming operations for matrix arrays is the product of the number of emissions, transducer elements, planes, lines in a plane, samples in a line, and the frame rate, resulting in approximately 61 trillion operations per second, which makes it infeasible for modern GPUs. Row-column arrays and dual-stage beamforming can significantly reduce this. The number of receiving elements is often 100 times fewer than those in fully populated arrays, and using dual-stage beamforming enables the beamforming of a single plane, which can then be extrapolated to a volume. This combination results in a calculation reduction of more than 10000 times compared to normal volumetric imaging, without compromising image resolution. A real-time GPU beamformer is presented, based on these two innovations. A 6 MHz Vermon 128 + 128 RCA probe and a Verasonics Vantage 256 scanner were used with an NVIDIA GeForce RTX 4090 GPU. The *in vivo* acquisition utilized 96 defocused emissions at a rate of 12 kHz for a volume acquisition rate of 125 Hz. The resulting beamforming rate for the full volume was 1440 volumes per second, greatly exceeding the real-time rate.

8:20

2aBAb2. Silicon-integrated beamformers for catheter-based ultrasound imaging systems. F. L. Degertekin (Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30332, levent@gatech.edu)

There is an increasing demand for catheter-based ultrasound imaging in minimally invasive interventional medical procedures in terms of resolution and field of view while retaining or reducing the device size. This requires arrays with larger number of elements and integration of transmit-receive beamforming electronics at the catheter tip to manage the number of cable connections to the backend imaging system. In this talk, we will review some of the recent developments in this area with applications in intracardiac echocardiography (ICE) and intravascular ultrasound (IVUS) imaging catheters. Specifically, we will describe Application Specific Integrated Circuits (ASICs) that implement various combinations of microbeamforming (sub-aperture beamforming) and time division multiplexing for receive beamforming to reduce the cable count in ICE catheters operating in the 5–10MHz range. These systems, which are suitable for integration with piezoelectric and micromachined ultrasound transducer arrays, also include fully programmable transmit beamformers. In addition, we will discuss 40MHz CMUT-on-CMOS IVUS imaging systems with plane wave transmit beamforming capability for use on 0.014" diameter guidewires to extend the reach of imaging in coronary arteries. Finally, we will discuss some future implementations where these imaging systems can be integrated with pressure sensors on microcatheters to more effectively guide coronary interventions.

8:40

2aBAb3. Rethinking ultrasound beamforming in the high-performance computing era. Di Xiao (Elec. and Comput. Eng., Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2J 0E2, Canada, di.xiao@uwaterloo.ca), Hassan Nahas, and Alfred Yu (Univ. of Waterloo, Waterloo, ON, Canada)

Delay-and-sum (DAS) beamforming is the predominant image formation algorithm for biomedical ultrasound imaging. In recent years, variations on DAS, such as minimum variance, delay-multiply-and-sum, or coherence-based beamforming, have all shown advantages over naïve DAS. However, DAS remains the most popular beamformer for its simplicity and efficiency. In our work, we combine modern tools, such as GPU programming and machine learning, with the classical DAS model; operationally, we leverage a high-frame-rate ultrasound paradigm for applications in flow and quantitative ultrasound. Moreover, our techniques are typically designed to operate with real-time capabilities on edge computing devices, highlighting the point-of-care advantages of ultrasound as a modality. GPU acceleration is critical for the software-based beamforming pipeline as radiofrequency, and image data processing is performed on consumer-grade computing hardware. We examine and compare acceleration provided by GPU DAS implementations, then use them to realize live vector doppler imaging and speed-of-sound estimation. Parallel computing has also spurred machine learning and AI research, which our lab has integrated with DAS for image formation tasks, such as data inference to improve frame rates and image quality. Overall, our work provides a modernized outlook on DAS while retaining the functionality of the classical formulation.

9:00

2aBAb4. Functional beamforming for medical ultrasound. Robert P. Dougherty (OptiNav, Inc., 1414 127th Pl. NE, #106, Bellevue, WA 98005, rpd@optinav.com)

The frame rate of medical ultrasound has recently been improved by avoiding scanning on transmit in favor of a relatively small number of unfocused excitations for each frame and forming the image numerically by beamforming the received data. Now the frame rate is limited by the beamforming computation time instead of the round-trip acoustic propagation time to all of the grid points in sequence. A potential downside is a reduction of the dynamic range of the image. A popular excitation method uses a sequence of plane waves formed for different angles. Recent research has focused on the application of advanced algorithms to improve the resolution and dynamic range relative to the famously poor results from conventional beamforming. Recent papers have proposed adaptive beamforming, which can be viewed as better for resolution than dynamic range. Functional beamforming is a specialist method for dynamic range. In this paper, the application of functional beamforming to medical ultrasound will be explained. Fortuitously, the formulation from passive aeroacoustic beamforming carries over with few changes. The cross-spectral matrix can be augmented with extra rows and columns to add the special depth resolution characteristic of sounding methods. Results using data from a challenge problem are presented.

9:20

2aBAb5. Ultrasound imaging and nonlinear beamforming with nulls. Michael L. Oelze (Elec. and Comput. Eng., Univ. of Illinois at Urbana–Champaign, 405 N Mathews, Urbana, IL 61801, oelze@illinois.edu), Rita Miller, Mick Gardner, and Zhengchang Kou (Elec. and Comput. Eng., Univ. of Illinois at Urbana–Champaign, Urbana, IL)

Recently, we introduced a nonlinear beamformer called null subtraction imaging (NSI). Currently, for most applications, (apodization. Images are formed by averaging the receive envelopes created with the second two apodizations and then subtracting the receive envelopes created using the first apodization, resulting in the inversion of the null acting as the mainlobe beam. The resulting images have three important properties: (1) the DC offset can be adjusted to narrow the NSI beam to super-resolve structures, (2) grating lobes can be reduced or mitigated, and (3) speckle is no longer fully developed. Therefore, for specific tasks, NSI can provide image enhancements. In this talk, we will discuss the ability of NSI to produce super-resolved microvessel images, super-resolved color flow images, improvements to B-mode images through grating lobe reduction, and use of NSI to improve image quality for wide field-of-view arrays with large pitch. (3) the flip of the second (1) a zero mean apodization that results in a null in the beam pattern at 0 deg, (2) the zero mean apodization with a slight DC offset value, and (3) the flip of the 2nd apodization. Images are formed by averaging the receive envelopes created with the second two apodizations and then subtracting the receive envelopes created using the first apodization, resulting in the inversion of the null acting as the mainlobe beam. The resulting images have three important properties: (1) the DC offset can be adjusted to narrow the NSI beam to super-resolve structures, (2) grating lobes can be reduced or mitigated, and (3) speckle is no longer fully developed. Therefore, for specific tasks, NSI can provide image enhancements. In this talk we will discuss the ability of NSI to produce super-resolved microvessel images, super-resolved color flow images, improvements to B-mode images through grating lobe reduction, and use of NSI to improve image quality for wide field-of-view arrays with large pitch.

9:40

2aBAb6. Stroboscopic effect elastography ultrasound: A novel elasticity imaging technique for point-of-care ultrasound devices. Sara Gomez-Ramirez (Phys., Univ. of Rochester, Rochester, NY 14618), Gilmer Flores (Biomedical Eng., Univ. of Rochester, Rochester, NY), and Benjamin Castaneda (Biomedical Eng., Univ. of Rochester, 204 Robert B. Goergen Hall, Rochester, NY, bcastane@ur.rochester.edu)

We present Stroboscopic Effect Elastography Ultrasound (SEE-U), a novel elasticity imaging technique that relies on conventional ultrasound beamforming capabilities and exploits the stroboscopic effect for visualizing shear-wave propagation without the need for ultrafast beamforming. In SEE-U, the tissue is subjected to a harmonic external mechanical excitation at frequency f_0, while ultrasound images are acquired at a pulse repetition frequency, that is, slightly detuned, $fs = f_0 - \Delta f$. This intentional detuning converts the high-, thereby, enabling time-resolved characterization of the shear-wave phase using conventional ultrasound frame rates. The complex Fourier coefficient at f_0 into an apparent low-frequency motion at the beat frequency Δf, thereby enabling time-resolved characterization of the shear-wave phase using conventional ultrasound frame rates. The complex Fourier coefficient at Δf yields the local displacement amplitude and phase, from which the spatial wavenumber k and the corresponding shear-wave speed $c_s = 2\pi f_0/k$ are subsequently estimated. This imaging methodology can provide quantitative elastography estimates using the current hardware architecture in point-of-care ultrasound commercially available devices, which typically implement traditional beamforming strategies. We will present experimental results in tissue-mimicking phantoms and *exvivo* bovine liver, directly comparing SEE-U against state-of-the-art elasticity techniques that rely on external vibration, such as reverberant shear wave elastography.

10:00–10:20 Break

10:20

2aBAb7. Comparative analysis and development of aberration correction techniques for transcranial ultrasound. Zixuan Tian (Elec. and Comput. Eng., Univ. of Illinois, Urbana Champaign, Urbana, IL 61801), Yun Jing (Acoust., Penn State Univ., State College, PA), and Aiguo Han (Dept. of Biomedical Eng., Virginia Polytechnic Inst. and State Univ., 325 Stanger St., Blacksburg, VA, aiguohan@vt.edu)

The skull poses significant challenges in transcranial ultrasound imaging and therapy. As ultrasonic waves propagate through the skull, wavefront distortions, referred to as aberrations, arise due to the skull's heterogeneous structure and large acoustic contrast relative to brain tissue. These aberrations substantially degrade image quality and reduce wave-focusing effectiveness and targeting accuracy in transcranial ultrasound applications. Aberration correction techniques can be used to mitigate these effects, particularly when the skull speed of sound distribution is known, for example, from X-ray computed tomography or ultrasound-based estimation methods. However, a tradeoff exists between correction effectiveness and computational efficiency, depending on the complexity of the underlying

model. To elucidate the impact of model complexity, we performed a comparative simulation-based analysis of several aberration correction techniques, covering different levels of model complexity. We demonstrate that full waveform correction is more effective than phase-only correction, and that phase correction based on full-wave-derived time delays is more effective than phase correction based on eikonal-equation-derived time delays, albeit at increased computational cost. Motivated by these findings, we propose a novel strategy to enhance the effectiveness of eikonal-based phase correction while preserving its computational efficiency, with broad implications for transcranial ultrasound applications. [Work supported by NIH R21EB032638.]

10:40

2aBAb8. Deep learning-based beamforming from sub-sampled acquisition with sparsity in both transmit/receive channel and Steering Angle. Zhiyu Sheng (Medicine, Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 624, Pittsburgh, PA 15213, zhs41@pitt.edu), Jiayi Wang, and Kang Kim (Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA)

Ultrasound data acquisition and image reconstruction have extensive complexity in hardware and software regarding acquiring, sampling, storing and computing a big amount of channel data. These make it difficult to realize 3-D or wearable ultrasound imaging systems without compromising image quality, field-of-view (FOV), frame (or volume) rate, *etc*., compared to the state-of-the-art cart-based ultrasound clinical or research platforms. This study used a deep learning-based approach with a novel channel data recovery framework to reconstruct full planewave-compounded RF data from sub-sampled acquisition with sparsity in both transmit/receive channel and steering angle. The model was trained and validated on RF data from 4000 images acquired from five participants at different body imaging sites. The performance of the approach was evaluated under each sub-sampling conditions (from low to high sparsity) compared to full sampling ground truth using metrics, including global peak signal-to-noise ratio (PSNR), global structural similarity (SSIM), and SSIM maps, derived via moving windows to assess recovery performance on local structures. The encouraging results indicate the capability of not only scaling down the bulky channel data, but also greatly reducing the complexity in hardware, and thus paving the way for developing wearable or 3-D ultrasound imaging systems without compromising image quality.

11:00

2aBAb9. Ultrasonic beamforming for non-destructive evaluation. Anthony Croxford (Univ. of Bristol, Bristol, United Kingdom, A.J.Croxford@bristol.ac.uk), Paul Wilcox (Univ. of Bristol, Bristol, United Kingdom), and Sergio Cantero-Chinchilla (School of Elec., Electron., and Mech. Eng., Univ. of Bristol, Bristol, United Kingdom)

Ultrasonic non-destructive evaluation had some unique properties that affect the options available for beamforming. Chief among these is the fact that targets are typically stationary and will have multiple large reflectors around the geometry. This leads to the ability to use beamforming solutions that are not possible in many other disciplines. In addition, there are multiple available methods to generate sound in components, from classically employed piezoelectric devices to laser-based systems. Each of these have distinct directivity profiles that alter the appropriate beamforming and imaging approaches. This paper will step through the state-of-the-art for imaging in NDE systems, showing how the unique nature of the problem can be exploited and optimized for a range of systems and problems, from improving the performance of repeat measurements to using multiple image views to inspect parts and how different inspection goals can alter the overall beamforming and array design approach.

11:20

2aBAb10. Sparsity-based ultrasonic guided-wave imaging of structural defects. Fauzia Ahmad (Temple Univ., 1947 N. 12th St., Philadelphia, PA 19122, tug80470@temple.edu)

We consider the problem of defect localization in thin-walled structures, such as plates and pipes, using ultrasonic guided waves. We discuss signal processing solutions, which rely on a sparse array of ultrasonic transducers in pitch-catch mode for generation and reception of guided waves and exploit the sparse nature of defects in the structure being monitored. Propagation models are developed to account for the multimodal nature of guided waves and presence of multiple propagation paths between each transmitter and receiver. This enables the construction of model-based dictionary matrices relating the signals arriving at the receivers to the locations of potential defects in the region of interest (ROI). Reconstruction of the ROI is achieved by inverting the resulting linear signal model under a group sparsity constraint, where the groups extend across the various transducer pairs and the different modes/paths. Experimental validations are provided for defects on an aluminum pipe and a metal plate.

11:40

2aBAb11. Ultrasonic guided-wave beamforming for the characterisation of defects in pipes. Michael J. Lowe (Dept. of Mech. Eng., Imperial College London, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

Ultrasound Guided-Wave Testing (GWT) is well established as a Non-Destructive Evaluation (NDE) method for pipelines. A rich research history was followed by industry take-up 25 years ago, with the capability delivered in products and international standards. In routine application, GWT is used to detect the presence of corrosion in pipes by looking for reflection of the guided waves from the corrosion. The guided waves travel in the wall of the pipe and are reflected due to the change of acoustic impedance caused by the corrosion thinning of the pipe. In addition to finding the location of corrosion by measuring the time travel of the reflected guided wave, it is also possible to perform some characterization of the size of the corrosion patch, by beamforming array imaging. The transducer for the guided waves is a multi-element device encircling the pipe, a form, which is already necessary in order to control the selected guided waves. The array imaging can then be possible by processing the signals received at the multiple elements. The talk will summarize the state-of-the-art for this NDE, including the specific challenges of performing the imaging around the pipe using guided-wave fields.

Session 2aCA

Computational Acoustics, Acoustical Oceanography, Physical Acoustics, and Underwater Acoustics: 3D Acoustics

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

Alexander S. Douglass, Cochair
Seattle, WA

Chair's Introduction—9:00

Invited Papers

9:05

2aCA1. Ray-based prediction of out-of-plane reflections from seamount topography. Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, thomas.jerome@arlut.utexas.edu), Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Julien Bonnel (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA), James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

During the 2023 New England Seamounts Acoustics Experiment (NESMA), multiple observations of out-of-plane reflections were identified from recordings of impulsive sound sources on a set of three near-bottom recorders. Analysis of arrival times in combination with 2-D and 3-D ray traces revealed that many observations could be attributed to interactions with specific topographic features on the slope of the nearby Atlantis II seamount. However, forward modeling of observations using ray traces yielded limited success in reproducing observations. It was hypothesized that the complex topography presents a significant challenge for prediction of out-of-plane arrivals due to practical limitations on topographic resolution, the influence of non-geometric effects, and model sensitivity. This presentation extends previous analysis by tracing rays from known source and receiver locations to search the topography for features that are likely to produce specular reflections. In combination with calculations of the Fresnel zone for predicted reflections, results are used to examine the scale and topographic character of predicted reflectors. Comparison of modeled arrival structure with recordings of impulsive sound sources collected during NESMA 2024 are used to further evaluate limitations of the method when applied to the complex topography typical of seamounts. [Work supported by ONR.]

9:25

2aCA2. 3-D boundary fitted parabolic-equation model and its application on simulating seafloor acoustic scattering. Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093, ytlin@ucsd.edu), William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Chad M. Smith (Graduate Program in Acoust., The Penn State Univ., State College, PA)

A 3-D parabolic-equation (PE) method with a boundary fitted grid is utilized for simulating underwater sound waves scattering from rugged seafloor. Specifically, a higher-order Split Step Padé algorithm with cross terms of spatial partial derivatives is adapted to approximate the square-root PE operator with the method of implicit alternating directions. Besides, a Galerkin method with non-uniform grid intervals is used to discretize the boundary fitted model mesh. The advantage of the boundary fitted grid is to improve the PE solution accuracy when handling acoustic wave scattering from irregular boundaries. The PE model will first be demonstrated by solving a 3-D wedge problem and then simulating single bottom bounce reflection and scattering off rugged seafloor observed in the area of New England Seamount Chain. To confirm numerical computation performance, the 3D model results will be compared with acoustic beamforming data from hydrophone arrays deployed at the edge of the seamount slope. [Work supported by the ONR Task Force Ocean program.]

9:45

2aCA3. Enhanced tunneling through curved nonlinear internal waves in a 3-D normal mode feature model. Matthew A. Milone (Scripps Inst. of Oceanogr., Univ. of California, 8622 Kennel Way, San Diego, CA 92093, mamilone@ucsd.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Nonlinear internal waves in shallow water have significant acoustic impacts, such as energy trapping in ducts between wavefronts for distances of tens of kilometers or more. A normal mode approach applied to a 3-D feature model in cylindrical coordinates is useful in studying effects of wavefront curvature. Earlier work simplified computations using an infinite-width front "well approximation," which accurately represented in-duct propagation, but neglected details of the acoustic field in exterior regions. Upon introducing finite-width wavefronts, additional propagation effects are observed, most notably an enhanced form of tunneling carried by horizontal companion modes. These modes are named for their relationship to horizontal trapped modes, which share similar phase patterns but exhibit greatest modal excitation in different regions. Characteristics of companion modes are presented, along with two necessary criteria for their existence: wavefronts must be both curved, and finite in width. A mode number conservation formula is used to generate curves for wavenumber dependence on radius of front curvature, illustrating why companion modes are not present in straight waveguides. Finally, comparisons are made between the square wave approximation used in this idealized model, and the more realistic wave profiles, which arise as solutions of the KdV equation.

10:00

2aCA4. Toward fast scattering solvers using iterative path expansions. Matthias Ospel (French-German Res. Inst. of Saint-Louis, 5 Rue du General Cassagnou, Saint-Louis 68300, France, matthias.ospel@isl.eu)

Numerical methods based on edge source diffraction integrals offer a promising route to fast acoustic scattering solvers with favorable high-frequency scaling, requiring discretization only along edges. In this work, we present an iterative solver for inhomogeneous Helmholtz problems with Neumann boundary conditions in which the solution is constructed as a sum of physically interpretable propagation paths, including incident, reflected, and edge diffracted wavefield contributions. A formal connection is established between an existing diffraction operator formulation and path-tracing approaches that have often been motivated heuristically. High-order diffractions are generated by successive path expansions, yielding a matrix-free solver that adaptively explores dominant contributions to the wavefield. Numerical results demonstrate fast convergence to reference solutions computed via direct boundary element methods, along with strong potential for parallelization.

10:15–10:30 Break

Invited Paper

10:30

2aCA5. Simulating acoustic modem performance using 3-D beam tracing in complex harbor environments. Michael B. Porter (Heat, Light, and Sound Res., Inc., 12625 High Bluff Dr. Ste. 211, San Diego, CA 92130-2054, Porter@HLSResearch.com), John C. Peterson (Heat, Light, and Sound Res., Inc., La Jolla, CA), Jennifer J. Johnson, Sandipa Singh, Timothy F. Duda, and Peter Traykovski (Woods Hole Oceanogr. Inst., Woods Hole, MA)

Ray and beam tracing methods are often appealing for handling 3-D problems in ocean acoustics. In contrast to other methods, they easily handle broadband propagation, e.g., of full waveforms—the process uses a simple sum and delay of copies of the source waveform based on the amplitudes and delays of the echoes in the channel. This is particularly attractive for modeling acoustic modem performance. In addition, they readily handle dramatic 3D effects, such as those caused by boundary reflections off harbor features, for instance, wharves, jetties, and quays. We discuss one such model in terms of its implementation and its application to studying modem performance in complex harbor areas. The model is applied to data collected in the Weymouth Fore River area, as well as the Taunton River in Massachusetts.

Contributed Paper

10:50

2aCA6. A database approach to identifying factors influencing acoustic modem performance. John C. Peterson (Heat, Light, and Sound Res., Inc., La Jolla, CA 92037-4501, jcp@jcp-consulting.com), Michael B. Porter (HLS Res., Inc., San Diego, CA), Sandipa Singh, Jennifer J. Johnson, Peter Traykovski, and Timothy F. Duda (Woods Hole Oceanogr. Inst., Woods Hole, MA)

The decoding performance of acoustic modems depends on a multitude of factors. Those factors that are strongly correlated with degraded performance will generally vary from one acoustic environment to another, over both space and time. A generalized process is presented for constructing a database that incorporates a wide range of potentially relevant factors in a temporally consistent manner, including but not limited to; decoding performance statistics from the acoustic modem of interest, oceanographic data, such as wind speed and tidal phase, and propagation metrics associated with the acoustic path, such as transmission loss and channel time spread. Propagation metrics derived from processed measurements, and numerical modeling using tools, such as Bellhop3D, are addressed. While there are many approaches to analyzing the database to identify factors that are highly correlated with degraded modem performance, we found significant value in techniques from interactive data analysis. Despite their simplicity, data brushing and linking methods were found to be particularly useful. These methods are easy to deploy using a number of data visualization software packages, and the results are fast and easy to interpret without special training in advanced statistics.

Invited Paper

11:05

2aCA7. 3-D sonar image reconstruction for detecting and classifying unexploded ordnance. Thomas E. Blanford (Univ. of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu), Albert Reed (Sandia National Labs., Albuquerque, NM), Suren Jayasuriya (Arizona State Univ., Tempe, AZ), Daniel C. Brown, and Shawn Johnson (Penn State Univ., State College, PA)

Low-frequency, downward-looking synthetic aperture sonar (SAS) can be used to detect and localize both surficial and buried unexploded ordnance (UXO) in very shallow water environments. Using a 2-D pulse-echo array deployed from a moving platform, the recorded echoes are reconstructed into 3-D imagery. The reconstruction process, however, is complicated by several factors. Acoustic propagation is anisotropic when imaging into the sediment volume, the array aperture is usually sparsely sampled in at least one dimension, and the sensor resolution is different in all three dimensions. Additionally, the burial depth of UXO and its pose relative to the sensor are unknown but affect the appearance of an object in the image. This presentation will describe recent investigations into two different image reconstruction algorithms to improve the quality of 3-D SAS imagery for the detection and classification of UXO. The first approach exploits spatial coherence information to improve the contrast of objects located near the sediment-water interface. The second approach uses iterative deconvolution and an analysis by synthesis reconstruction using implicit neural representations. The results will quantitatively and qualitatively demonstrate the benefits and drawbacks for reconstructing high-resolution, 3-D acoustic imagery with these techniques.

Contributed Papers

11:25

2aCA8. Target echo strength calculation for a moving source using time-dependent physical optics. Asaf Kor (Mech. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel), Amir Boag (Elec. and Comput. Eng., Tel Aviv Univ., Tel Aviv, Israel), and Lea Beilkin (Mech. Eng., Tel Aviv Univ., Tel Aviv, Israel, leabeilkin@tauex.tau.ac.il)

We examine the problem of calculating the echo strength of a stationary target in response to a moving source in 3-D space. The target is assumed to be sufficiently smooth and large compared to the wavelength, allowing us to apply the physical optics approximation to model the wave scattering. Due to the motion of the source, we do not use the more conventional frequency-domain physical optics method. Instead, we develop a time-domain variant of this approach, explicitly accounting for the source motion. Our study focuses on the near-field regime, where source motion induces a complex wave field beyond a simple Doppler effect. The proposed algorithm accommodates arbitrary source maneuvers, high source velocities relative to the speed of sound, and emission by the source of arbitrary shaped signals.

11:40

2aCA9. High-performance-computing models of 3-D wave propagation through cetacean anatomy. Carlos García A. (Dept. of Geosciences, Univ. of Padua, Padua, Italy), Aida Hejazi-Nooghabi (Geophys., Hamburg Univ., Hamburg, Germany), Andrea Colombi (ZHAW—Zürich Univ. of Appl. Sci., Zürich, Switzerland), Steffen De Vreese (Universitat Politécnica de Catalunya, Barcelona, Spain), Jean-Marie Graïc (Dipartimento di Biomedicina Comparata e Alimentazione—BCA, Univ. of Padua, Padua, Italy), and Lapo Boschi (Dept. of Geosciences, Univ. of Padua, via Gradenigo 6, Geosciences Dept., Padova 35131, Italy, lapo.boschi@unipd.it)

Cetaceans rely on sound for vital tasks, including communication, environmental sensing, and prey detection. Although several hypotheses address the acoustic pathways involved in the overall cetacean hearing process, the mechanisms by which sound and vibrations propagate through diverse tissues, particularly for echolocation, remain poorly understood. Addressing such issues might be key to understanding how cetaceans achieve extremely high precision in echolocation, outperforming the sound localization skills of most terrestrial mammals. We explore the potential of using SPECFEM3D, an open-source spectral element method solver, to model 3-D sound propagation through a CT-scan-based model of the head of a bottlenose dolphin (*Tursiops truncatus*). SPECFEM3D provides highly efficient time-domain simulations, essential for capturing complex wave interactions in biological tissues. Optimized for parallel supercomputing, it supports large-scale simulations in non-homogeneous media, enabling analysis of wave propagation through varied tissues. The spectral element method is applied here to high-frequency biophysical modeling of marine mammals for the first time. Despite some significant challenges, notably in generating hexahedral meshes for intricate biological structures, we submit that the spectral element method provides a highly efficient framework to study the mechanics of audition in biological media.

Session 2aMU

Musical Acoustics: Cross-Cultural Musical Instruments and Instruments Across Culture

Jonas Braasch, Cochair
School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180

May Pik Yu Chan, Cochair
Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing University of Pennsylvania, Philadelphia, PA 19104-6228

Chirag Gokani, Cochair
Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

9:00

2aMU1. The strange survival of the serpent. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Peter Guthrie Tait Rd., Edinburgh EH9 3FD, United Kingdom)

The serpent is a Western European musical wind instrument developed in the 16th century to support the singing of plainchant in churches. It consists of a conical tube around two meters long, with a sinuous shape from which it takes its name. It is sounded by buzzing the lips against a cup-shaped mouthpiece. Two widely separated groups of three toneholes are positioned so that they can be covered by the three main fingers of each of the player's hands. The relatively small diameters of the toneholes and the highly irregular spacing of the tonehole lattice could be considered as acoustic imperfections, resulting in an instrument on which accurate intonation is largely dependent on the lip control of the player. However, the serpent survived in church use for several centuries, became an important member of military bands in the late 18th century, was included in symphonic and operatic scores by major composers in the 19th century, and in the last half century has found a new life in jazz, film music and avant-garde compositions. This talk examines the relationship between the acoustics of the serpent and the musical qualities that have seduced successive generations of players and composers.

9:20

2aMU2. Analysis of vocal performance techniques in modern *Biwa* music using pitch analysis. Mizuki Somura (Kyushu Univ., Faculty of Design, Fukuoka, Fukuoka 815-8540, Japan, somura@design.kyushu-u.ac.jp) and Tokihiko Kaburagi (Kyushu Univ., Fukuoka, Fukuoka, Japan)

Satsumabiwa and *chikuzenbiwa* are genres of traditional Japanese narrative music in which vocal recitation is accompanied by the *biwa*, a short-necked lute. These genres developed in Japan after the late nineteenth century and are collectively referred to as modern *biwa* music (*kindaibiwa*). Previous studies of modern *biwa* music have primarily relied on musical analyses of audio recordings subjectively transcribed into Western staff notation. Because notational systems differ across schools and existing notations do not specify detailed melodic content, comparative analysis using staff notation has revealed musical characteristics of different schools and changes in performers individual styles (Somura, 2022). However, staff notation cannot accurately represent pitch and duration, and transcription inevitably involves subjectivity. Consequently, more objective analysis methods have been sought. This paper presents part of an ongoing study on the musical analysis of vocal recitation in modern *biwa* music based on computational pitch analysis. As a case study, results concerning specific vocal performance techniques are reported. This pitch-based approach has the potential to be applied to other genres of Japanese traditional music that do not rely on Western staff notation. [Work supported by FY2025 Interdisciplinary Program of the Humanities and Social Sciences, Kyushu University.]

9:40

2aMU3. Topological data analysis of Korean traditional music and AI-composition. Jae-Hun Jung (Mathematics, POSTECH, Rm. 325 Mathematical Sci. Bldg., Pohang 37673, Korea, jung153@postech.ac.kr)

Compared to Western music, Korean traditional music lacks a unified theoretical framework, and compositional analysis is often repertoire-specific. To investigate the compositional principles of Korean traditional music, we propose a topological data analysis approach of Korean traditional music based on persistent homology. Music is first transformed into music networks, modeled as directed or undirected graphs. We then define appropriate distance functions (metrics or pseudometrics) on these networks and compute multiscale topological features using simplicial complex constructions such as the Vietoris–Rips complex. This analysis reveals several notable findings. Although Korean traditional music is intrinsically monophonic, we observe implicit harmonic structures through an overlap

matrix that captures how cycles in the network are interconnected over time. We also identify an injection relationship in 1-D homology across different path-representable distance definitions. Topological analysis is particularly useful for AI-based composition of Korean traditional music, where data availability is limited, and conventional data-driven learning is challenging. Instead of relying on large training datasets, we develop a topology-preserving AI composition framework that mimics the underlying topological structure of the music. In this talk, we present the proposed methods, computational implementation, and related challenges.

10:00

2aMU4. Studying vocal-tract adaptations for different tone generators on a soprano saxophone body using ultrasound imaging of the tongue. Matthias Heyne (Dept. of Commun. Disord., State Univ. of New York at New Paltz, One Hawk Dr., New Paltz, NY 12561, heynem@newpaltz.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Studies of wind-instrument performance using various imaging techniques have documented vocal-tract adjustments in saxophone, brass, and didgeridoo playing (e.g., Lemoine, 2016; Heyne *et al*., 2019; and Zolotas & Bird, 2010), yet comparing findings across cultures/instrument types remains challenging because resonator geometries and performance approaches vary widely. We examined vocal-tract behavior in soprano saxophone performance while systematically exchanging tone generators from different cultural musical traditions. Building on the second author's hyperspecialization for saxophone technique, we mounted a cornetto mouthpiece, adapted bassoon and Chinese bawu reeds on the saxophone body, and also used the latter as rim flute—alongside the standard saxophone mouthpiece. We recorded midsagittal and coronal tongue ultrasound images with synchronized lateral microcamera and audio signals using a Telemed ultrasound machine and ALPHUS headset (Chen *et al*., 2024). Mouthpieces that preserved an open/closed impedance configuration—such as the cornetto—elicited tongue-position patterns similar to those of the traditional saxophone setup. In contrast, playing the saxophone as a rim flute, which creates an open/open configuration, produced markedly different tongue strategies for impedance matching the conical resonator. These findings highlight how different tone-generators can require different vocal-tract behaviors. The research illuminates the acoustic negotiations required when using different tone generators.

10:20

2aMU5. Variability of phonological contrast realization: The case of Japanese voicing contrast in speech and singing. Rina Furusawa (Linguist, Yale Univ., 37 Hillhouse, New Haven, CT 06511, rina.furusawa@yale.edu)

This study examines tacit knowledge of multi-layered phonological contrasts by investigating the stability of phonetic cues to voicing contrasts across speech and singing. Phonetic cues to voiceless stops, such as voiceless intervals and f0 perturbation, may hinder the faithful realization of melody in singing. We test two hypotheses. One posits that speakers use the same voicing cues in speech and singing. The other posits that singers adjust contrast realization to accommodate melodic constraints. The latter adaptive hypothesis predicts reduced voiceless intervals and f0 perturbation in singing, along with compensatory strategies in other dimensions. An experiment involved 11 Japanese speakers producing 24 four-mora Japanese names with either /ta/ or /da/ in both speech and singing. The results support the adaptive hypothesis. In singing, /t/ showed shorter positive VOT, reduced f0 perturbation, and more voicing during closure. These changes move in the direction of reducing the voiceless-voiced contrast. Also, /d/ had more negative VOT in singing, enhancing the /t/–/d/ contrast in VOT. This lengthening may compensate for the reduction of contrast in other phonetic dimensions. These findings suggest that speakers actively control the realization of voicing-related cues across modalities, bearing implications for theorizations of the phonetics–phonology interface.

10:40–10:50 Break

10:50

2aMU6. Musical instruments feature prominently in prehistoric archaeology and legends of multiple cultures. Steven J. Waller (Rock Art Acoust., 1952 Sonoma Ln., Lemon Grove, CA 91945, wallersj@yahoo.com)

Acoustic field investigations of sound reflections at archaeological sites support the theory that beliefs in echo spirits and supernatural thunderous reverberation motivated our ancestors to create prehistoric canyon petroglyphs and cave paintings. Depictions of music, such as flute players, drummers, and dancers, are quite common visual elements in rock art placed in sound reflective locations around the world. In addition, ancient flutes have been found in resonating cave locations. The name "Pipers' Stones" is a generic term for many megalithic stone circles in the region of the British Isles and Western Europe. As demonstrated by blindfolded auditory scene analysis experiments, wind instruments, such as two bagpipes, can cause sound wave interference patterns in which the nodes of destructive interference can be perceived as acoustic shadows of a ring of gigantic stones blocking the sound. This mysterious auditory illusion could have tricked the ears of our ancestors and served as a blueprint for Stonehenge. Ancient myths from various parts of the world emphasize the magical qualities of the flute, particularly its fertility associations. These examples of musical instruments featuring prominently in prehistoric archaeology and legends of multiple cultures emphasize the importance of including psychoacoustics in studies of archaeological sites.

11:10

2aMU7. Acoustical and musical factors shaping steelpan and handpan performance practices. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

Cross-cultural adaptations of steelpans have occurred worldwide, covering many musical styles and traditions, along with the adaptation of the handpan and related instruments. The steelpan originated on the island of Trinidad in the middle of the 20th century. Since its development, the steelpan family has spread all over the world and across multiple musical genres. The steelpan can be played as a solo instrument, but ensembles of steelpans are common and can range from duets to steelbands comprised of dozens of pannists performing together. In recent decades, another musical instrument made of steel, the handpan, has grown in popularity. This instrument, although related to the steelpan, is notably different in its musical usage. While the handpan can be heard around the globe, the musical genres typically performed on it are not as varied as those performed on the steelpan. Similarly, while the handpan can be played in an

ensemble, it is more typically performed by a soloist. There are acoustical and musical features of these instruments that contribute to their distinctly different usages.

Contributed Papers

11:30

2aMU8. Modal-based optimization of sound post placement for uniform soundboard response in the Santour. Alireza Fazel (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd., Ann Arbor, MI 48105, arfazel@umich.edu) and John Granzow (Performing Arts Technol., Univ. of Michigan, Ann Arbor, MI)

The Santour is a traditional Persian hammered string instrument whose sound is produced by a thin wooden soundboard excited through movable bridges. Beneath the soundboard, five internal wooden sound posts support string loads and connect the board to the resonant cavity. While primarily structural, these posts influence vibrational energy distribution, tonal balance, and response consistency. Despite their acoustic importance, no quantitative framework currently exists for optimizing sound post placement. This paper presents a modal-based computational framework to improve vibrational uniformity through parametric placement. The soundboard is modeled as a thin orthotropic wooden plate using a finite-element shell formulation in Karamba3D within the Rhinoceros 3D environment, with fixed perimeter boundaries and five discrete supports. A modal-averaged bridge-response metric is derived from normalized out-of-plane displacements across the first 10 natural modes. An evolutionary optimization algorithm, implemented via Galapagos, is utilized to minimize response dispersion across the edge and center bridge sets. The optimized configuration achieved standard deviations of 0.037 for edge bridges and 0.057 for center bridges, compared to 0.124 and 0.074 in a conventional Santour. These results represent a two- to three-fold reduction in modal imbalance, providing a substantial improvement in acoustic uniformity.

11:45

2aMU9. The prepared piano as a modulator. Divyamaan Sahoo (Graduate Program in Acoust., College of Eng., The Penn State Univ., 14 Eng. Collaborative Res. and Education Bldg., 556 White Course Dr., University Park, PA 16802, divyamaansahoo@gmail.com)

Inspired by Henry Cowell's experiments, John Cage established a technique to "prepare" a piano by carefully mounting small objects on the strings of the piano to change its sound. The timbral palette of the instrument can be greatly expanded, conjuring sounds of a gamelan ensemble and various idiophones that challenge ethnomusicological description. The method of "preparation" describes a mass-loaded boundary condition on piano string vibrations. When loading piano strings with an object, the sound, we hear when the string is played appears to be a modulation of vibrations of the string and the object. Since piano string vibrations are well known and have been studied analytically, the vibrations of the prepared object could be inferred by demodulation, according to this hypothesis. In effect, the prepared piano functions as a modulator, coding object resonances into the audible spectrum. This talk describes the table of preparations in 23 of Cage's prepared piano compositions, presenting spectral measurements and cross-cultural comparisons, and investigates the modulator hypothesis of the prepared piano as a means to detect faults in the prepared objects by listening to their sound.

Session 2aNS

Noise: Community Noise Testing and Analysis I

Alexandra Loubeau, Cochair
NASA Langley Res. Ctr., MS 463, Hampton, VA 23681

Kieren McCord, Cochair
Pacific Northwest National Lab., 902 Battelle Blvd., Richland, WA 99354

Chair's Introduction—9:00

Invited Papers

9:05

2aNS1. Planning framework for environmental impact assessment and management of a Tactical Unmanned Aircraft System. Ching-Wen Kuo (Odyssey Systems Consulting Group, Ltd., 2610 7th St., WP AFB, OH 45433, cxk425@gmail.com), Frank S. Mobley (711 Human Performance Wing, Air Force Reseach Lab., Dayton, OH), and Alan T. Wall (711 Human Performance Wing, Air Force Reseach Lab., Wright Patterson Airforce Base, OH)

In response to the technology advancement on Unmanned Aircraft Systems (UAS), the Federal Aviation Administration (FAA) has established seven dedicated test sites since 2012 to facilitate their research and integration into the National Airspace System. As UAS technology further evolves to support urban mobility, logistics, and surveillance, the FAA has implemented programmatic environmental noise assessments for drone package delivery operations in the United States. Concurrently, the strategic importance of tactical UAS in national security continues to drive a rapid expansion of their global application. This work seeks to establish a comprehensive framework for environmental impact assessment and management of a tactical UAS. The initial effort focuses on defining a noise measurement protocol to facilitate the creation of a robust acoustic database, classified by vehicle weight, dimensions, and other performance characteristics. In addition, this work expects to evaluate and demonstrate representative metric(s) to ensure rigorous cross-comparison and verification. Ultimately, this effort aims to create an empirical sound source database for integration into an acoustic model, enabling high-fidelity verification and validation of environmental noise impact.

9:25

2aNS2. Correlating high-altitude wind conditions with rocket ascent sonic boom characteristics in Southern California communities. Jason D. Bickmore (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, jdb387@byu.edu), Mark C. Anderson, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA), Luca J. Westfall (Dept. of Geography, Brigham Young Univ., Provo, UT), Benjamin G. Allen, and Brooke I. Hunsaker (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Southeastern trajectories of rockets, launched from Vandenberg Space Force Base (VSFB), cause sonic booms generated during ascent to reach communities in Santa Barbara and Ventura Counties, California. Resident complaints motivated an effort to both measure and model the sonic booms. Because PCBoom-based models suggest that a significant shift in boom focal region along the coast can be caused by seasonal changes in stratospheric winds, this paper examines measurements of launches made at different times of year. First, a data-driven strategy for categorizing pre-focus, focus, and post-focus boom waveforms is discussed. Second, day-of-launch weather is correlated with boom metrics and waveform characteristics. These analyses could be useful in helping inform residents of the likelihood of sonic booms based on current weather conditions. [Work sponsored by SLD-30 at VSFB.]

9:45

2aNS3. Introduction to QUIETBASE: Quantitative investigation and evaluation tool for base acoustics and surrounding environments. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu), Kieren McCord (Pacific Northwest National Lab., Richland, WA), Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA), Alan T. Wall (Air Force Res. Lab., Wright Patterson Airforce Base, OH), Tracy L. Fuentes (Pacific Northwest National Lab., Seattle, WA), Arthur N. Popper (College Park, MD), and Makayle S. Kellison (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Within Department of Defense installations, the high density of threatened and endangered species and the proximity of communities to noise-generating activities necessitate frequent and costly noise impact assessments under the National Environmental Policy Act (NEPA). As mission activity increases, repeated NEPA assessments can slow operational tempo and mission readiness. Although various tools can model noise from aircraft, blasts, and sonic booms, current approaches rely on nonstandard validation practices and historical metrics, such as maximum A-weighted level or day–night average sound level, that may not be well correlated with community or

environmental response to high-intensity, low-frequency noise sources. This paper introduces QUIETBASE (Quantitative Investigation and Evaluation Tool for Base Acoustics and Surrounding Environments), a new web-based framework designed to expedite and simplify accurate NEPA noise assessments. QUIETBASE will integrate physics-based noise predictions with spectral filters that account for terrain, structures, vegetation, and species-specific hearing. Modeled or measured noise spectra can be translated into impact-relevant metrics and visualized using geospatial tools. An initial QUIETBASE demonstration for the Environmental Security Technology Certification Program focuses on rocket launch noise and sonic booms, with validation for threatened and endangered bird species at Vandenberg Space Force Base.

10:05

2aNS4. The magical number.7, plus or minus.2: Some limitations of equal-energy noise metrics. Andrew Christian (Flight Vehicle Acoust. Branch, NASA Langely, 2 N. Dryden St., Hampton, VA 23681, andrew.christian@nasa.gov)

The equal-energy hypothesis (EEH) suggests that human annoyance can be best predicted by integrating noises in a way that preserves overall acoustic energy. This concept underpins nearly all noise metrics used for air vehicles today—both for current commercial aircraft and emerging technologies, such as drones. The results of several recent NASA psychoacoustic tests have produced best-fit models that do not conform to the EEH. Further, the models have a numerical coincidence around the number.7 (depending on parameterization), indicating a trend toward large numbers of events and long-single events being more annoying than the EEH would indicate. If true, this coincidence would extend further as it would help to explain several adverse noise effects that are seen in real-world situations: that the noises of hovering air vehicles may be more annoying than those of passing aircraft, that there has been an increase in the apparent sensitivity of communities surrounding commercial airports over time (i.e., as the number of aircraft movements has increased, even as the individual aircraft become significantly quieter), and that residents who live underneath flight tracks but far away from airports appear more impacted than those receiving an equivalent noise dose nearer to the airport.

10:25–10:40 Break

Contributed Papers

10:40

2aNS5. Quantifying the differences between communities within a survey using the community tolerance level methods. Ken Kaliski (RSG, 55 Railroad Row, White River Junction, VT 05001, ken.kaliski@rsginc.com), T. R. Haac (RSG, White River Junction, VT), and Ben Hoen (Lawrence Berkeley National Lab., Berkeley, CA)

The community tolerance level, or CTL, has been recognized as a convenient tool for providing a decibel-denominated difference of noise tolerance between communities. For example, this method has been used for comparing the tolerance to road, rail, and air traffic noise in different countries (e.g., Morinaga, 2017). There are a variety of ways community, or groups of individuals, can be defined even within a single survey. For wind turbine social surveys, communities need not be restricted to geographical boundaries. We can have communities of uncompensated or compensated individuals. We can have communities that took the survey via mail versus phone. Communities might also be differentiated if they live near a new versus old wind project, or, separately, moved in after the turbines were erected versus before. In this presentation, we describe and use a tool to calculate the CTL for these and other different communities that responded to a wind turbine social survey. We will discuss the implications for conducting social surveys on noise annoyance and for providing decibel adjustments for predicting high annoyance.

10:55

2aNS6. Annoyance of loudness-matched samples of wind farm noise collected by citizen scientists. Anne C. Balant (Dept. of Commun. Disord., State Univ. of NY at New Paltz, 1 Hawk Dr. HUM09, New Paltz, NY 12561, balanta@newpaltz.edu) and Heather L. Lai (Eng. Programs, SUNY New Paltz, New Paltz, NY)

We present the results of preliminary listening tests on samples of wind farm noise obtained during the early phases of a long-term, community-based data collection at two sites in New York State. Both sites are in rural communities with residents who were living in the community prior to installation of the turbines. One site has multiple sampling locations on one property, and the other has multiple locations throughout the community. Sampling points were within one mile of multiple turbines. Data were collected by citizen scientist volunteers using weather-proofed systems with type 1 sound level meters. Noise samples were obtained both before and after deployment of the wind turbines, in different weather conditions, and at different times of day. Samples differ widely on the overall sound level and the presence or absence of amplitude modulation. Some include audible environmental sounds due to vehicles, wildlife, *etc*. Samples were rated by young adults with no prior listening training who passed a hearing screening. Testing was conducted in a single-walled audiometric test suite. An adaptive psychophysical method was used to adjust the stimuli to have equal perceived loudness. Participants then rated the annoyance of the stimuli via magnitude estimation and a paired comparison method.

11:10

2aNS7. Sound matters: Exploring noise awareness and behavioral reponses on the Utah Valley University pedestrian bridge. Yi Yin (Social and Behavioral Sci., Utah Valley Univ., 800 West University Parkway, Orem, UT 84058, Yi.Yin@uvu.edu), Ainun S. Israd Hakim (College of Humanities and Social Sci., Utah Valley Univ., Orem, UT), Brian D. Patchett, and Bonnie Andersen (Phys., Utah Valley Univ., Orem, UT)

Noise has become a growing concern in urban areas due to its influence on public health, comfort, and daily life (Fink, 2019; Montes González *et al*., 2023). Sound exposure on the UVU Pedestrian Bridge may affect users in multiple ways, from hearing-related risks to increased stress and discomfort. However, research on noise levels at this bridge and their influence on pedestrian behavior remains limited. To fulfill this gap, this study examines pedestrians' noise awareness, perceptions, and behavioral responses while crossing the bridge. We collected 409 responses through an anonymous survey that included questions on noise awareness, reactions, usage patterns, and demographics. Statistical analysis will be used to compare differences in pedestrian experiences across groups and explore the relationship between noise awareness and behavioral responses. This research aligns with the Utah Pedestrian Safety Action Plan (2020), which emphasizes creating safe and comfortable spaces for individuals of all ages and abilities. By analyzing noise awareness and pedestrian behaviors, this study aims to inform future design, safety improvements, and educational strategies that enhance pedestrian comfort and reduce noise-related impacts in everyday walking environments.

11:25

2aNS8. Individual response trends in a psychoacoustic test comparing UAVs and road vehicles. Andrew Christian (Flight Vehicle Acoust. Branch, NASA Langely, 2 N. Dryden St., Hampton, VA 23681, andrew.christian@nasa.gov), Laura Thomas (OSTEM Intern, NASA Langley, Miami, FL), and Aaron B. Vaughn (Flight Vehicle Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

This presentation discusses a reanalysis of data from a psychoacoustic test in 2017 that compared annoyance responses due to the noise of uncrewed aerial vehicles (UAVs) and road vehicles. This time, instead of averaging the responses across subjects, the regression model is applied to a single subject's data at a time. The analyses show that the individual subjects' responses were not simply distributed randomly around the mean response—individuals had strategies for listening that were markedly different than the average would suppose and were based on responding to different attributes of the sounds. Similarly, the model that is useful for fitting the averaged data is not found to be particularly useful to describe nearly any individual's response. This conclusion is compared with the results of other recent tests that have shown similar conceptual trends. These results are discussed in the context of the assumptions made by multilevel statistical analyses: What does it mean when one reads the typical comment about psychoacoustic results that "large individual variability was observed?"

11:40

2aNS9. Beats of multiple tones applied to multirotor aircraft. Ze Feng (Ted) Gan (Mech. and Aerosp. Eng., Michigan Technolog. Univ., 1400 Townsend Dr., 911 RL Smith Bldg., Houghton, MI 49931), Vitor Tumelero Valente, Kenneth S. Brentner, and Eric Greenwood (Aerosp. Eng., The Penn State Univ., University Park, PA)

Previous work by the authors [6th ASA/ASJ Joint Meeting, 2025] showed evidence supporting the importance of beats for the noise generated by unmanned aerial vehicles (UAVs), particularly in generating significant values for the sound quality metric of fluctuation strength. Beats have been studied in the literature for two tones, but UAVs typically have four or more rotors, each of which generates tonal noise. Therefore, the beats of more than two tones represent an important knowledge gap in the literature, which this presentation addresses. Even when rotors were set to the same setpoint for the blade passage frequency (BPF), beats were generated by small BPF variations, which followed Gaussian distributions. The spectral peaks in the envelope of time-varying SPL corresponded to the BPF differences between all rotor pair combinations. The BPF distribution and noise generated by a single rotor were used as inputs into the developed noise prediction methodology to predict the probability distributions of beat frequencies and SPL modulation amplitudes. The predictions showed good agreement with noise measurements of 2 and 6 rotors, where beats were significantly audible with fluctuation strength > 0.1.

TUESDAY MORNING, 12 MAY 2026 FRANKLIN HALL 10, 7:55 A.M. TO 12:00 NOON

Session 2aPA

Physical Acoustics, Biomedical Acoustics, Noise, Education in Acoustics, and Computational Acoustics: Celebration of Allan D. Pierce's Contributions to Acoustics I

T. Douglas Mast, Cochair
Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Andrew A. Piacsek, Cochair
Phys., Central Washington Univ., 400 E. Univ. Way, Dept. of Phys., Ellensburg, WA 98926-7422

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Bookending a career in acoustics: A. D. Pierce and quantum mechanics: *Ad astra*. Kenneth G. Foote (Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu)

A. D. Pierce's scholarship is legendary. This has encompassed all aspects of publication, from problem formulation through manuscript composition. Arguably, Pierce's first problem was the topic of his doctoral thesis (1962) at the Massachusetts Institute of Technology: electron–lattice interactions, aiming to be foundational for a quantum mechanical theory of macroscopic equilibrium thermodynamics. This was rejoined 53 years later, immediately following retirement as ASA Editor-in-Chief, through a presentation at an American Physical Society section meeting. The particular subject was the decay of mechanical vibrations by electromagnetic radiation. Neither work was formally published in the journal literature, failing to meet Pierce's exacting standards. However, a theoretical

problem was defined, regarded here as both fundamental and operationally important in astrophysics. The connections are established in two questions: how do vibrations in stars, as well as in the 40kg mirrors of the laser interferometer gravitational-wave observatories in Hanford, WA, and Livingston, LA, dissipate, and how long does this take?

8:20

2aPA2. Reminiscence of a kind and generous mentor. Yves H. Berthelot (118 Hawthorn Court, Lake Frederick, VA 22630, yvesberthelot1@gmail.com)

It was an incredible opportunity, albeit rather intimidating, to meet Allan Pierce during my days as a graduate student. At the time, we were both working on underwater sound generated by moving laser beams. I discovered that Allan was a kind and generous mentor who took me under his wings when I joined the faculty at Georgia Tech. This non-technical presentation will include reminiscences of his mentorship and of his masterful ability to build a strong acoustics program at Georgia Tech. It will also briefly touch on two of Allan's contributions: the generation of underwater sound by laser beams moving at sonic speed; and the diffraction of outdoor sound by irregular topography using Matched Asymptotic Expansions (MAE) theory.

8:40

2aPA3. Allan D. Pierce—theoretician and historian. Mark F. Hamilton (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

The name Allan Pierce first came to my attention as a graduate student when I encountered his classic textbook on acoustics that had just come out, and I recognized immediately that the author was very special on several levels. I did not fully appreciate Allan's passion for contributing to the theory and history of physical acoustics until he participated in writing Chapter 4 of *Nonlinear Acoustics* (Hamilton and Blackstock, editors, Springer Open Access 2024. 10.1007/978-3-031-58963-8). We asked Allan to write the section on the general solution of the Burgers equation, a classical evolution equation in nonlinear acoustics. He returned this section accompanied by an unsolicited eight-page section containing a wonderfully thorough history of the origins of the Burgers equation, beginning with its roots in fluid mechanics (Section 4.5.1). We also asked Allan to write the sections on shock waves. When discussing this material with Allan, I mentioned in passing that existing theory for reflections of waves from weak shocks was restricted to perfect gases, and this motivated him to contribute an unsolicited six-page derivation of the reflection theory for general fluids (Section 4.3.3). Various other initiatives taken by Allan that demonstrate his passion for acoustics will also be discussed.

9:00

2aPA4. Allan Pierce and variational principles. James G. McDaniel (Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, jgm@bu.edu)

The application of variational principles to the formulation of solutions for problems in acoustics and vibration constitutes a central and recurring theme in Allan Pierce's distinguished body of work. His scholarship is characterized by intellectual rigor, mathematical precision, and a disciplined focus on variational principles that yield unified derivations of governing equations and boundary conditions. Of particular significance are his collaborations with Jerry Ginsberg and successive cohorts of graduate students, through which he developed coupled variational frameworks that integrate acoustic and structural dynamics into a coherent theoretical and computational edifice. These formulations gave rise to robust numerical methods that preserve fundamental conservation properties and deliver enhanced stability and accuracy for challenging problems in radiation and scattering, especially those involving heavy fluid loading where fluid–structure interaction dominates system response. This presentation will survey Pierce's contributions in this domain, tracing the development of core ideas, elucidating key methodological innovations, and assessing the enduring impact of his research on contemporary practice in theoretical acoustics, structural acoustics, and computational modeling.

9:20

2aPA5. Fundamental and nearly omnipresent: Allan Pierce's contributions to understanding of attenuation from relaxation processes. T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

One of Allan Pierce's many long-standing interests in acoustics has been the physics of sound attenuation. He has made multiple seminal contributions to the theory and applications of acoustic absorption caused by relaxation processes, working from first principles of physical acoustics. These contributions include a rigorous development of absorption caused by multiple molecular relaxation processes in his classic textbook as well as a theory for acoustic attenuation and dispersion in inhomogeneous media with many spatially distributed relaxation processes. Theoretical results include detailed thermodynamic descriptions of energy and volume relaxation, dispersion relations for a continuously weighted distribution of relaxation processes, and a time-domain wave equation for propagation in a medium with spatially distributed relaxation processes. These results are consistent with experimental measurements, including those observing power-law frequency dependence of attenuation over a wide frequency range, while maintaining physically expected behavior in low- and high-frequency limits. Applications include effects on sonic boom rise times, interpretation of frequency-dependent attenuation of seafloor mud, and simulation of ultrasound propagation in human tissue.

9:40

2aPA6. Allan Pierce and ASA Publications. James Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 86 Water St., Falmouth, MA 02543, jlynch@whoi.edu), Arthur E. Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Kelly Quigley, Elizabeth A. Bury, Katherine Setzer, and Helen Wall Murray (Publications, Acoust. Society of America, Melville, NY)

In addition to his numerous technical contributions, Allan Pierce served as the Editor-in-Chief (EIC) of Acoustical Society of America (ASA) Publications from 1999 to 2014, the second longest serving EIC. His geographical orbit during that period stretched from Boston University (BU), his academic home, to the West Barnstable ASA Publications Office (near his home) to the Woods Hole Oceanographic Institution (WHOI). This talk will describe Allan's interactions with the Editorial Staff, ASA headquarters, and WHOI, BU and other academic personnel during that time. Numerous well-known ASA members followed Allan in this orbit, and the interactions were both interesting and entertaining.

10:00–10:20 Break

10:20

2aPA7. Explosions, infrasonics, microbarographs, and Allan Pierce. Joe W. Posey (2426 Redbud Ln., Charlottesville, VA 22911, jwposey@me.com)

More than 50 years ago, Allan Pierce was my graduate school advisor at MIT. I can attest to his outstanding qualities as a mentor, educator, scholar, and friend. I worked with him for 4 years on an Air Force grant to study the infrasonic disturbance generated in and propagated through the atmosphere from an explosive source. The research was well-documented in contemporary refereed journals, including a cover article in the journal Nature. Highlights of the work are summarized here, along with comments on Allan's actions and training methods regarding graduate students.

10:40

2aPA8. In the Sonic Boom Arena with Dr. Allan D. Pierce. Domenic J. Maglieri (122 Oak Rd., Williamsburg, VA 23185-3215, sonicboomexpert1@verizon.net)

I had the pleasure of working with Allan on six separate occasions over a 25-year span, beginning in 1968. His six entries addressed boom problems that would be encountered during SST operations. Included were spikes on the on-track boom signatures, resulting from variability in the lower layers of the atmosphere. He and I extended this early work to look at boom signature variability at lateral locations out to cutoff. He joined Sanai and Toong ballistic range experiments aimed at determining the maximum amplitude of focus booms from accelerated flight and the focus generated by refraction when flying at Mach cutoff. Acoustic disturbances were observed in the New England area. Allan teamed with Rickley of the FAA, acquired measurements, and provided ray tracing computations that showed the disturbances were secondary booms from incoming Concorde flight to Kennedy airport. He noted that for weaker booms, molecular relaxation effects play a major role in increasing shock rise time, further softening the boom. His last task looked at whether a focus boom exists at the lateral edge of the carpet boom where the rays are refracted.

11:00

2aPA9. Allan Pierce's contributions to solving the sonic boom problem. Victor W. Sparrow (Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

Years ago, we didn't know enough about propagation through a realistic atmosphere to build a quiet supersonic aircraft. Then along came Allan Pierce. Firstly, Allan was one of the first, if not genuinely the first, to tell us that turbulence was the primary cause of sonic boom signature distortions in the atmospheric boundary layer. Secondly, Allan significantly advanced our understanding of secondary sonic booms. His work with Ed Rickley conclusively showed that Concorde flights into the U.S. created the low rumbles heard by many New England residents. This provides insights on how to minimize such secondary boom impacts. Thirdly, and perhaps most importantly, Allan led the way to understand the proper effect of humidity on the rise times of sonic boom shocks. More humid air gives shorter rise times. Hence, we now have accurate boom propagation models for the design of quiet supersonic aircraft. [Work supported by the FAA through ASCENT Project 57 under the Office of Environment and Energy (AEE). Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

11:20

2aPA10. Sonic booms with swallowtails: Lessons in acoustics from Allan Pierce. Andrew A. Piacsek (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

A full understanding of sonic boom propagation through a turbulent atmosphere, like many complex problems, developed in stages. By the early 1970's, Allan Pierce had made significant contributions toward explaining the observed spatial and temporal variability of sonic boom signatures recorded at the ground [J. Acoust. Soc. Am. **49**, 906–24 (1971)]. A key insight was that turbulence would effectively ripple sonic boom wavefronts, which would either focus or defocus, according to geometric acoustics. Focused booms could experience wavefront folding, characterized by a swallowtail pattern. The theory explained some features of observed sonic booms and lab experiments of the era, but solutions were, by necessity, approximate. When NASA reignited research in this area in the early 1990's, an opportunity arose to test the wavefront folding theory with numerical propagation models (that also included molecular relaxation), which is where I entered the sonic boom arena as Allan's graduate student at Penn State. Along with the results of my doctoral work implementing a time domain model of weak shocks with curved wavefronts, I will share the invaluable lessons I learned from Allan about how to be a scientist.

11:40

2aPA11. A model experiment using spark-produced *N* waves to study the effect of turbulence on waveform and rise time. Bart Lipkens (Pharmaceutical and Administrative Sci., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, blipkens@wne.edu)

In the 1990s NASA engaged in a High-Speed Research Program to evaluate the feasibility of overland sonic boom propagation. There were several research groups active at various Universities. The author was part of the team at the University of Texas at Austin, led by David Blackstock. At the same time, Allan Pierce directed a significant effort of sonic boom propagation studies at Penn State University. The author's work focused on a model experiment in which spark-produced *N* waves propagated across a turbulent flow field of a plane jet. The *N* waves were measured by a wide band condenser microphone and the turbulence was measured by hot-wire anemometry. Waveform distortion, rise time, and peak pressure were measured as function of propagation distance and turbulence intensity. The observed waveform distortions were similar to sonic booms propagating through a turbulent atmosphere. On average, turbulence increases rise time and reduces peak pressure. The data were used to test sonic boom propagation models. Allan Pierce had developed a wave front folding model [J. Acoust. Soc. Am. **49**, 906–24 (1971)] that predicts sonic boom or weak shock rise time. The model experiment rise time data were found to agree well with the wave front folding model.

TUESDAY MORNING, 12 MAY 2026 FRANKLIN HALL 8, 8:00 A.M. TO 11:35 A.M.

Session 2aPP

Psychological and Physiological Acoustics: Individual Factors in Spatial Speech Unmasking

G. Christopher Stecker, Cochair
Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131

Nirmal Kumar Srinivasan, Cochair
Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., 8000 York Rd., Towson, MD 21252

Chair's Introduction—8:00

Invited Papers

8:05

2aPP1. Unique contributions of acoustic and cognitive factors to spatial release from masking. Nirmal Kumar Srinivasan (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, nsrinivasan@towson.edu)

Spatial Release from Masking (SRM) refers to the improvement in speech intelligibility that occurs when target and masker sounds are spatially separated. While traditionally attributed to acoustic cues, such as interaural time and level differences, growing evidence indicates that non-acoustic factors also shape the magnitude of SRM. This study quantifies the relative contributions of acoustic cue sensitivity and cognitive abilities on SRM using behavioral speech-in-noise tasks paired with a comprehensive cognitive battery. Cognitive factors were assessed using the Trail Making Task (attention shifting and processing speed), Flanker Task (inhibitory control), Stroop Task (interference resolution), and a working memory task. Results show that acoustic cues robustly support SRM in predominantly energetic masking conditions; however, under informational masking, individual differences in cognitive performance significantly modulate SRM magnitude. Individuals with stronger inhibitory control, faster attentional switching, and higher selective auditory attentional capacity demonstrated greater SRM, even with identical acoustic cues. These findings highlight SRM as a joint outcome of peripheral auditory processing and central cognitive mechanisms. Understanding how these factors interact has implications for hearing-aid processing strategies, assessment of hearing impairment, and the design of spatial audio systems that better support real-world communication.

8:20

2aPP2. Do working-memory and attention-control measures predict variability in binaural speech unmasking? Brittany T. Williams (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, brittany.williams1@northwestern.edu), Angela M. AuBuchon, and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Spatial release from speech masking (SRM) refers to improved speech recognition when speech maskers are perceived to be separated from target locations. Binaural manipulations, such as conflicting interaural time and level differences, have been shown to reduce SRM and perceived lateral position. We assessed the impact of perceived target-masker separation by measuring spatial judgments of binaurally manipulated speech and correlating those judgments to individual listeners' SRM scores. The results account for 2–3 dB of variation in SRM across the measured lateralization range. Individual SRM scores, however, varied more than 10 dB, suggesting that other factors primarily account for variation in speech unmasking. Past studies suggest that differences in working memory and attention could be responsible. Here, we relate variation in SRM and spatial perception to variation in working memory as assessed by an auditory running digit span task and selective attention as assessed by the Stroop Squared, Simon Squared, and Flanker Squared tasks. Results suggest the importance of considering non-acoustic aspects of listening ability, which may limit access to benefits of spatial hearing [Work supported by NIH R01-DC016643.]

8:35

2aPP3. Individual differences in spatial release from masking across the adult lifespan. Daniel Fogerty (Speech and Hearing Sci., Univ. of Illinois Urbana–Champaign, 901 S. Sixth St., Champaign, IL 61820, dfogerty@illinois.edu) and Meisam Arjmandi (Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC)

The ability to spatially separate multiple talkers into different auditory streams is critical for understanding speech in everyday environments. Sensitivity to spatial separation can result in spatial release from masking, in which listeners recognize speech better when target and competing talkers originate from different rather than co-located positions. However, substantial individual differences in spatial release from masking can be observed, potentially contributing to functional differences for understanding speech in real-world spatialized environments. This study examined individual differences in spatial release from masking for a large cohort of adults aged 20–79 years. Listeners were tested on co-located and separated conditions of speech-on-speech masking using the coordinate response measure. They also completed cognitive measures from the NIH cognition toolbox and auditory measures including pure-tone and words-in-noise thresholds. Cognitive and auditory factors were evaluated as predictors of spatial release from masking across the adult lifespan. Initial analyses from a subset of participants indicate a significant role for working memory in explaining individual variability in spatialized speech processing.

8:50

2aPP4. Associations between talker head orientation benefit and extended high-frequency thresholds. Allison Trine (Speech and Hearing Sci., Univ. of Illinois Urbana–Champaign, 901 S Sixth St., Champaign, IL 61820, at17@illinois.edu), Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois Urbana–Champaign, Champaign, IL)

Talker head orientation is an important spatial aspect of a multi-talker auditory scene. Talker head-orientation related (THOR) cues provide a benefit for speech-in-speech recognition in scenarios where a target talker is facing the listener and masker talkers are facing away. We have observed individual listener differences of up to $\sim$10 dB in THOR benefit. Due to talker directivity effects, extended high frequencies (EHFs; $>$ 8 kHz) show the greatest attenuation as head orientation angle increases, thus EHF cues in speech contribute to the THOR benefit. Consequently, we hypothesized that listener EHF sensitivity, as measured by EHF pure-tone thresholds, would predict the THOR benefit for speech-in-speech recognition. We tested this hypothesis using datasets from four experiments that assessed target speech recognition using facing versus non-facing masker talkers. Significant relationships of weak-to-moderate strength between 16-kHz pure-tone thresholds and THOR benefit were observed in two of the four studies. Potential explanations for these mixed findings will be discussed. [Work supported by NIH Grant R01-DC019745.]

9:05

2aPP5. Effects of auditory injury on binaural listening in service members with normal hearing thresholds. Gregory M. Ellis (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., 4494 Palmer Rd. N, Bethesda, MD 20814, gellis@alakaina.com) and Douglas S. Brungart (Audiol. and Speech Pathol., Walter Reed National Military Medical Ctr., Bethesda, MD)

Recent studies have shown that noise exposures that produce temporary threshold shifts (TTS) can temporarily degrade NoSpi tone-in-noise detection, and that individuals who report frequent auditory symptoms after noise exposure often experience increased hearing difficulties and chronic degradations in both NoSo and NoSpi tone-in-noise detection despite having hearing thresholds in the normal range. The present study used data from two groups of Service Members (SMs) to determine whether these effects extended to speech-in-noise performance. Both groups of SMs completed a speech-in-noise test, pure-tone audiometry, and a questionnaire about TTS history. The first sample completed a version of the Modified Rhyme Test (MRT) with monaural, diotic, and binaural listening conditions in speech-shaped noise. The second completed the Spatial Digit Test (SDT), a test that requires listeners to use differences in interaural timing cues to segregate two simultaneous digits. SMs with a history of frequent TTS performed worse on both speech-in-noise and speech-on-speech masking tasks. However, the results did not provide strong evidence that TTS impairs binaural listening more than monaural listening. [The views expressed in this presentation are those of the authors and do not necessarily reflect the official policy of the Department of War or the U.S. Government.]

9:20

2aPP6. The roles of selective attention, working memory, and age in bilateral speech interference for single-sided deafness cochlear-implant users. Michael A. Johns (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., 2110 Prichard Rd., Silver Spring, MD 20902, mikejohns15@gmail.com), Sandeep A. Phatak (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), and Joshua G. Bernstein (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD)

Cochlear implants (CIs) for single-sided deafness (SSD) can partially restore the ability to use binaural interactions to improve masked speech intelligibility. When target and masking speech are presented to the acoustic-hearing (AH) ear, presenting a copy of the masker to the CI ear facilitates spatial release from masking (SRM) and improves speech perception. The reverse is not true, however; presenting a copy of the masker to the AH ear can impede CI-ear speech perception. This "bilateral speech interference" varies greatly across listeners and understanding the source of this variability is crucial to maximizing SRM. Bilateral speech interference was assessed in 19 SSD-CI listeners (ages 20–77 years) using the coordinate response measure corpus. Participants also completed two non-auditory selective-attention tasks (Stroop, Flanker) and a working memory task (Reading Span). Bilateral speech interference was significantly associated with older age and poorer selective attention, while better working memory was significantly associated with improved performance overall. The role of domain-general selective attention points to possible clinical applications including selective-attention training or pre-operative screening for risk of interference. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy of the Department of War or U.S. Government.]

9:35–9:50 Break

9:50

2aPP7. Cognitive factors influencing outcomes with advanced noise management. Varsha H. Rallapalli (Commun. Sci. & Disord., Univ. of South Florida, 4202 Fowler Ave. PCD 1017, Tampa, FL 33620, varsha8904@gmail.com), Sophia Kreismer, and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Speech intelligibility in noise varies widely across hearing aid users despite noise management features. Traditional approaches, such as adaptive beamforming and single-channel noise reduction, can reduce background noise but may introduce signal distortions that degrade speech intelligibility, particularly for listeners with poorer cognitive abilities. Newer deep neural network-based noise reduction (DNN) can potentially improve signal-to-noise ratio while minimizing distortions. This study aims to characterize how individual cognitive abilities influence outcomes with DNN relative to adaptive beamforming across acoustic scenes. Listeners with hearing loss completed sentence intelligibility tasks using wearable hearing aids programmed with adaptive beamforming and DNN. Target speech and competing maskers were presented from on- and off-axis locations in co-located and spatially separated configurations, with masker types including gender-matched two-talker speech or diffuse noise. Participants also completed a cognitive test battery assessing working memory, processing speed, and attention control. Results-to-date indicate that on average beamforming results in higher intelligibility with two-talker maskers, whereas DNN provides better intelligibility in diffuse noise and for off-axis targets, with large individual variability. The presentation will discuss how variability in intelligibility across these noise management strategies relates to individual cognitive profiles and susceptibility to signal distortions.

10:05

2aPP8. Individual variation in sound localization accuracy with hearing aids and associations with speech understanding in spatialized speech mixtures. Elin Roverud (Boston Univ., 635 Commonwealth Ave., Dept of Speech, Lang. & Hearing Sci., Boston, MA 02215, erover@bu.edu) and Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

Listeners with hearing loss report difficulty understanding speech in the presence of background talkers (the "cocktail party problem") even while wearing hearing aids. In this listening situation, a listener must be able to focus on the target talker using cues related to the talker's voice and spatial location. Hearing aids provide necessary amplification for listeners with hearing loss, but they can also disrupt sound localization, which may counteract the positive effects and impact speech recognition in spatialized speech mixtures. To investigate this possibility, we conducted a study in which normally hearing listeners performed two tasks with and without hearing aids. In one task, listeners localized individual words presented from loudspeakers. In another task, a five-talker mixture was presented and listeners identified words spoken by a target talker at a specified location. We observed striking individual variations in the degree of disruption in localization accuracy with hearing aids, which were associated with reductions in word identification accuracy. The results highlight that there are listener-specific effects of hearing aids on spatial perception that may impact speech understanding in cocktail party environments, although additional work is needed to evaluate these effects in listeners with hearing loss. [Work supported by NIH NIDCD R01DC015760.]

10:20

2aPP9. Reverberation delays maturation of spatial release from masking in complex listening environments. Z. Ellen Peng (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ellen.peng@boystown.org), Darby Durbin, Victoria Sweeney, and Trevor Scarlett (Boys Town National Res. Hospital, Omaha, NE)

Children often struggle to understand speech in noisy environments, particularly when competing talkers are present. Spatial release from masking (SRM)—the benefit gained by separating target and masker sources—can mitigate this challenge. However, indoor environments with reverberation are known to reduce binaural cues critical for SRM among adults. How reverberation impacts the developmental trajectory of SRM in children remains unclear. This study examined the minimum angular separation between target speech and a two-talker masker required to improve intelligibility beyond co-located performance, isolating spatial versus speech-based unmasking. Ninety-one children (6–17 years) and 25 young adults with normal hearing completed the task in free-field and simulated reverberant conditions. Preliminary findings reveal a strong age effect for co-located speech perception and, critically, for minimum angular

separation only under reverberation. Younger listeners required larger spatial separation than older children and adults, indicating that the ability to process distorted binaural cues matures slowly and continues into adolescence. These results underscore the vulnerability of developing spatial hearing in reverberant environments and have implications for classroom acoustics and auditory interventions.

10:35

2aPP10. Factors underlying spatial release from masking after brain injury. Tess K. Koerner (Oregon Health & Sci. Univ., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, koern030@gmail.com), Aaron Seitz (Northeastern Univ., Riverside, CA), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Previous research has shown that age and hearing sensitivity can have independent effects on spatial speech unmasking, and that variability in performance can be further explained by differences in cognitive function. Examining spatial speech unmasking in patients with acquired cognitive deficits, such as those stemming from mild traumatic brain injury (mTBI), could provide a useful comparison for better understanding how these factors contribute to variability in performance. The current work measured speech understanding on a spatial release from masking task in a large sample of individuals with ($n = 71$) and without ($n = 88$), a history of mTBI to explore how individual differences in various factors contribute to performance. While it was predicted that individuals with auditory or cognitive processing deficits stemming from mTBI would have greater performance deficits compared to age- and hearing-matched control participants with no history of head injury, no such differences across participant groups were observed. However, additional analyses provided evidence for individual factors that may further explain variability in performance within participant groups and that should be explored in future work, especially for those with auditory difficulties stemming from mTBI. [Work supported by R01DC018166.]

10:50

2aPP11. Improvement in speech-in-noise listening with deep neural network enabled hearing aids. Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, eozmeral@usf.edu), Carrie A. Secor, and Nathan C. Higgins (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Despite technical innovations in amplification and signal processing, hearing aid users still frequently report difficulty understanding speech in noise. Recent advances in hearing aid technology using deep neural networks (DNNs) promise to expand benefits of amplification beyond traditional noise reduction features. The present study evaluated the efficacy of a commercially available DNN-enabled hearing aid in a free-field speech understanding task using the coordinate response measure corpus with noise background. We compared multiple configurations of the DNN-enabled device (DNN-on versus DNN-off) as well as comparison devices with traditional noise reduction technologies. Results showed a significant benefit of the DNN-based noise reduction represented by higher CRM accuracy at all target locations, with the highest behavioral performance improvement observed at lateral target locations (± 120 deg) for the DNN-enabled device compared to the same device with the DNN-off or the comparison devices. Furthermore, individual behavioral and cognitive measures were shown in some cases to have direct associations with this benefit. The results reported here demonstrate the benefits of unique noise reduction features to improve listener performance across multiple objective measures, and listener variability may in part be explained by suprathreshold and cognitive abilities.

11:05–11:35 Discussion

Session 2aSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustic Metamaterials I

Christina Naify, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Alexey Titovich, Cochair
Carderock Div., Naval Undersea Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700

Bogdan-Ioan Popa, Cochair
Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Chair's Introduction—8:00

Invited Papers

8:05

2aSA1. Acoustic impedance matching between air and water. Hesam Bakhtiary Yekta (Mech. Eng., Rutgers Univ., Piscataway, NJ 08854) and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norrisandrew@gmail.com)

The ratio of the acoustic impedances of air and water is less than 1/3600, presenting a significant challenge to sound transmission, a problem that has received some recent interest. A review of proposed methods for achieving transmission of order unity will be followed by a detailed description of a purely solid design comprising parallel elastic plates: a "flex-layer." The impedance matching metasurface is sub-wavelength in thickness, with maximum transmission frequency related to flexural resonances. The very small air–water impedance ratio allows accurate asymptotic approximations for the energy transmitted and the bandwidth. The concept of optimal transmissibility will be discussed and it will be shown that the flex-layer design can achieve the theoretical maximum transmission bandwidth.

8:25

2aSA2. Acoustic Willis scatter with thermallytuned asymmetry. Stephanie G. Konarski (Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723, stephanie.konarski@jhuapl.edu), Alec K. Ikei, and Caleb F. Sieck (Code 7160, U.S. Naval Res. Lab., Washington, DC)

This work employs Willis media and multi-material 3-D printing to explore in situ tuning of acoustic materials. Willis materials couple the kinetic and potential energy through additional parameters, such as through coupled polarizabilities that relate the monopolar and dipolar motion at the microscale for acoustically small scatterers. While Willis media are most simply defined by reflection coefficients that depend on the incident wave direction, coupled polarizabilities further quantify the asymmetric response of a unit cell in a background fluid. The unit cell of interest contains two thin plates, separated by an air gap, which were 3-D printed from both resistive and non-resistive materials. When heated, the plates exhibit thermally tuned mechanical properties that introduce geometric asymmetry within the unit cell. Measurements were obtained using a four-microphone, two termination impedance tube approach and compared to analytic results for acoustic reflection and transmission coefficients, as well as monopole, dipole, and coupled polarizabilities. [Work supported by ONR.]

Contributed Papers

8:45

2aSA3. Design of locally resonant metamaterials with optimized spatial distribution on plates for reducing the antiresonance dip in sound insulation. Klara Chojnacka (Dept. of Mech. and Vibroacoustics, AGH Univ. of Krakow, Al. Adama Mickiewicza 30, Krakow 30-059, Poland)

Locally resonant vibroacoustic metamaterials enhance the sound insulation of structures by introducing mass-spring elements tuned to specific frequency ranges. A major drawback of such systems is the reduction in sound insulation near the upper boundary of the induced stop band. This work presents an optimization framework for non-periodic distributions of resonant elements on thin plates, targeting the reduction of the antiresonance-related dip while maintaining the total mass below that of a fully periodic metamaterial. Firstly, the spatial layout of resonant elements was optimized to suppress the problematic frequency range by exploiting the base plate's displacement field. Next, the remaining available surface area was used for secondary resonators tuned to the antiresonance range. Their placement was optimized to eliminate the characteristic sound insulation drop and to improve the overall effectiveness of the metamaterial without increasing the overall mass. All configurations were compared to a plate with equivalent mass to verify their effectiveness. Topological optimization was implemented using the method of moving asymptotes coupled with a finite element model of a plate with resonant elements in COMSOL Multiphysics and MATLAB. The resulting designs are manufactured and validated through sound insulation measurements in a reverberation chamber according to ISO 10140-2.

9:00

2aSA4. Numerical and experimental investigation of membrane-type acoustic metamaterials for sound-insulating enclosure design. Sara M. Kopeć (Dept. of Mech. and Vibroacoustics, AGH Univ. of Krakow, Al. Adama Mickiewicza 30, Krakow 30-059, Poland) and Klara Chojnacka (Dept. of Mech. and Vibroacoustics, AGH Univ. of Cracow, Krakow, Poland)

Membrane-type acoustic metamaterials represent a promising class of lightweight structures capable of achieving significant low-frequency sound attenuation through localized resonance phenomena. Unlike conventional noise-control materials governed by the mass law, they enable substantial transmission-loss enhancement without a notable increase in mass, making them attractive for applications, where weight constraints are critical. This contribution presents a numerical model describing the vibroacoustic behaviour of membrane-type resonant structures and examines design strategies for tailoring their acoustic performance. Additionally, the potential of combining such metamaterials with traditional insulating materials was evaluated to enhance the low-frequency attenuation of noise-control enclosures. The numerical predictions were validated through experimental measurements of the sound reduction carried out under reverberant conditions for a prototype enclosure wall composed of a conventional insulating material supplemented with membrane-type metamaterial modules. Comparison of the measured data with the numerical results enabled an assessment of the model's accuracy and confirmed its ability to reproduce the overall acoustic behaviour of the hybrid system. The findings demonstrate the effectiveness of membrane-based metamaterials in targeted low-frequency noise reduction and highlight their potential for application in lightweight sound-insulating structures.

9:15

2aSA5. Broadband acoustic transmission between water and air using a finite solid impedance matching structure. Hesam Bakhtiary Yekta (Dept. of Mech. & Aerosp. Engineering, Rutgers Univ., Piscataway, NJ 08854, hb438@scarletmail.rutgers.edu) and Andrew N. Norris (Mech. Eng., Rutgers Univ., Piscataway, NJ)

Sending sound from water to air, or vice versa, is difficult due to the large acoustic impedance mismatch. This talk presents a "finite" version of a solid impedance-matching structure, made of three parallel elastic plates connected by discrete rigid ribs. Numerical analysis and FEA simulations show that as the number of ribs increases, the performance approaches the ideal "infinite" case studied in our earlier work. A minimum Q-factor of 6.5 is observed at 1000 Hz. The effect of the rib number, boundary conditions, and material loss factor are investigated. Results show that fixed–fixed boundaries perform similarly to pinned–pinned boundaries when the plates are long enough. Finally, it is investigated that increasing material damping reduces acoustic transmission, as expected. Radiation patterns show strong agreement between theoretical and simulation results.

9:30–9:45 Break

9:45

2aSA6. Comparing air-flow performance of ventilated acoustic metamaterials using single number parameters. Francesco Martellotta (Dept. Architecture, Construction and Design, Politecnico di Bari, Via Orabona 4, Bari 70125, Italy), Stefania Liuzzi, and Chiara Rubino (Dept. Architecture, Construction and Design, Politecnico di Bari, Bari, Italy)

Ventilated acoustic metamaterials offer a diverse range of solutions for noise control applications, spanning from mechanical equipment and duct silencers to windows. While acoustic performance is typically evaluated according to well established existing standards and, despite a certain degree of variability, this allows a straightforward comparison of different designs, when dealing with air permeability the situation is far less clear. In fact, most of the published studies do not include a characterization of air flow performance, and those that provide data do it in a way that can hardly be useful for comparison purposes. Taking advantage of a recent proposal that borrows the measurement method from ISO 9053-1 standard (so that equipment can be easily found in many acoustic laboratories), and extends it to materials with significantly higher air permeability, two figures of merit depending on the flow regime (specific airflow resistance and still-air discharge coefficient) can be used for comparing different solutions in a straightforward way. Examples of applications and comparisons among different ventilated acoustic metamaterials are shown to support the usefulness of the proposed approach. [Work funded by the European Union—NextGenerationEU—Project Title METAWAVE (New METAmaterial Window for simultaneous Acoustic, Ventilation and Energy performance)—Grant 2022RA785R-PE8-PRIN2022.]

10:00

2aSA7. Acoustic metasurface with narrow coiled channels for tunable low-frequency absorption. Caner Baydur (College of Design and Innovation, Tongji Univ., Siping Rd. No 1239, Shanghai 200092, China, canerbaydur@tongji.edu.cn), Meral Bayraktar (Dept. of Mech. Eng., Yildiz Tech. Univ., Istanbul, Turkey), Xu Wang (Inst. of Acoust., School of Phys. Sci. and Eng., Tongji Univ., Shanghai, China), Yongqi Lou (College of Design and Innovation, Tongji Univ., Shanghai, China), and Yong Li (Inst. of Acoust., School of Phys. Sci. and Eng., Tongji Univ., Shanghai, China)

How to improve low-frequency sound absorption has been a substantial question in the acoustics community. To tackle this challenging task, innumerable efforts have been conducted so far. Most proposals have overcome this issue by increasing the overall thickness or making heavier the structure. In this study, we report an effective method of how to enhance acoustic performance without increasing total weight or thickness of a coiled channel-based sound-absorbing metasurface. By precisely narrowing the channel width, we increase the degrees of freedom, which in turn shifts the absorption peaks (below 150 Hz). We believe our proposal is a simple yet powerful design technique that shifts the absorption peak with minimal intervention, making it highly promising for prospective applications in thin, lightweight metamaterials. [Work supported by the National Key R&D Program of China, the National Science Foundation of China, and the Fundamental Research Funds for the Central Universities.]

10:15

2aSA8. Acoustic Resonance-Based Absorptive Metamaterial Incorporating Micro-Lattices via Fused Deposition Modeling. Jay D. Kline (Univ. of Rochester, 84 Wintergreen Way, Rochester, NY 14618, jaydk5050@gmail.com) and Christina Naify (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This study aims to develop an acoustic metamaterial featuring broadband absorption and minimal geometric complexity by combining inhomogeneous Helmholtz resonator (HR) arrays and micro-lattices, allowing for a cheap and scalable material that can be fabricated as a single part via fused deposition modeling. We first investigate the impact of HR neck geometries on optimal absorption given a fixed cavity geometry. Discrepancies between specified and printed geometries are measured and found to have a significant qualitative impact on the acoustic performance of designs. Adjustments are proposed and found to reconcile this difference. Exposed infill patterns are then combined with resonators both in parallel and in sequence, and are tested with a rigid-backed impedance tube. In particular, parallel combination successfully reduces print time and material cost while simultaneously increasing absorption across all frequencies compared to either individual metamaterial.

10:30

2aSA9. Effects of variability in 3-D printed acoustic liner geometry on acoustic impedance. Laura Thomas (OSTEM Intern, NASA Langley, 10555 W Flagler St., Miami, FL 33174, lthom184@fiu.edu), Matthew B. Galles, and Andrew Christian (Flight Vehicle Acoust. Branch, NASA Langely, Hampton, VA)

Acoustic liners can be installed within the nacelle of turbofan engines to reduce the emitted noise. These liners are generally composite panels with a honeycomb core sandwiched between a hard panel and a perforated panel which act as resonators which can be tuned to attenuate particular frequencies by manipulating the geometry of these elements. 3-D printing is an attractive way to fabricate liner prototypes to be tested in a laboratory environment to aid in concept and model development. This project sought to understand the effects on acoustic impedance resulting from the variability

in as-printed prototype liner geometry due to the precision of fused deposition modeling (FDM) 3-D printing. Eleven liner samples were prepared using both variable and fixed layer height FDM processes. The depths of the cavities were measured and were found to be typically precise within a printed sample but sometimes biased away from the intended depth. The samples were tested in a normal incidence tube and impedances were educed. The resonant frequencies of the samples were found to be systematically below those targeted by around 5%. Also, large variability was observed in acoustic resistances at frequencies much lower than the resonant targets.

10:45

2aSA10. Symmetry breaking and non-linear mode analogues in a three-mass mechanical system. Abrar N. Faiaz, M Arif Hasan, Donte McMullen (Mech. Eng., Wayne State Univ., Detroit, MI 48202), and Sobers Abraham (Mech. Eng., Wayne State Univ., Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, hv3534@wayne.edu)

Symmetry breaking in a passive mass–spring system with unequal spring constants reshapes the normal mode landscape and provides a platform for tailoring acoustic responses. For systems with three or more masses ($N \geq 3$), adjusting the spring set $\{ki\}$ partitions the stiffness-ratio simplex into 2^{N-1} regions, each associated with a binary modal signature that encodes nodal parity, edge localization, and mode-shape symmetries relevant to vibration and sound control. Using eigenvalue analysis with parameter continuation, modal patterns are tracked as the ratios $\{ki\}$ vary, revealing modes that interchange the positions of mass oscillations and thereby induce a robust partitioning of parameter space into 2^{N-1} distinct, signature-labeled regions. Small amounts of disorder and viscous damping preserve these signatures, suggesting a topology-like protection mechanism for mode localization in a purely linear mechanical–acoustic setting. A constructive synthesis rule is presented: by sorting the stiffnesses, one can realize a desired binary signature and hence design targeted pathways for energy funneling, bandgap placement, or vibration isolation. Prototypes with 3–5 masses and molecular dynamics simulations validate controlled modal switching and resilience to imperfections, highlighting a route toward compact, programmable mechanical filters, acoustic metamaterial building blocks, and vibration isolation elements that are independent of geometrical periodicity or nonlinearities. [Work supported by NSF grants 2204382 and 2242925.]

TUESDAY MORNING, 12 MAY 2026 FRANKLIN HALL A, 9:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Acoustics Voice Quality and Prosody (Poster Session)

Takayuki Arai, Chair
Information and Communication Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo, 102-8554, Japan

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

Contributed Papers

2aSC1. Comparing spectral shape and spectral moment measures in characterizing the sibilant-nonsibilant contrast in children's speech. Eugene Wong (Univ. of Minnesota, 164 Pillsbury Dr. SE, University of Minnesota, Minneapolis, MN 55455, wong0703@umn.edu) and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

Previous studies suggest that spectral shape parameters (Shadle, 2023; Wong and Munson, 2025) are superior to spectral moments (Forrest *et al.*, 1988) in quantifying acoustic differences between English sibilants /s/ and /ʃ/. It is unclear whether this finding extends to contrast between the sibilant /s/ and non-sibilant /θ/. The current study compared how these two sets of measures predict listener's ratings of the phoneme goodness in children's production of /s/- and /θ/-initial words. Twenty-two adult listeners rated a total of 177 /s/ and /θ/ sounds produced by 43 children of 2 to 5 years of age along a visual analog scale. Children's productions were narrowly transcribed, and collapsed into 6 categories: correct /s/, correct /θ/, substitution of /s/ for /θ/, substitution of /θ/ for /s/, and two intermediate categories that is either sounding closer to /s/ or closer to /θ/. Regression analyses showed that the degree of sibilance, measured as the difference between the peak amplitude in 3000 to 9000 Hz and the minimum amplitude in 550 to 3000 Hz, best reflected the listener's ratings, when compared to spectral variance and center of gravity. However, neither parameters were able to fully distinguish all six transcription categories.

2aSC2. An Acoustic Analysis of English Prominence in L1 and L2 Speakers. Faris N. Alotaibi (Linguist, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, falotai@gmu.edu) and Matthew C. Kelley (English, George Mason Univ., Fairfax, VA)

This study examines the acoustic correlates of prominence in English words as produced by native speakers of English and L2 speakers with Arabic, Chinese, or Spanish as native languages. The aim of the study is to assess the impact of L1 transfer. The acoustic correlates of stress examined are F1, F2, intensity, pitch, and vowel duration. The study also explores the effect of syllable count on these correlates. The results characterize the differences between the nonnative speakers and the native speakers in stress production. Of note, pitch was the only correlate to show relatively uniform performance across all groups. The findings reveal distinct L1-specific influences in each nonnative group. The predictor of syllable count exhibited no significant effect on F1, F2, intensity, or pitch but did exert a statistically significant influence on vowel duration. Overall, the results support the core claims of the Speech Learning Model, particularly the assertion that L1

transfer shapes the production and perception of L2 phonetic features. They also support two out of four assimilation categories in the Perceptual Assimilation Model.

2aSC3. Does reduced pitch variability in depression reflect impairments to laryngeal control? Evidence from clinical voice tasks. Mingli Zhou (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, minglizhou2029@u.northwestern.edu), Erika Exton (Linguist, Northwestern Univ., Evanston, IL), Lauren Grzelak, David Klemballa (Psychiatry and Behavioral Sci., Northwestern Univ., Chicago, IL), Joseph Keshet (Elec. & Comput. Eng., Technion – Israel Inst. of Technol., Haifa, Israel), Vijay Mittal (Psych., Northwestern Univ., Evanston, IL), Stewart Shankman (Psychiatry and Behavioral Sci., Northwestern Univ., Chicago, IL), and Matthew Goldrick (Linguist, Northwestern Univ., Evanston, IL)

Speech in depression is often described as monotonous, characterized by decreased mean pitch, reduced pitch range, slower speaking rate, higher jitter, higher shimmer, and lower harmonic-to-noise ratio. It remains unclear whether these acoustic patterns arise from impaired laryngeal motor control or higher-level speech planning and/or emotional state change. We critically examined the laryngeal motor control hypothesis. Participants included those with current depression (n = 19) and healthy controls (n = 19). Following prior work, we used an interview that elicits spontaneous speech, examining whether individuals with depression show reduced pitch variability. To examine participants' ability to change or maintain laryngeal muscle configurations, we examined two laryngeal control tasks, pitch glide (/u/ rising from the lowest to highest pitch and vice versa) and sustained phonation (/a/ held steady for as long as possible). Multiple pitch and voice quality measures were extracted and Functional Principal Component Analysis was applied to pitch contours. Mixed-effects linear regression analyses revealed significantly reduced pitch variance in spontaneous speech in depression. However, no significant group differences were found in any measures of pitch glide or sustained phonation. These findings suggest that laryngeal motor control is not the source of altered pitch patterns in spontaneous speech in depression.

2aSC4. Prosodicallyconditioned temporal variation in whispered versus normal speech. Annabelle R. di Lustro (Linguist, Cornell Univ., 159 Central Ave., Morrill Hall, Ithaca, NY 14850, ard248@cornell.edu) and Sam Tilsen (Cornell Univ., Ithaca, NY)

Whispered speech prototypically lacks vocal fold vibration and thus exhibits substantial spectral differences from speech with normal vocalization, including the absence of F0-related prosodic cues. Furthermore, whispered speech is aerodynamically disadvantaged because subglottal pressure is exhausted more rapidly, due to higher airflow. These spectral and aerodynamic factors make competing predictions regarding temporal differences between whispered and normal speech: speakers may compensate for impoverished prosodic cues by extending duration in prosodically strong positions; alternatively, they may address aerodynamic needs by lengthening stop closures and shortening high airflow segments (vowels, fricatives). The present study assessed the segmental, prosodic, and global temporal differences between whispered and normal productions of The North Wind and the Sun passage (n = 16 self-reported native English speakers). Results indicate that the main prosodic adjustment of whisper is an exaggeration of phrase-final lengthening, suggesting compensation for the absence of F0-related prosodic cues. Regression model fixed effects show that whisper reduced speech rate by 0.6 words/s and decreased pausing by 4% of passage duration, suggesting these temporal adjustments are not due to respiratory needs. Our results suggest speakers make prosodically conditioned temporal adjustments to facilitate listener comprehension or to augment feedback, rather than segmental timing adjustments to preserve airflow.

2aSC5. Laryngeal elevation in native and non-native lexical tone production. Fengyue Lisa Zhao (Dept. of Linguist, Cornell Univ., 159 Central Ave. Ithaca, NY 14850, fz227@cornell.edu), Sam Tilsen (Dept. of Linguist, Cornell Univ., Ithaca, NY), and Mark Tiede (Dept. of Psychiatry, Yale Univ., New Haven, CT)

Prior acoustic studies have found differences between native and non-native speakers in precision of lexical tone production, but corresponding differences in articulatory mechanisms remain underexplored. This study presents a novel comparison of laryngeal elevation in native and non-native speakers during tone production using coronal-orientation laryngeal ultrasound. Although both extrinsic laryngeal elevation and intrinsic cricothyroid contraction can contribute to vocal fold tension and f0 modulation, we hypothesized that non-native speakers rely more on extrinsic laryngeal elevation, whereas native speakers rely more on intrinsic control, predicting weaker correlations between laryngeal elevation and f0 in native speakers. Two native Mandarin speakers and two phonetically trained non-native speakers produced pitch sweeps and the four Mandarin lexical tones on [ma] while synchronized audio and ultrasound data were collected. Laryngeal elevation was estimated using two measures: an intensity centroid method and a lasso-regression method identifying ultrasound regions predictive of f0. Across participants, laryngeal elevation was positively correlated with f0. Crucially, native speakers exhibited a larger reduction in the elevation–f0 correlation from pitch sweeps to tones compared to non-native speakers. This pattern suggests that lexical tone acquisition may involve reweighting articulatory strategies, shifting from general extrinsic laryngeal elevation toward intrinsic control of f0.

2aSC6. Speaking Fundamental Frequency Differences Over Time and Across Tasks. Nayeli Calderon (Dept. of Speech-Lang. Patholody, California State Univ. Long Beach, Long Beach, CA), Shazneen Chowdhury (Dept. of Neurosci. and Psych., CUNY Queen's College, New York City, NY), Kassandra Fontanez (Communicative Sci. and Disord., New York Univ., 665 Broadway #9, New York City, NY, kfsandra05@gmail.com), Jasey Ukpong (Dept. of Speech Pathol., Xavier Univ. of Louisiana, New Orleans, LA), Line Lloy, and Susannah Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

This study examined variation in speaking fundamental frequency (SFF), looking at both change over time and differences across speech tasks. Speech samples were remotely collected from 46 cisgender women and 40 cisgender men in two tasks: read sentences and sustained vowels. Results showed no evidence of changes in F0 over time for either gender. However, a significant task effect was observed, with higher SFF in the sustained vowels task compared to read sentences. In addition, within-task differences were also found. The intrinsic fundamental frequency (F0) effect was replicated among women with high vowels having a higher F0 than low vowels. Among read sentences, which came from the CAPE-V clinical voice assessment, F0 was higher in the sentence that included multiple voiceless stops. These findings indicate that task type and sentence characteristics influence average F0 and should be considered when interpreting clinical measures and comparing results across studies. While remote data collection allows for sampling across a wide geographic region, task adherence may be an issue in speech tasks that require non-habitual voice use, such as sustained vowel tasks. Future work will expand to include spontaneous speech and gender diverse talkers.

2aSC7. Voice quality's acoustic space is 3-D. Mykel L. Brinkerhoff (Linguist, Univ. of Washington, University of Washington Dept. of Linguist, Box 352425, Seattle, WA 98195-2425, mlbrinke@uw.edu)

Research on phonation has increasingly focused on its phonetic and phonological roles across languages. However, there remains a lack of consensus on how best to capture the acoustic aspects of phonation. This challenge is compounded by the fact that voice qualities may utilize different acoustic cues to signal phonation contrasts. Additionally, recent research has argued that phonation is inherently multidimensional. This study, based on fieldwork voice-quality data from 10 speakers (five male and five female) of Santiago Laxopa Zapotec, addresses this and demonstrates that three dimensions need to be considered when analyzing phonation: spectral slope, aperiodic voicing/aspiration noise, and the amplitude of the fundamental (i.e.,

residual H1*). A two-step analysis was performed: a multidimensional scaling analysis followed by a random forest analysis to determine the optimal number of dimensions and identify which acoustic measures were most closely correlated with each dimension. The results of this analysis show that three dimensions were sufficient to adequately capture the phonation contrasts within Santiago Laxopa Zapotec. The first two dimensions are correlated with spectral slope and aperiodic voicing/aspiration noise. However, the additional dimension was highly correlated with the amplitude of the fundamental (i.e., residual H1*).

2aSC8. Acoustic characteristics of Mandarin vowels in native-speaking children: Effects of consonantal context. KuanYi Chao (World Lang. and Cultures, Univ. of Utah, 255 S Central Campus Dr., Salt Lake City, UT 84112, kuanyi.chao@utah.edu)

While considerable research has examined Mandarin vowel acquisition in non-native speakers, comprehensive acoustic documentation of native Mandarin vowel systems remains limited. This study provides the first systematic acoustic baseline of Taiwan Mandarin vowels in native-speaking children with particular attention to consonantal context effects on vowel production. Thirty-three children (aged 8–11 years) completed an auditory repetition task producing five Mandarin monophthongal vowels (/i/, /y/, /u/, /a/, /ə/) in three consonantal contexts: no onset, /l/, and coronal consonants (/è/ or /ʂ/). Productions were recorded and analyzed in Praat, and formant frequency (F1, F2) values were extracted and plotted to visualize the Mandarin vowel space and within-category variability across contexts. Results revealed that children produced five distinct Mandarin vowel categories. Importantly, consonantal context showed a significant effect, particularly on high vowels: /i/ was produced backed in /l/ and coronal contexts, while /u/ showed opposing fronting patterns ($p < 0.001$) in the same contexts. Non-high vowels, however, showed minimal context effects. These findings establish acoustic baselines for native Mandarin vowels in children, providing critical reference data for L2 learners of Mandarin. The results further demonstrate that consonantal context is a significant phonetic factor in vowel production with implications for understanding L2 vowel acquisition and phonetic coarticulation.

2aSC9. (Up)talking by ourselves: Final rising intonation in the absence of interlocutors. Ben Parkhurst (Columbia Univ., 70 Morningside Dr., New York, NY 10027, bdp2126@columbia.edu), Line Lloy, and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Previous studies of uptalk have primarily sourced data from sociolinguistic interviews and speech corpora of direct conversations, and only men's and women's uses of uptalk have been widely reported on. This study extends this work by examining uptalk in the novel context of spontaneous speech that lacks an interlocutor and by also including non-binary speakers. As uptalk is primarily described in the literature as having the social function of signaling the speaker's turn, its presence in the absence of an interlocutor may suggest that uptalk has become an established prosodic feature of American English. Participants (49 men, 56 women, and 80 non-binary) were recorded online in the absence of another speaker. They were asked to describe their microphone setup and the process of making a peanut butter and jelly sandwich (PB&J). In the microphone task, only 25% of utterances exhibited uptalk, whereas in the PB&J task, 40.57% did. These findings suggest that uptalk is used by speakers even in the absence of an interlocutor when holding the floor with rising intonation is not needed.

2aSC10. Tone and intonation in Mandarin monolingual and Mandarin-English bilingual children. Jie Yang (Speech Lang. Pathol., Univ. of St. Augustine for Health Sci., 5401 La Crosse Ave. Austin, TX 78739, jyang@usa.edu)

Mandarin Chinese employs tone at the word level and intonation at the utterance level to convey meaning contrasts, in addition to consonants and vowels. Mandarin speakers use global and/or localized fundamental frequency (f0) adjustments to maintain linguistic intelligibility at both word- and utterance-level, especially when the f0 directions of tone and intonation are different. The present study compares how monolingual Mandarin-speaking children and Mandarin-English bilingual children manage the interaction of tone and intonation at different developmental stages. Monolingual Mandarin-speaking children and young adults, as well as Mandarin-English bilingual children and young adults, completed the speech production tasks in Mandarin. Stimuli were statements (falling intonation) and questions (rising intonation) ending with monosyllabic Tone 2 and Tone 4 words, e.g., 毛 [mao2]/帽[mao4]/. The accuracy of the tone of the final word was compared across sentence types and among age groups. The average f0 and magnitude of f0 change were measured within words and across sentence types, and compared between languages and among age groups. Results indicate the influences of sentence type, language background, and developmental age.

2aSC11. Schwa Syncope and Stress in English *-ative*: Testing Structural versus Rhythmic Accounts. Bihua Chen (Indiana Univ., Ballantine Hall 704 1020 E Kirkwood Ave. Bloomington, IN 47401, bc1@iu.edu) and Stuart Davis (Indiana Univ., Bloomington, IN)

The English suffix *-ative* alternates between [-è.tɪv] and [-ətɪv]. Previous analyses attribute this to underlying vowel length and destressing (Nanni, 1977; Davis, 1988) or allomorph selection based on inter-stress duration (Stanton, 2019). We propose that *-ative* stress reflects structural constraints linked to English Schwa Syncope (ESS) as well: environments that favor schwa deletion (e.g., *operative)* also favor the pronunciation [-ətɪv]. To test this, we are conducting a production experiment and two perception experiments with native American English speakers. In production, participants read novel and low-frequency *-ative* words (e.g., *marinative*) in two sentence frames differing in prosodic position: *It's a(n) _ process. It's _*. Two perception experiments further test whether listeners' preferences align more closely with ESS environment or inter-stress duration, which will allow a direct comparison between structural and rhythmic accounts. Preliminary results from the production experiment show that in ESS-favored environments, *-ative* was produced with schwa (i.e., [-ətɪv]) in 82% of utterance-medial contexts and 83% utterance-finally. In ESS-disfavored environments, schwa realization dropped to 20% and 26%, respectively, supporting the ESS-based account. Data collection is ongoing. All experimental results will be presented at the conference.

2aSC12. Clear speech effects on Mandarin fricatives. Yung-hsiang Shawn Chang (Dept. of English, National Taipei Univ. of Technol., 1, Sec. 3, Zhongxiao E. Rd., Taipei 10608, Taiwan, shawnchang@mail.ntut.edu.tw) and Yu-Wen Chen (Dept. of Speech-Lang. Pathol. and Audiol., National Taipei Univ. of Nursing and Health Sci., Taipei, Taiwan)

Clear speech generally enhances segmental contrasts, yet little research has examined how these enhancements are realized for Mandarin fricatives. This study investigated (1) which acoustic properties of Mandarin fricatives are modified in clear speech and (2) whether these modifications increase acoustic distinctions among places of articulation. Thirty-five native speakers of Taiwan Mandarin produced five fricatives (/f, s, ʂ, è, x/) in an /a/ vowel context in conversational and clear speech conditions, following the elicitation paradigm of Maniwa *et al*. (2009). Eleven acoustic measures (including spectral peak, spectral moments, F2 at vocalic onset, relative amplitudes at F3, F4, and F5, and normalized fricative duration and amplitude) were extracted. Linear mixed-effects models revealed significant clear speech effects on center of gravity, skewness, kurtosis, F2 onset, and F4 relative amplitude. Most measures also exhibited significant style × place interactions, indicating that clear speech selectively modify place-relevant cues. To determine if these modifications improved place distinctiveness, linear discriminant analyses (LDA) were performed separately for each speech style. Classification accuracy increased from 78.3% in conversational speech to 83.9% in clear speech. These results demonstrate that clear speech strategically modifies specific acoustic dimensions to improve acoustic separation of Mandarin fricatives.

2aSC13. Spectral sensitivity of the Lombard effect in English speakers. Andrew Lubanszky (York Univ., 4700 Keele St., Toronto, ON M3J 1P3, Canada, lubanszky@gmail.com) and Chandan Narayan (York Univ., Toronto, ON, Canada)

This research examines the spectral sensitivity of the Lombard Effect in L1 English speakers with respect to sibilant consonants. Though the effects of noise conditions on Lombard speech have been explored previously, a 'second-order' Lombard effect is yet to be studied. Eliciting speech under individualized noise which approximates a speaker's typical Lombard Speech—that is, speech already produced under broadband noise conditions—may prove the Lombard Effect multifaceted. Does tailored noise lead to speech which is differentiable from that produced in broadband conditions? Spectral analysis indicates weak-to-moderate sensitivity to spectral characteristics of noise in the majority of participants, demonstrated by adjustments in both the frequency and temporal domains. Notable changes from broadband to speaker-specific noise include raising peak frequency, increased segment duration, and decreased skewness. Together, these changes differentiate speech between conditions in speakers who demonstrated typical Lombard Speech under broadband noise. Changes in the frequency domain are highly speaker-specific, however, and occasionally segment-specific. Findings may suggest that acoustic augmentations which differentiate the speech signal from tailored noise may be accompanied by articulatory changes which increase segment stability. Further work is needed to verify articulatory changes and statistical significance of trends for other spectral measures across noise conditions.

2aSC14. Length and Spectral Dynamics of Rising Diphthongs in Nuer. Irina Monich (Wayne State Univ., Detroit, MI), Toby Fowler (Linguist, Univ. of New Mexico, Albuquerque, NM), and Stefon M. Flego (Linguist, Univ. of New Mexico, University of New Mexico, Albuquerque, NM 87106, stefonflego@unm.edu)

Nuer, a West Nilotic language spoken in South Sudan and Ethiopia, is typologically unusual in that it exhibits intraparadigmatic alternations across a three-way division of vowel length categories: short /V/, long /VV/, and overlong /VVV/. Nuer has a series of rising-sonority diphthongs, e.g., /wa/, /we/, /ja/, that participate in this three-way length distinction (e.g., /lwaak/ ∼ /lwaaak/, *cow byre*.nom ∼ gen). In this study, we investigate the relationship between vowel length category and spectral dynamics in Nuer's rising diphthongs. In particular, we find that when short, onglides overlap with the preceding consonantal constriction to such a degree that the vowel often sounds homophonous with the corresponding monophthongal phoneme (/lwak/ → [l^{w} are minimal (e.g., both /lwaak/ and /lwaaak/ can have similar duration, but still differ in their spectral dynamics). This suggesting that these temporal patterns are phonemicized rather than falling out mechanically from durational differences.*both* vocalic components have prominent steady states (/lwaaak/ → [luːaːk]). Furthermore, we show that these dynamic characteristics of vowel length are present even in tokens where differences in vowel *duration* are minimal (e.g. both /lwaak/ and /lwaaak/ can have similar duration, but still differ in their spectral dynamics). This suggesting that these temporal patterns are phonemicized rather than falling out mechanically from durational differences.

2aSC15. Temporal Dynamics of Monophthongs, Diphthongs, and Triphthongs in Icelandic. Stefon M. Flego (Linguist, Univ. of New Mexico, University of New Mexico, Albuquerque, NM 87106, stefonflego@unm.edu), Jenna Mason (English, Virginia Tech, Blacksburg, VA), and Savannah Rivas (Linguist, Univ. of New Mexico, Albuquerque, NM)

Icelandic has a notably large inventory of contrastive vowel classes in stressed syllables, including eight monophthongs /a ɔ ɛ i ɪ u ʏ œ/, five closing diphthongs /ai au ei ou øi/, four opening diphthongs /ja jɛ ju jœ/, and two triphthongs /jau jou/. All 19 of these categories behave as independent vowel classes in a number of respects. For example, they all contrast with one another across a wide range of consonantal contexts of varying sonority profiles (e.g., [strɪːka] '*draw a line*' versusversus [sjaulʊʏr] '*escape*,' [sɪlʊʏr] '*silver*'). They all also exhibit predictable alternations in vowel length conditioned on following consonantal environment (e.g., [fjœːðʏr] ∼ [fjœðrym] '*selfies*'). They all also exhibit predictable alternations in vowel length conditioned on following consonantal environment (e.g., [fjœːðʏr] ∼ [fjœðrym] '*feather*'.nom.sg ∼ dat.pl, [plauːr] ∼ [plau^{h}t] '*blue*'. m.nom.sg ∼ n.nom.sg). In this study, we provide phonetic description of these vowel classes, with special attention paid to the relationship between vowel length and formant trajectory shape. In particular, our data show that some monophthongs, particularly the mid vowels /ɛ œ ɔ/, are quite diphthongal when long, i.e. [eːɛ̯ øːœ̯ oːɔ̯], consistent with previous research. On the other hand, phonemic diphthongal and triphthongal vowel categories show highly reduced movements across the vowel space when short.

2aSC16. From Aspiration to Tone: A Comparative Acoustic Study of Laryngeal Contrasts in the Bengali-Sylheti-Assamese Continuum. Pooja Gambhir (Amity Inst. of Information Technol., Amity Univ., AIIT Dept. I1 Block third Fl., Amity University Noida, Noida, Uttar Pradesh 201301, India, pgambhir@amity.edu), Shyam S. Agrawal (Electronics and Commun., KIIT Group of Colleges, Gurugram, Haryana, India), and Amita Dev (AKGEC, Ghaziabad, Uttar Pradesh, India)

The evolution of laryngeal contrasts from aspiration to tone and frication represents a pivotal phonetic shift across the Eastern Magadhan linguistic continuum. While Standard Bengali maintains a robust four-way laryngeal contrast in its stop inventory (voiced/voiceless and aspirated/unaspirated), its sister languages, Sylheti and Assamese, have undergone significant phonetic restructuring. Utilizing an audio dataset of 16 speakers, this study provides a comparative acoustic mapping of these shifts. The analysis focuses on two primary acoustic correlates: Voice Onset Time (VOT) to measure stop aspiration and fundamental frequency (F0) to track tonogenesis. Preliminary results suggest a compensatory relationship in Sylheti, where the loss of historical aspiration contrasts correlates with the emergence of lexical tone. Simultaneously, the study examines the spirantization of aspirated stops in Assamese, where laryngeal tension is redistributed into frication. By quantifying these laryngeal contrasts, the research offers a mathematical framework for the "Aspiration-to-Tone" evolution. These findings contribute to the understanding of sound change in South Asian languages and provide essential acoustic data for the development of dialect-robust speech processing technologies.

2aSC17. Acoustic correlates of non-coronal fricatives in Akuzipik. Giulia M. Soldati (Linguist Program, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, gmasella@gmu.edu), Sylvia L. Woodrose Schwartz, and Matthew C. Kelley (Linguist Program, George Mason Univ., Fairfax, VA)

The present study investigates the acoustic properties of obstruents in Akuzipik, an endangered Alaska Native language spoken primarily on St. Lawrence Island, Alaska. Currently, there are approximately 540 speakers of the language, all of whom are English-Akuzipik bilinguals. This project expands the scope of investigation of previous work by presenting a systematic acoustic analysis of previously unexamined obstruents: labiodental, velar, uvular, and glottal fricatives. Voicing contrasts are present in all categories except the glottal. Velar and uvular fricatives also exhibit (lip) rounding contrasts. The stimuli consist of words in isolation and in example sentences, all drawn from an existing online dictionary. Each target word contains at least one fricative in word-initial, word-final, or intervocalic environment. Spectral measurements and formant transitions at the (C)V(C) boundaries will be analyzed for correlates distinguishing the different categories. This study is part of a broader multipurpose documentation and reclamation project, which develops and provides resources for individuals interested in learning, or learning about, the language.

2aSC18. Vowel space normalization across development: Validating formant ratio measurements for children's speech. Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., 1579 S Renwick Blvd., Bloomington, IN 47401, mjhenry5519@gmail.com), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

Vowel measures are often normalized to aid in cross-talker or cross-group comparisons. Yet, the ability of normalization procedures to reduce unwanted variation (i.e., improved stability) while maintaining meaningful phonetic variation (i.e., improved interpretability) for vowel space

dispersion metrics across development, remains underexamined. This study compares two formant ratio-based metrics (e.g., vowel articulation index and formant centralization ratio) with the traditional area-based measures of triangular and quadrilateral vowel space area using a synthesized vocal tract growth model spanning five stages from infancy to adulthood. Vowel space dispersion metrics were calculated under three scaling conditions: non-scaled, uniform scaling, and average formant spacing (ΔF) normalization. Ratio-based metrics consistently showed lower variability than traditional area-based measures, regardless of scaling. Uniform scaling further enhanced the stability of ratio metrics while preserving interpretable age-related trends. In contrast, ΔF scaling compressed area-based values and reduced their sensitivity to developmental changes. These findings highlight that metric selection and normalization method interact systematically across development. Ratio-based metrics, especially when paired with perceptually informed normalization, offer a robust and interpretable framework for capturing developmental shifts in vowel articulation, with implications for research and clinical applications in speech development.

2aSC19. Task Effects on Acoustic Cue Patterns in American English Child Speech. Anastasiia Meshcheriakova (MIT, 77 Massachusetts Ave., Cambridge, MA 02139, miau@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (MIT, Cambridge, MA)

Explicit analysis of the acoustic cue patterns that signal contrastive phonological categories in utterances spoken by adults is an important tool for the study of phenomena such as context-governed phonetic variation, the phonetics-phonology interface and phonological change, and for the identification of cognitive representations that are important for modeling human speech processes. Practical applications of individual-acoustic-cue level analysis include differential diagnosis of speech disorders, tracking of intervention efficacy, and L2 teaching and learning. In particular, applying cue-level analysis to child speech addresses an especially compelling gap in speech technology: the lack of automatic, scalable speech-based screening tools for children at risk of later phonological disorders, such as dyslexia, which can be most effectively treated if diagnosed early. We present preliminary results from analysis of two speech-based screening tasks: CTOPP elision and non-word repetition. Using a method for automatic detection of acoustic cue patterns in child speech, we find that the underlying phoneme /t/ is produced with different patterns of acoustic cues between the two tasks. That is, the default production, with abrupt closure and release landmarks, was at least twice as common in the non-word repetition task. These results suggest that evaluations of speech production should control for task-related effects.

2aSC20. Lexical effects on temporal versus spectral variability in developing speech. Melissa A. Redford (Linguist, Univ. of Oregon, 1290 Linguist Dept., University of Oregon, Eugene, OR 97403-1290, redford@uoregon.edu), Doug Shiller (Université de Montréal, Montreal, QC, Canada), and Lucie Menard (Linguist, UQAM, Montréal, QC, Canada)

Developmental theories differ in whether phonological target specification and motor implementation mature via coupled or independent mechanisms. These theories make competing predictions about how lexical knowledge influences different types of production variability during development. We examined temporal variability (a proxy for motor implementation, reflecting movement stability) and spectral variability (a proxy for phonological development, reflecting target attainment) across lexical contexts in four English-French simultaneous bilingual 5-year-olds and four native English-speaking adults with varying French exposure, all from Montreal. Speakers produced 15 repetitions of four /s/-initial CVC real words and four nonwords. Analyses of the coefficient of variation from /s/ duration and COG revealed expected age effects (children > adults). Critically, analyses of bivariate inner bagplot area showed an Age × Lexical Status interaction for temporal variability (/s/ duration × vowel duration): children exhibited greater variability on real words than nonwords, while adults showed no lexical effect. No such interaction emerged for spectral variability (COG × amplitude). If upheld when more speaker data is analyzed, these differential lexical effects suggest movement control and target attainment develop via partially independent mechanisms: stored motor routines appear word-specific and variable in developing children, while phonological targets show minimal lexical sensitivity.

2aSC21. Final falling F0 facilitates quicker identification of utterance completion. Abbey L. Thomas (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., Minneapolis, MN 55455, abbeyt@umn.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Utterance-final cues like falling pitch contour allows listeners to prepare to respond promptly when their conversation partner finishes speaking. We hypothesized that this ability might be compromised in listeners with poorer pitch perception such as those using cochlear implants. During the experiment, listeners heard 128 lists of 3–4 color-shape phrases (e.g., "a green square and a blue circle…") where the final phrase was manipulated to have a falling F0 (indicating completion) or a sustained elevated F0 (indicating continuation). Listeners were asked to repeat the last item they heard as soon as the talker finished speaking; thus, response time was the outcome measure for determining the effect of the falling F0 contour. Typical hearing listeners responded more quickly to a list ending with falling pitch, more slowly to noise-vocoded stimuli lacking pitch cues, and even more slowly to lists ending in high plateaued pitch (indicating the task's sensitivity to perception of continuation and to perturbation in pitch perception). Ongoing data collection with CI listeners is expected to show a reduced within-listener difference in response time to lists ending with the two pitch contours, suggesting these listeners might delay responding in conversation even if the words were perceived correctly.

2aSC22. Effects of Individual Familiarity versus Lexical Frequency on Weak Syllable Reduction. Nadia Clement (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, nadiac@uoregon.edu) and Melissa A. Redford (Linguist, Univ. of Oregon, Eugene, OR)

Lexical frequency is commonly cited in usage-based accounts of language change, including reduction-driven phonological change. But what drives the effect? Does it reflect listener- oriented factors or speaker-centric processes? If listener- oriented, lexical frequency should predict reduction degree better than an individual's familiarity with a word; if speaker-centric, familiarity should be the stronger predictor. This is because while lexical frequency is correlated with subjective familiarity at the population level, these measures diverge at the individual level, particularly for lower-frequency words. To investigate this, we compared weak-syllable reduction in higher- and lower-frequency words controlled for phonological structure. Twenty-five speakers of West Coast American English produced the words in out-of-focus sentential contexts and rated each word for subjective familiarity. Both frequency and familiarity predicted variation in the relative duration and amplitude of weak syllables, but subjective familiarity was the stronger overall predictor, especially for lower-frequency words (where higher-frequency words showed ceiling effects in familiarity). These results support a speaker-centric account of reduction and suggest that the individual-level factor of familiarity, rather than the population-level factor of frequency, drives phonological change.

2aSC23. Abstract withdrawn.

2aSC24. Demonstration of acoustic vowel sound production: Part II. Phase effects. Ava B. Twitty (Dept. of Speech-Lang. Pathol. and Audiol., Towson Univ., Phys. Dept. U.S. Naval Acad., Annapolis, MD 21402, avat1226@gmail.com) and Murray S. Korman (Phys., U. S. Naval Acad., Annapolis, MD)

A do-it-yourself DIY system of apparatus demonstrates how to generate the vowel sound "ah" [/a/]. The system components are motivated by Helmholtz's ["On the Sensation of Tone as a Physiological Basis for the Theory of Music," 1870] sound synthesizer apparatus which includes electro-mechanical coil driven tuning fork oscillators placed near individual Helmholtz resonators. System components include (1) a single tuning fork "buzzer" type oscillator representing the rich harmonic vocal tract excitation GS(t), (2) a vocal track filter response VT(f) represented by selected formant peaks f1, f2, f3, f4 (i.e. four Helmholtz resonator tuning fork oscillators—each driven by GS(t)), (3) detection and display of the superimposed radiated sound VT(t). Formant peaks for "ah" were chosen (256, 640, 1024 and 2048 Hz) from the microphone FFT of voice trials. [Original work: H. Helmholtz (1821–1894), early synthesizers: R. Koenig (1832–1901), M. Kohl (1853–1908), pioneers: G. Fant (1919–2009) and K. Stevens (1914–2013).]

2aSC25. Acoustic patterns of voice quality in Korean heritage speakers. Sylvia Cho (Linguist, Simon Fraser Univ., 8888 University Dr., Burnaby, BC V5A 1S6, Canada, sylvia_cho@sfu.ca)

This study examines how multilingual experience shapes voice quality production in Korean heritage speakers. Production data were collected from 28 speakers in Vancouver, Canada, who produced Korean and English speech in read and extemporaneous styles. Acoustic analyses targeted fundamental frequency (F0), long-term average spectra (LTAS), and long-term formant distributions (LTFs), supplemented by a psychoacoustic voice model capturing covariation among perceptually salient measures. Results reveal robust within-speaker cross-language differences across acoustic domains. Speakers produced higher mean F0 and wider F0 ranges in Korean than in English, with effects strongest in read speech and more pronounced for female talkers. Long-term formant analyses showed language-dependent shifts in F2 distributions, consistent with differences in vocal-tract configuration, while higher formants (F3, F4) remained stable, reflecting speaker-specific anatomy. LTAS measures further demonstrated systematic language effects in global spectral energy and spectral tilt, indicating distinct long-term voice settings across languages. Analyses using the psychoacoustic model showed that although between-speaker differences accounted for most variance, consistent within-speaker language-related shifts emerged, particularly along F0-related dimensions. Overall, heritage speakers maintain language-specific voice quality patterns despite shared vocal anatomy, underscoring voice quality as a structured component of multilingual phonetic systems.

TUESDAY MORNING, 12 MAY 2026 FRANKLIN HALL 12, 8:00 A.M. TO 10:25 A.M.

Session 2aSP

Signal Processing in Acoustics and Acoustical Oceanography: Bayesian Signal Processing I

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180

Zoi-Heleni Michalopoulou, Cochair
New Jersey Inst. of Technol., Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ 07102

Ali Djafari, Cochair
Renminxi Rd. 1818, Shao Xing, China

Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. Bayesian dictionary learning detection and impulse response deconvolution in an uncertain underwater environment. Behrooz Mosallaei (Elec. Comput. Eng., New Jersey Inst. of Technol., Newark, NJ), Ali Abdi (Elec. Comput. Eng., New Jersey Inst. of Technol., Newark, NJ), and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ, michalop@njit.edu)

Reliable detection of weak signals in complex propagation environments requires knowledge of both the transmitted waveform and the underlying channel. This knowledge is, however, uncertain or completely unavailable. In this work, we focus on a library of possible probing signals, for example, a linear frequency modulated chirp waveform and a sinc waveform. We develop a block-matrix signal

model that captures the convolution between the transmitted waveform and the unknown channel impulse response; we then formulate the detection problem in terms of structured matrices that incorporate prior probabilities on the possible waveforms in the library. This approach allows us to rigorously analyze detection performance, derive optimal Bayesian detectors, and extend the formulation to dictionary learning–based blind detection, that incorporates prior information on possible source waveforms. Extracting information on the impulse response is also expected to facilitate the solution of problems of critical importance, such as geoacoustic inversion. [Work supported by ONR.]

8:25

2aSP2. Trans-dimensional bayesian matched-field geoacoustic inversion. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, School of Earth and Ocean Sci., University of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca) and Julien Bonnel (Scripps Inst. of Oceanogr., San DIego, CA)

This paper considers Bayesian matched-field geoacoustic inversion, which involves estimating seabed parameters and uncertainties by inverting complex (frequency-domain) ocean acoustic-field data recorded at an array of hydrophones. A trans-dimensional inversion is developed that estimates the posterior probability density by sampling probabilistically over the number of layers and corresponding parameters of the seabed model, over the order and corresponding parameters of an autoregressive residual-error model, and over source/receiver parameters (source position and complex spectrum, array tilt), which are generally unknown. The large number of unknowns is reduced by applying closed-form maximum-likelihood relationships linking the (fixed-dimensional) spectral and error-variance parameters to the (trans-dimensional) geoacoustic and autoregressive parameters. The number of seabed layers and the spectral and error parameters are marginalized out in considering inversion results in terms of marginal posterior probability profiles for the geoacoustic parameters. The inversion is applied to estimate an elastic seabed model, including compressional and shear parameters, for the New England Mud Patch by inverting broadband (20–2000 Hz) matched-field data collected as part of the 2017 Seabed Characterization Experiment.

8:45

2aSP3. Bayesian optimization with local Gaussian processes for geoacoustic inversion. William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wfjenkins@ucsd.edu), Peter Gerstoft (Scripps Inst. of Oceanogr., UC San Diego, San Diego, CA), and Yongsung Park (Woods Hole Oceanogr. Inst., Woods Hole, MA)

Bayesian optimization is a global optimization technique that has previously been demonstrated for underwater source localization and geoacoustic inversion. Whereas conventional methods typically require thousands of forward model evaluations to estimate parameters, Bayesian optimization can approximate the optimal solution within dozens or hundreds of evaluations. The method uses a Gaussian process surrogate model to approximate the objective function, which is defined as the correlation between predicted and measured data. An acquisition function is then used to select new points in the parameter space for evaluation. In this work, we enhance the Bayesian optimization approach by incorporating trust regions based on observed data, allowing for local fitting of Gaussian processes and improved optimization performance. The effectiveness of this method is demonstrated using both simulated and experimental data.

9:05–9:20 Break

9:20

2aSP4. A Bayes factor discriminant for active sonar under depth uncertainty. Paul J. Gendron (ECE, Univ. of Masschusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, pgendron@umassd.edu) and Kenneth Bowers (ECE, Univ. of Masschusetts Dartmouth, Dartmouth, MA)

A Bayes factor active sonar (BFAS) discriminant is constructed for depth uncertain acoustic scattering bodies in refractive ocean waveguides for relatively high-frequency broadband reception along vertical arrays. The approach accounts for environmental information regarding the refractive media, as well as surface and volume reverberation models. Uncertainty in the reverberation field and the scatterers' depth, is incorporated through proper marginalization rather than maximization as in more conventional Generalized Likelihood Ratio Tests. The discriminant yields a set of time-varying quadratic forms in beam-delay space, optimally balancing uncertainty of the scatterer and reverberation and noise subspaces in the minimum average risk sense. Utilizing waveguide information, BFAS combines multi-path arrivals, optimally attenuating reverberation subspaces while preserving the target subspace, effectively increasing Signal-to-Reverberation plus Noise Ratios (SRNR) despite target depth uncertainty. Depth-Invariant Modes (DIMs) provide a valuable expansion of the discriminating information as the logarithm of the Bayes Factor, thereby providing lower bounds on the performance of the BFAS. These bounds illustrate that even with depth uncertainty, the BFAS outperforms a single specular arrival detector with perfect knowledge of the scattering body's depth. Performance across various refractive and shallow-water environments lends credence to the Bayesian multi-path combining approach. [Work supported by ONR]

9:40

2aSP5. A Bayesian approach to long-range source localization in the complex propagation environment across the Gulf Stream front. Katherine Hoekstra (Appl. Ocean Sci. and Eng., MIT/WHOI Joint Program, Bigelow Lab., 98 Water St., Woods Hole, MA 02543, katherine.hoekstra@whoi.edu), Julien Bonnel (Scripps Inst. of Oceanogr., San DIego, CA), John Colosi (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Matthew Alford, Charlotte Bellerjeau (Scripps Inst. of Oceanogr., La Jolla, CA), Matthew Dzieciuch (Univ. of California, San Diego, Scripps Inst. of Oceanogr., San Diego, CA), Yongsung Park (Woods Hole Oceanogr. Inst., Woods Hole, MA), John Ragland (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Gunnar Voet (Scripps Inst. of Oceanogr., La Jolla, CA)

Long-range low-frequency acoustic propagation across strong oceanographic fronts presents a challenging source localization problem. During the spring 2023 pilot of the New England Seamounts Calibrated Acoustic Fluctuation Experiment (NESCAFE), a source-receiver mooring pair was deployed on either side of the Gulf Stream with a separation of roughly 150 km. The receiver mooring recorded 23 transmissions from the source mooring, which was at an unknown depth, on two large aperture vertical arrays. The 23 received time fronts exhibit a rich multipath structure with individual arrivals undergoing between 6 and 17 surface and bottom interactions. Arrival time and angle measurements, extracted through deconvolution and beamforming, capture key propagation physics but are subject to ambiguity arising from complex multipath arrivals and environmental uncertainty. We propose a Bayesian estimation method which efficiently solves for an estimate of the posterior mean and variance using a Gaussian observation model of acoustic arrival times and angles. This probabilistic formulation enables the coherent integration of multiple boundary-interacting arrivals and provides an uncertainty-aware approach to localization in a highly complex propagation environment. The results highlight the information inherent in deep-sea multipath arrivals and the utility of Bayesian framing for long-range acoustic source localization across dynamic ocean fronts.

9:55

2aSP6. Evaluating deep learning for seabed properties using Bayesian optimization. Lily M. Holladay (Phys. and Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, lily.holladay@gmail.com), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Sourav Sahoo, and Gaye Bayrakci (National Oceanographic Ctr., Southampton, United Kingdom)

The seabed significantly influences shallow-ocean sound propagation. Sound interacts with the seafloor, where seabed properties determine the bottom loss. Thus, proper modeling of seabed properties is crucial for accurate source localization. While passive sonar inference of seabed parameters has been accomplished using Bayesian optimizations, a deep learning approach may be a simpler yet effective approach. To begin evaluating this approach, a 1D CNN is trained to perform the task of estimating seabed properties. The input data are single-frequency sound levels as a function of range from a towed source. Synthetic training data from a wide range of seabed properties generated by a range-independent normal mode model for ocean environments are used to train the CNN, which is then applied to measured data collected northeast of Scotland. Sound from a towed source was recorded on a bottom-moored hydrophone and processed to obtain single-frequency spectral density levels versus range. These data are input to the trained CNN and to a Bayesian optimization, which returns the posterior probability distributions (PPDs) for seabed parameters. A comparison between the CNN results and the PPDs points to potential improvements in the deep learning approach, including application of Bayesian networks. [Work supported by Office of Naval Research.]

10:10

2aSP7. First steps toward simultaneous Bayesian environmental inversion and source localization for acoustic transmissions across the Gulf Stream front. John Ragland (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, john.ragland@whoi.edu), John Colosi (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Katherine Hoekstra (Appl. Ocean Sci. and Eng., MIT/WHOI Joint Program, Woods Hole, MA), Matthew Dzieciuch (Univ. of California, San Diego, Scripps Inst. of Oceanogr., San Diego, CA), Yongsung Park (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Matthew Alford, and Charlotte Bellerjeau (Univ. of California, San Diego, Scripps Inst. of Oceanogr., La Jolla, CA)

The 2023 pilot for the New England Seamounts Calibrated Acoustic Fluctuations Experiment (NESCAFE23) was an experiment to characterize acoustic propagation through the complex acoustic environment of the Seamounts and Gulf Stream current system. Due to a mooring failure the observations consist of only twenty-three, 200–300 Hz acoustic transmissions over a single day across the Gulf Stream front at a range of approximately 150 km. In addition, there was a hydrographic section along the acoustic path 30 days later. The receptions contain several fully refracted acoustic paths, and many acoustic paths that reflect off the bottom and surface as many as 15 times. These steeper angle paths are less sensitive to the vertical structure of the sound speed than the fully refracted paths. When the environment is accurately known, the source localization problem is fully determined and is robustly solved using linear Bayesian methods. However, if the constraint of a known environment is relaxed, the problem becomes underdetermined. In this presentation, we demonstrate biases to the linear Bayesian source localization due to using different model environments, formulate the simultaneous source localization and environment inversion in a Bayesian framework, and present first steps toward evaluating the inversion [work supported by ONR]

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Methods and Theory in Underwater Acoustics

Prajna Jandial, Chair
Scripps Inst. of Oceanogr., Univ. of California San Diego, 1 Miramar St., 929704, La Jolla, CA 92092

Contributed Papers

9:30

2aUW1. Universal Adaptive Beamforming: A Bayesian Approach. Diego A. Cuji (Elec. and Comput. Eng., Stony Brook Univ., 360 Huntington Ave. Boston, MA 02115, diego.cujidutan@stonybrook.edu) and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

We present a Bayesian universal beamforming framework for array signal processing when the signal's angle of arrival is unknown. Motivated by ideas from universal coding, prediction, and estimation, we discretize the angular domain into a finite hypothesis set and derive sequential posterior weights over candidate steering vectors by marginalizing the unknown waveform under closed-form Gaussian likelihoods. For an i.i.d. Gaussian source in spatially white noise, the posterior reduces to an exponential-weights (softmax) recursion driven by cumulative matched-filter output power, enabling soft angle estimation and online tracking while producing a soft combiner that blends hypothesis-specific signal estimates. We extend the derivation to colored interference-plus-noise via an interference covariance model, yielding evidence statistics based on whitened matched-filter outputs together with determinant-based penalty terms that capture hypothesis-dependent covariance changes. The same Bayesian approach is augmented with an explicit noise-only hypothesis to perform sequential signal detection. The resulting updates connect naturally to optimal adaptive beamformers, including MVDR and linear MMSE.

9:45

2aUW2. Occupancy grids for bearing-only localization and tracking. Erin M. Fischell (Acbotics Res., LLC, 82 Technol. Park Dr., East Falmouth, MA 02536, efischell@acbotics.com), Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE), and Martin Siderius (Portland State Univ., Portland, OR)

A common problem in passive acoustic localization is that of bearing-only tracking, with an objective of estimating latitude and longitude position of one or more contacts using only the bearing information from a beamformed solution of one or more hydrophone arrays. The conventional methods for achieving this require motion and noise models that can be difficult to implement in variable environments and for heterogeneous systems. In an occupancy grid approach, instead of treating the beamformed output as a single bearing peak, a statistical model representing likelihood of occupancy in a uniform Easting/Northing grid is generated for each observation. By updating the occupancy grid iteratively with motion estimates, contacts will converge based on multiple observations. This was demonstrated for two very different field scenarios for multi-pinger localization from a towed tetrahedral array on a fishing boat, and for multi-fixed-array localization of synthetic seal calls [Work Funded by NSF and NFWF].

10:00

2aUW3. The convergence of the Legendre-Galerkin technique for approximating underwater acoustics eigenvalues. Richard B. Evans (99F Hugo Rd., N. Stonington, CT 06359, rbevans@99main.com)

The rate of convergence of the Galerkin technique for approximating underwater acoustics eigenvalues with a Legendre polynomial basis has been theoretically demonstrated [Evans, J. Theo. Comp. Acoust., 33, No. 4, (2025)] and also by comparison with a benchmark test case solution. An accuracy comparison with an alternative Lanczos-tau numerical method [H. Tu *et al*., J. Sound Vib., 492 (2021)] based on Chebyshev polynomials can be made when the characteristic function having the eigenvalues as its zeros is available. A computable characteristic function for a more general physical case could be used adaptively to improve the Legendre-Galerkin eigenvalues with a rational approximation and Chebyshev polynomials [Y. Nakatsukasa *et al*., SIAM J. Sci. Comp., 40, No. 4 (2018)].

10:15

2aUW4. Probability density function PDF generated via an *X-Y* waveshaper. Murray S. Korman (Phys., U. S. Naval Acad., Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21403, mskorman167@gmail.com) and Kevin L. Mcilhany (Phys., U. S. Naval Acad., Annapolis, MD)

In the Underwater Acoustics and Sonar course, seniors learn about the PDF of "processed noise alone" along with the PDF of "processed signals plus noise." In some cases the process is energy detection where the process is the sum of squares of random noise n or signals plus noise s + n. Often the noise PDF is Gaussian but one could consider other PDFs like skewed Gaussian or a uniform distribution. We present a novel Raspberry Pi 5—hardware device demo where one transforms a uniform PDF into a different PDF, like a Gaussian. Here we use electronic hardware devices for (1) the PDF measurements and (2) the uniform random number generator. The Inverse Transformation Sampling Technique is performed on the Raspberry Pi.

10:30–10:45 Break

10:45

2aUW5. Model-informed acoustic glider localization in a seamount region with Gulf Stream influence. Jack Slater (Virginia Inst. of Marine Sci., 1309 Greate Rd., Gloucester Point, VA 23062, joslater@vims.edu), Donglai Gong (Virginia Inst. of Marine Sci., Gloucester Point, VA), Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), and William R. Harris (AOSE, MIT-WHOI Joint Program, Cambridge, MA)

Underwater gliders are uniquely suited for ocean acoustic research given their near silent operation and collocated sound-speed measurements. They are, however, subject to significant position uncertainty while under the surface. Building on previous efforts, we present recent work on improving localization of mobile receivers using a moored fixed source and glider-based receivers. During the New England Seamount Acoustics Experiment (NESMA), a fleet of gliders were deployed and equipped with passive acoustic hydrophone systems to study multi-path propagation and source-receiver localization in a bathymetrically complex seamount environment with Gulf Stream influence. Mid-frequency linear frequency modulated pulses were emitted for 5 min every hour from a nominal depth of 900 meters by a source moored atop the Atlantis-II seamount. The gliders carried tetrahedral or single hydrophone systems along a transect over the top of the seamount. Pulse receptions have been identified in the glider data

down to 1000 meters and out to over 50 kilometers in range. Propagation modeling between the source and the positions of the glider receivers indicate potential for effective receiver localization using relative arrival timing of surface interacting and fully refracted paths. The comparison between the modeled and observed arrival structures are utilized to refine the glider's position.

11:00

2aUW6. Implementing a time-aligned passive underwater acoustic network via Starlink. Robert T. Taylor (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, rtaylor119@utexas.edu), Hari C. Dubakula, Mikail F. Sadic (The Univ. of Texas at Austin, Austin, TX), Thomas S. Jerome, and Michael J. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Distributed networks of ocean-acoustic sensors have been limited by a triple constraint: time alignment, broad spatial coverage, and affordability, where only two of the three can simultaneously be optimized. The development of the Starlink satellite network opens a new avenue for continuous access to acoustic data in some of the world's most remote places. Additionally, the growth of commercially available solar-power technology has reduced the financial burden associated with off-the-grid energy solutions. Together, these remove the triple constraint, where a network of low-cost nodes can provide time-synchronized and affordable data with the added benefit of real-time remote access. This work presents the successful efforts to produce two individual acoustic nodes capable of continuously streaming raw timeseries data to a cloud server for remote access by users. The precise time synchronization enables coherent processing of the data, including beamforming and time-difference-of-arrival techniques. Details of the hardware and software used to construct these nodes are provided, as well as the testing and validation of the developed network. Results include a localization experiment conducted at a lake test station proving the accuracy of the system. This effort expands the practical scope of distributed, real-time, passive acoustic monitoring.

11:15

2aUW7. Low frequency bending mode excitation via modulated radiation pressure using an acoustic lens. Sterling M. Smith (Phys. & Astronomy, Washington State Univ., Phys. and Astronomy, Washington State Univerity, Pulman, WA 99164, sterling.m.smith@wsu.edu), Christopher T. Powers, Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA), Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Ahmad T. Abawi (HLS Res., San Diego, CA)

For some targets of interest in underwater acoustics, such as unexploded ordnance (UXO), the fundamental bending mode can be extremely narrow in frequency and by reciprocity, poorly coupled to low-frequency incident plane waves. To be able to fully investigate these low-frequency modes, this work shows that Modulated Radiation Pressure (MRP) offers a distinct advantage to exciting targets by offering another way of coupling acoustic energy to objects. By focusing modulated ultrasound on antinodes of the target mode with a properly selected modulation frequency, energy is coupled into the target and the mode can be driven. This is done by using a generalization of the modulated radiation force method introduced by Marston and Apfel [J. Acoust. Soc. Am. **67**, 27–37 (1980)]. Further, the focus can be swept across the target, yielding mode shape information. This effort continues prior work, stepping up from a 15 cm focal length lens to 1 m and 1.5 m. Many different LF flexural and compressional modes of different targets can be driven using the lenses, including a replica aluminum UXO. Work includes tank-based and lake-based experiments. [Work supported by the U.S. Office of Naval Research]

11:30

2aUW8. Horizon of the Acoustic Far-field. Alexey Titovich (Carderock Div., Naval Undersea Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700, alexey.titovich@navy.mil)

In the acoustic far-field, sound radiated or scattered by an object exhibits predictable geometric spreading loss and an angular variation independent of range. While the transition from near-field to far-field is gradual and lacks a universally agreed-upon boundary, its effective distance generally increases linearly with frequency. Likewise, the acoustic attenuation in the ocean (primarily due to boric acid and magnesium sulfate relaxation) increases quadratically with frequency imposing a formidable limit. Given a fixed aperture and maximum achievable source level, a frequency exists beyond which the far-field is practically unobservable due to signal attenuation. This defines an acoustic far-field horizon. This presentation will derive a closed-form approximation for this limiting frequency and its corresponding range, offering a theoretical framework for understanding this phenomenon.

11:45

2aUW9. A rapid simulation tool for end-to-end underwater acoustic measurements. Zachary Jensen (Dept. of Defense, 300 Hwy. 361, Crane, IN 47522, zachary.r.jensen3.civ@us.navy.mil)

Characterizing underwater acoustic transducers often involves navigating trade-offs between signal-to-noise ratio, far-field conditions, and physical constraints like tank size, depth, and fixture geometry. This presentation introduces a lightweight simulation tool developed to enable engineers and researchers to rapidly model and optimize entire measurement systems. This physics-based virtual test bed models the complete signal chain of interconnected hardware components. Users can define parameters for each stage of the signal chain, including: signal (tonal, noise, chirp waveforms), DACs (bit depth, voltage range), amplifiers (Gain, noise, THD), transducers (impedance, TVR, RVS), acoustic environment (transducer locations, tank geometry, first-order reflections), receive electronics (hardware filters, pre-amplifiers), ADCs (quantization, bit error rate). The framework provides powerful, multi-faceted visualizations to support intuitive understanding, including a 3D plot of the experimental geometry and reflection paths, a multi-stage graphic showing the signal's evolution through the entire chain, and a gated FFT analysis of the acquired digital data. Future work will focus on transducer directional characteristics and non-linear effects. In scenarios where-back-of-the-envelope calculations are not sufficient and detailed computational models are unnecessary—this tool fills the gap by offering fast, practical insights and reducing costly trial-and-error during setup.

Session 2pAAa

Architectural Acoustics, Noise and ASA Committee on Standards: Acoustics in Public Accommodations Spaces Part II

Kenneth Good, Cochair
Architecture Acoust., Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17601

Keely M. Siebein, Cochair
Siebein Acoust., 625 NW 60th St., Ste. C, Gainesville, FL 32607

Samuel H. Underwood, Cochair
Univ. of Nebraska-Lincoln, 1110 S 67th St., Omaha, NE 68182-0816

Chair's Introduction—1:00

Invited Papers

1:05

2pAAa1. Influence of cantilevered and sloped façade geometries on urban noise propagation: A case study of the taipei performing arts center. Veerin Udomsopakit (Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd. Sec. 4, Taipei 10607, Taiwan, d11313804@mail.ntust.edu.tw), Lucky S. Tsaih, Finka Soelistyo, and Yosua W. Tedja (Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan)

The Taipei Performing Arts Center is characterized by a radical geometric composition where three theater volumes, the spherical Globe Playhouse, the asymmetrical sloped Grand Theater, and the flat Experimental Theater, that protrude from a central cube. While architecturally distinct, the impact of such non-orthogonal surfaces on urban noise propagation remains understudied. This research quantifies the effects of these suspended, irregular geometries on external Sound Pressure Level (SPL) distribution by comparing the existing design against a simplified orthogonal control model using ray-tracing simulations. The first analysis focused on the Grand Theater's inclined undersurface. Results indicate that this sloped cantilever acts as a reflector, directing noise toward upper-level adjacent buildings. Substituting the slope with a flat surface parallel to the ground yielded SPL reductions of 4.2 dB at street level (+1.6m) and up to 9.9 dB at the 12th floor (+36m). The second analysis assessed spatial noise distribution. The existing complex geometry generated broader high-SPL zones (58–60 dB) across the site. Conversely, the orthogonal prototype confined high-intensity noise to the roadway, significantly increasing the area of acoustic shelter ($\leq$55 dB). These findings demonstrate that façade inclination is a critical determinant in urban acoustics, necessitating the optimization of complex forms to mitigate vertical and horizontal noise spread.

1:25

2pAAa2. A Conceptual framework for exploring the effect of soundscape perception on place attachment in university dormitories. Semiha Yilmazer (Architecture, Hasan Kalyoncu Univ., Hasan Kalyoncu University, Dept. of Architecture, Gaziantep 27000, Turkey, semiha@bilkent.edu.tr), Zeynep Uğurlu, Ceren Şahmaran, Oya Yıldız, and Cem Kaan Şen (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

The soundscape approach is not concerned with the sound level of an acoustic environment, but how it relates to human perception, how it is conceived by the individual, and how it should be measured and managed. Based on soundscape perception on place attachment, identifying the relationships between sound and the built environment can provide valuable information for decision-makers to promote sensation, satisfaction, and positive emotions. This study examines how soundscape perception varies across different spatial conditions in dormitories and how these perceptions relate to residents' sense of place attachment. The research is conducted at Bilkent University, where dormitories are situated in five diverse perceived acoustic environments. This study presents the qualitative analysis based on the semi-structured interviews conducted with 45 participants to develop a conceptual framework. The findings indicate that residents do not merely respond to different sound environments but actively interpret soundscapes in relation to daily routines, expectations, and residential meanings. While some sound sources are described as disruptive to daily activities, others contribute to liveliness, familiarity, or a sense of social presence, depending on spatial configuration and individual interpretation.

1:45

2pAAa3. Assessing room acoustics in transportation hubs for passenger comfort and public address system intelligibility. Peter Holst (Resonance Acoust., San Francisco, CA) and Spencer Zack (Resonance Acoust., 364 Bush St., Fl. 2, San Francisco, CA 94104, szack@resonanceac.com)

The team at Resonance Acoustics is currently collaborating on the design effort for a refresh project at the LAX Tom Bradley International Terminal. Drawing on our recent acoustical design experience for other major renovations of airport terminals, concourses, and train station public address systems, we will present strategies for analyzing room acoustics and public address system design using 3D modeling software (e.g., Odeon, EASE, CATT) for large-volume environments that can be unwieldy. In addition to assessing paging speech intelligibility, we will outline acoustical design strategies for "high-stress" passenger path-of-travel areas as well as other hospitality/retail areas found in modern airports.

2:00

2pAAa4. Evaluating the effects of externalization and reproduction methods on speech intelligibility. Paula Sedlacek (Acoust., The Penn State Univ., 323 Eng. Collaborative Res. and Education Bldg., University Park, PA 16802, pbs5654@psu.edu), Navin Viswanathan (Commun. Sci. and Disorder, The Penn State Univ., University Park, PA), Eugene J. Cainsworth (GN Adv. Sci., Minneapolis, MN), and Michelle C. Vigeant (Acoust., The Penn State Univ., University Park, PA)

Listening situations with multiple talkers impose an increased cognitive load on listeners, making it difficult to isolate the target voice. Four reproduction methods were reviewed to evaluate their effect on speech intelligibility in a multi-talker scenario: mono headphone, stereo headphone, externalized stereo headphone, and spatial loudspeaker reproduction. For the externalized stereo, the algorithm employed real-time convolution of the signal with a non-individualized head-related transfer function based on head-tracking. Speech stimuli of the format "Ready [call sign] go to [color] [number] now" was spoken by all four talkers simultaneously and each listener was asked to isolate the call sign "Charlie" and identify the corresponding color and number. This scenario was created using four talkers, two male and two female, that were spatially placed at $+/-72$ and $+/-36$ deg along the azimuth plane where 0 deg is directly in front of the listener. Headtracking, heart rate and skin conductance data were collected to estimate listening effort. Talker placement and order of reproduction methods were randomized across participants. Our results will evaluate whether and how externalization and reproduction methods affect listening performance and effort. [Work supported by a gift from GN Advanced Science and by the NSF Scholarships (S-STEM) under Grant No. 2130169.]

2:15–2:30 Break

2:30

2pAAa5. Research and development of an affordable, wearable, sound-level-meter to notify users to promote hearing health awareness. Natalie Ramos (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, 200 Bloomfield Ave. West Hartford, CT 06117, naramos@hartford.edu) and Heui Young Park (Mech., Aerosp., and Acoust. Eng., Univ. of Hartford, West Hartford, CT)

The importance of hearing health is not usually discussed as often as it should be, with many individuals not even aware of the detrimental, irreversible effects of hearing loss. To help demonstrate the importance of hearing health in every day applications, an affordable, wearable sound level meter has been developed that can notify people of exposure to dangerously loud environments. This presentation will describe the prototyping and design considerations of the wearable device, using commercially available hardware and software. Preliminary results for a prototype will also be presented. The wearable sound level meter could also be easily modified for other undergraduate, hands-on projects. Utilizing this device in specific environments such as construction sites, power plants, concert venues, sports arenas, and school dormitories and facilities, can be helpful in reducing hearing loss from long-term exposure. [Work supported by The Women's Advancement Initiative's Dorothy Goodwin Project Grantee Program, University of Hartford.]

2:45

2pAAa6. Measurement-informed orientation of an amphitheater surrounded by natural rock formations using in-situ impulse response analysis. Joseph Morris (Technol., Resolut Group, 181 East 5600 S, 200, Murray, UT 84123, jmorris@resolutgroup.com)

Outdoor amphitheaters are often architecturally oriented to emphasize surrounding scenery; however, acoustic performance ultimately governs audience experience and venue success. This case study presents an iterative design methodology for amphitheater orientation within a complex natural landscape characterized by multiple adjacent rock formations, where early design decisions were guided by in-situ acoustical testing. Impulse responses were measured on site using multiple representative stage source locations. Reflection timing was evaluated using qualitative energy-time curve comparisons to assess the influence of nearby terrain features on early and delayed reflections. Comparative analysis across candidate orientations demonstrated that proximity and alignment with a dominant rock formation significantly altered reflected energy arriving within perceptually relevant time windows. An orientation minimizing strong delayed terrain-induced reflections was identified, supporting improved audible clarity and reduced reflection interference. Quantitative findings were subsequently communicated to non-expert stakeholders through informal on-site listening demonstrations, which served as perceptual validation of the acoustically informed design recommendation and facilitated stakeholder acceptance. This study demonstrates the necessity of field-specific acoustic measurement in natural amphitheater design and proposes a transferable framework for orientation selection in terrain-driven acoustic environments.

3:00

2pAAa7. Acoustic monitoring and noise source characterization in a hospital ward. Jadwiga E. Hyla (Mech. and Vibroacoustics, AGH Univ. of Krakow, Mickiewicza av. 30, AGH Bldg. D1, Kraków 30-059, Poland, hylajadzia@gmail.com), Tadeusz Wszołek, Marcjanna Czapla, and Szymon Ruchniewicz (Mech. and Vibroacoustics, AGH Univ. of Krakow, Kraków, Poland)

Hospital environmental noise, generated by medical equipment, technological devices, and staff communication, is a significant factor affecting patient recovery. Prolonged exposure to excessive noise levels may cause sleep disturbances and impair postoperative recovery. As part of the study, continuous noise monitoring was conducted in a hospital in Kraków. The collected data were subjected to detailed analysis, and basic acoustic parameters relevant for the assessment of noise annoyance in a hospital environment were determined. Emphasis was placed on the evaluation of statistical noise levels characterizing noise dynamics, e.g., equivalent or maximum levels, which were analyzed separately for each hour of the day. Additionally, acoustic measurements were performed in selected hospital rooms to identify crucial sound sources located near patients. A literature review was also carried out, focusing on factors contributing to patient discomfort, survey methodologies applied in similar studies, and approaches to noise measurement in medical facilities. Further analysis will involve identifying periodic acoustic events and linking them to specific medical procedures or activities. Patient surveys will be conducted in cooperation with hospital staff, and correlating these results with acoustic data is expected to provide a more accurate identification of the most critical noise sources.

3:15–3:35 Discussion

Session 2pAAb

Architectural Acoustics and Noise: Advances in Acoustics of Mass Timber Assemblies

Evelyn Way, Cochair
Maxxon Corp, 920 Hamel Rd., Hamel, MN 55340

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

Invited Papers

1:00

2pAAb1. The oregon acoustic research laboratory: Design of a new test lab for mass timber. Michael Raley (Univ. of Oregon, 2000 4th Ave. Canby, OR 97013, mraley@pac-intl.com) and Mark Fretz (Architecture, Univ. of Oregon, Portland, OR)

For many years, the TallWood Design Institute, a collaboration between the University of Oregon and Oregon State University, has played a key role in advancing mass timber in Oregon, the Pacific Northwest, and beyond. A primary contribution of TallWood has been the testing and research necessary to develop new technologies and to establish mass timber's compliance with existing code requirements. As TallWood sought to add acoustical testing to its portfolio, it encountered numerous challenges, including the paucity of nearby acoustical test labs and the inability of many labs to handle large mass timber panels. Thus began the initiative to build an acoustical test lab in Oregon. After many years of work, the University of Oregon has secured a substantial federal grant through the Economic Development Administration (EDA), along with additional funds from state and university sources, to build the Oregon Acoustic Research Laboratory (OARL). The lab was designed to provide high STC and IIC flanking limits, quick throughput for testing assemblies, and the ability to handle mass timber and other large panels. The project broke ground in December of 2025 and is slated for completion in Spring of 2027. This presentation will cover the impetus for OARL, its place in the broader mass timber industry, its design, and the goals for its operation. It will also discuss the unique perspective of an acoustician occupying the role of the end user on a large construction project where acoustical performance is critical.

1:20

2pAAb2. Mass timber considerations for K-12 school buildings. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Blake Krapfl (DLR Group, Minneapolis, MN)

This presentation will review two case studies where mass timber construction was used in K-12 education facilities. The first case is a high school technical education building with hybrid mass and steel timber construction, with acoustical challenges of very large and high bay spaces for automotive training and building trades training. The second case is an elementary school where the design team was directed to expose as much of the timber structure as possible while also achieving CHPS Verification including acoustic performance credits.

1:40

2pAAb3. Field evaluation of acoustic performance in mass timber school buildings. Denis Blount (LSTN Consultants, 76 Beaver St., 2nd Fl., New York, WA 10005, denis@lstnconsultants.com)

The use of mass-timber structural systems in educational facilities is increasing due to their potential to reduce embodied carbon; however, acoustic field performance remains less well documented. This study presents field measurements from three recently completed mass-timber schools in the Pacific Northwest, evaluating airborne and impact sound insulation, room acoustics, and vibration relative to the requirements of ANSI S12.60 for classroom acoustics. Typical floor assemblies consisted of cross-laminated timber (CLT) or mass plywood panels (MPP) with acoustic underlayments and mass toppings of either gypsum-based concrete or conventional concrete. In-situ measurements of apparent sound transmission class (ASTC), apparent impact insulation class (AIIC), and reverberation time were conducted using standardized field methods and compared with design targets. Preliminary findings are: 1. CLT ceilings in classrooms can be fully exposed with adequate sound absorbing wall paneling. ie no sound absorbing ceiling finishes ceiling required for 0.6s RT. 2. Flanking transmission paths significantly influence both ASTC and AIIC performance. Ie Design for flanking at the perimeter of sound isolating assemblies is critical. 3. The selection and detailing of impact isolation underlayments is critical to achieving acceptable impact noise control. 4. Pending further study of the results, opportunities to further optimize mass-timber assemblies for cost, constructability, and embodied carbon while maintaining acoustic quality may exist.

2:00

2pAAb4. The intersection of structural and acoustic design in mass timber construction: Quantifying tradeoffs and optimizing performance. Simon Cleghorn (Pliteq Inc., 4211 Yonge St., Toronto, ON M2P 2A9, Canada, scleghorn@pliteq.com) and Aedan Callaghan (Pliteq Inc., Toronto, ON, Canada)

In mass timber construction, it is understood that early multidisciplinary design and trade coordination is key to delivering a successful project. The intersection of acoustic and structural design is one area that continues to present discussions on mass timber projects, as there exist several design decisions which have simultaneous and contradictory impacts on acoustic and structural performance. Thus, it is important to quantify the benefits and harms to both areas of performance, such that the correct design is selected to balance the goals of the project at hand. This work will briefly explore topics that exist at the intersection of structural and acoustic design. The topics investigated include the effect of deadload on flanking transmission, the impact of fastener spacing on both acoustic performance and vibration controlled span length, continuous versus simply supported spans in ceiling-ceiling flanking transmission, structural design efficiencies accessible through the use of dry acoustic floor-ceiling systems, and floor vibration data in the presence of an acoustic interlayer.

2:20

2pAAb5. Structural deflection and acoustic isolation performance in mass timber assemblies. Evelyn Way (Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, eway@maxxon.com) and Jeffrey Olson (Boise Cascade, White City, OR)

As mass timber becomes a more widely used structural system, there is a natural incentive to find the most cost-effective systems possible. Lab testing on thin mass timber panels (3-ply or 3-ply equivalents) with exposed wood ceilings have proven to be even more challenging to achieve comfortable levels of acoustic isolation than 5-ply or 7-ply assemblies, particularly where codes apply to multifamily housing. While thicker topping slabs and/or thicker sound mats are a possible solution, another variable, the deflection of the panel, is explored by comparing mass timber panels of different grades but the same thickness. Higher grade lumber or veneers results in stiffer panels, lower deflections, and higher acoustic isolation performance.

2:40

2pAAb6. Influence of boundary conditions on the acoustic radiated power of cross-laminated timber floors. Henrique T. Lima (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., 1515 Saint-Catherine St W, Montreal, QC H3G 2W1, Canada, henrique.lima@mail.concordia.ca), Joonhee Lee, and Ghazanfarah Hafeez (Dept. of Bldg., Civil and Environ. Eng., Concordia Univ., Montreal, QC, Canada)

Boundary conditions play a critical role in governing the vibration and acoustic performance of cross-laminated timber (CLT) floors, yet their representation remains a major source of uncertainty in both design and simulation. This study investigates the influence of different floor support configurations on the vibroacoustic response of a CLT floor panel. A 3 m $\times$ 3 m CLT floor specimen is excited using a standardized tapping machine, and the acoustic radiated power is subsequently estimated from measured vibration data using an integral transformation method. Three simply supported boundary conditions are examined: (1) support on all four edges, (2) support on the two edges along the major direction with the remaining edges free, and (3) support on the two major-direction edges plus one minor-direction edge. The results show that variations in the effective support layout lead to significant differences in low-frequency acoustic behavior, with the lowest modal frequencies observed at 23.3, 29.5, and 32.6 Hz for the three boundary conditions, respectively. These findings highlight the sensitivity of CLT floor acoustic behaviors to boundary condition assumptions and underscore the need for careful consideration of support modeling in both experimental testing and predictive simulations.

Session 2pABa

Animal Bioacoustics: Animal Vocal Communication and Physiology

Xavier Mouy, Chair
Applied Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543

Contributed Papers

1:00

2pABa1. Who said the dogs howled? Pilot mathematical models for standardizing laryngeal vibratory mechanism assessment via enhanced acoustical estimation of glottal quotient. Charles R. Armstrong (Scinguistics, 1312 Guilford Ave., Baltimore, MD 21202, crabiolinguistics@gmail.com), Hypatia von Sva (Scinguistics, Berlin, Germany), Daniel Pisani (Scinguistics, Sudbury, ON, Canada), Emilia Blåsten (Mathematics, Lappeenranta-Lahti Univ. of Technol., Lahti, Finland), and Jojo Bertemes (Scinguistics, Luxembourg, Luxembourg)

LVMs (laryngeal vibratory mechanisms; M0–M3) hold profoundly understudied, but putatively communicative use among chordates. Current indirect inferences of LVM in literature rely either on subjective ascription of emotion and/or on relative pitch/amplitude shifts rather than objective, isolable measures. Glottal quotient, mainly close/open here, yields possibly the most straightforward and objective quantitative analyses of mechanism-correlable vocal quality. From 0–1, **OQ**. From M0–M3, OQ tends to increase with mechanism number. However, invasive EGG (electroglottographic) and videographic measures run the risk of altering the behaviors they measure. We propose means for more robust, minimally-invasive harmonic estimation of glottal quotient, tested against EGG. We assess novel formulae for acoustically estimated OQ (**CQ** (close quotient) measure the proportion of time that the glottal cycle spends in relative open motion phase (OQ) versus at their point of maximal closure (CQ). This can be depicted as: $\mathbf{OQ} + \mathbf{CQ} = \mathbf{1}$. From M0-M3, OQ tends to increase with mechanism number. However, invasive EGG (electroglottographic) and videographic measures run the risk of altering the behaviors they measure. We propose means for more robust, minimally-invasive harmonic estimation of glottal quotient, tested against EGG. We assess novel formulae for acoustically estimated OQ ($\boldsymbol{OQ}_z$), as well as regression functions ($\mathbf{f}(\boldsymbol{OQ}_z)$) to convert this estimate to corresponding dEGG-measured OQ values ($\boldsymbol{OQ}_\sigma$). The use of these estimations may preserve the verisimilitude of LVM behaviors observed, as well as disambiguate vocal observations spuriously attributed to LVMs, such as sex-linked OQ differences in humans.

1:15

2pABa2. A comparison of the squeaking vocalizations of North American wild canids. Jennifer N. Schneider (Dept. of Psych., LCC Int. Univ., Kretingos 36, Klaipeda LT-92307, Lithuania, jschneider@lcc.lt) and Dustin Ranglack (Utah Res. Station, USDA APHIS WS, National Wildlife Res. Ctr., Millville, UT)

The affiliative squeaking vocalization, consisting of sequences of low-amplitude, high-frequency tonal units, has not been systematically described or studied in coyotes (*Canis latrans*), despite their close relationship with wolves (*Canis lupus* and *Canis rufus*). To compare the structure of coyote squeaks to those of wolves, we observed three breeding pairs at the USDA Predator Research Station in Millville, Utah using camera traps and continuous audio recordings from five recorders. Like wolves, the number of squeaks per vocalization varied greatly. The frequencies bandwidth within which coyotes squeaked (3.0–11.0 kHz) was broad, more similar to red wolves (2.7–9.5 kHz) than gray wolves (1.8–6.0 kHz). Individual squeaks more closely resembled those of gray wolves, with larger average frequency ranges and longer durations than those of red wolves. Squeaks of all three species display a variety of contours, with flat or bump-like shapes being among the most common. Like red wolves, coyotes produced sound units with a second fundamental frequency (0.5–1.6 kHz), either in the form of frequency jumps or biphonations. The low-frequency component did not travel as far as the high-frequency component, which may affect eavesdropping conspecifics. North American wild canids share a common affiliative vocalization, despite differences in social structure complexity.

1:30

2pABa3. Sociality as a driver for auditory sensitivity in rodent species. Luberson Joseph (Biology, Oklahoma State Univ., Stillwater, OK 74078) and Elizabeth McCullagh (Biology, Oklahoma State Univ., 2106 Copperfield, Stillwater, OK, elizabeth.mccullagh@okstate.edu)

Sociality has evolved repeatedly in rodents, but its influence on auditory processing remains unclear. We examined auditory brainstem responses (ABRs) in nine wild-caught rodent species spanning diverse evolutionary lineages to test whether sociality predicts auditory sensitivity. Phylogenetic comparative analyses revealed that social species—both monogamous and group-living—exhibited lower auditory thresholds for tone and click stimuli compared to solitary species. Group-living species also showed faster normalized latencies and greater relative amplitudes of the binaural interaction component (BIC) across interaural time differences (ITDs), suggesting enhanced binaural processing. In contrast, monaural ABR measures (interpeak latencies and amplitude ratios of waves I and IV) did not differ among social groups across increasing sound intensities. These findings indicate that sociality is associated with heightened auditory sensitivity and improved binaural integration in rodents, supporting the hypothesis that the evolution of complex social behavior may favor auditory systems optimized for effective communication.

Session 2pABb

Animal Bioacoustics: General Topics in Animal Bioacoustics

Xavier Mouy, Chair
Applied Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543

Contributed Papers

2:45

2pABb1. Pseudo-spectral model of 3-D elastic-wave propagation through toothed-whale head anatomy, and implications for biosonar. Fawad Ali, Carlos García A. (Dept. of Geosciences, Univ. of Padua, Padua, Italy), Aida Hejazi-Nooghabi (Geophys., Hamburg Univ., Hamburg, Germany), and Lapo Boschi (Dept. of Geosciences, Univ. of Padua, via Gradenigo 6, Geosciences Dept., Padova 35131, Italy, lapo.boschi@unipd.it)

The sound localization and biosonar system of toothed whales is exceptionally performant. How such performance is achieved is unclear, given that (i) toothed whales have no pinnae, and (ii) although their auditory pathways have been studied in detail, no specific feature that could functionally replace the pinna has been identified. We employ a pseudo-spectral time domain numerical scheme to model three-dimensional elastic wave propagation through a toothed-whale head including soft tissues. Computed tomography scans were used to build a velocity-density model of a bottlenose dolphin's head, parametrized on 1.11 mm voxels. We validate our wave propagation solver, identifying a range of frequencies and scale lengths where it adequately captures the complexities of wave propagation through anatomy. We next focus on the toothed whale's ability to determine the elevation of sound sources, where anatomy plays a crucial role. Sinusoidal bursts with 45 kHz central frequency, emitted by far-field sources at elevations from $-90°$ to $+90°$, were modeled at the locations of left and right inner ear. We find that their elevation can be established, via correlation, solely based on the "coda" of the incoming signal, whose waveform is controlled by refraction through and reflection off multiple anatomical structures.

3:00

2pABb2. Array geometry evaluation and optimization for bioacoustic localization. Akash Ram, Arham Bafna, Rucha Parekh (Dwarkadas J. Sanghvi College of Eng., Mumbai, India), Eric Snyder, Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), and Irina Tolkova (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, ivt2@cornell.edu)

Across marine and terrestrial bioacoustics, time-difference-of-arrival localization of animal vocalizations is valuable for studying spatiotemporal distributions, estimating abundance, and tracking individuals. However, successful localization relies on choosing a suitable array geometry and spacing—both creating a trade-off between detectability and spatial coverage. In this work, we draw on concepts from distance sampling to formulate acoustic array design as a non-convex optimization problem, maximizing the probability of detection of a call across multiple units throughout the study area. In addition to evaluating common array geometries with this metric, we calculate genetic-algorithm-based and gradient-descent-based solutions and compare performance. Next, we demonstrate this framework for two case studies, considering a hydrophone deployment for baleen whale calls and a microphone deployment for birdsong. Finally, we provide an accessible web-based application for user-defined array evaluation and optimization.

3:15

2pABb3. Analyzing echolocating bat interactions with mist nets using acoustics. Amaro Tuninetti (Dept. of Biological Sci., Univ. of New Hampshire, Brown University, Box 1821, Providence, RI 02912-9067, amaro.tuninetti@unh.edu), Emily-Kate Molnár (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH), Devaughn Fraser (Connecticut Dept. of Energy and Environ. Protection, Hartford, CT), and Laura Kloepper (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

Trapping tools such as mist nets have become standard tools worldwide for a wide range of bat research, surveying, and monitoring efforts. These tools require setup by experts and exploit limitations of bat echolocation by incorporating elements too thin to return perceivable echoes to approaching bats. However, there is a lack of work investigating how bats actively interact with these tools and how their echolocation behavior changes. Here, we report preliminary results from two nights of mist netting in southern Connecticut, where a high-speed thermal camera and an array of ultrasonic microphones were used to monitor bat behavior over multiple hours of mist netting. At least three bat species were recorded acoustically (*Myotis leibii, Myotis lucifugus*, and *Eptesicus fuscus*). We observed a high number of circular, non-foraging flights in the immediate proximity of the mist net, suggesting an active avoidance of the net by some individuals. We did not observe any distress calls and very few social calls emitted by recently-caught individuals, nor did we observe significant changes to echolocation calls immediately prior to net entrances. Further data collection is required to compare capture rates across different species and contexts and disentangle the existence of any species-specific differences.

3:30

2pABb4. Echolocation detection probability of *Ziphius cavirostris* estimated using bioacoustic tag data and Monte Carlo simulation. Kacy Hamilton (La Costa Canyon High School, La Costa Canyon High School, Maverick Way, Carlsbad, CA 92009, kacyhamilton1@gmail.com), Ainsleigh Lloyd, Joshua Jones, and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Density estimation from passive acoustic monitoring depends critically on accurate estimates of detection probability. This challenge is especially acute for deep-diving beaked whales, which produce highly directional echolocation signals. We use high temporal resolution bioacoustic tag data to quantify how diving depth, body orientation, speed and click production rates jointly influence acoustic detectability during foraging dives of goose-beaked whales (*Ziphius cavirostris*) in the Southern California Bight. A Monte Carlo simulation was used to estimate the probability of detecting individual clicks, as well as the probability of detecting one or more clicks within 1-min time bins for groups of echolocating animals. The simulation computes expected received levels by combining tag-derived dive behavior and orientation with acoustic propagation modeling. Animal positions relative to fixed hydrophones were sampled from a uniform spatial distribution consistent with observation sites in the Southern California Bight. These results provide a quantitative framework for translating tag-derived click

records into detection probability estimates, improving the interpretation of passive acoustic data and enabling more robust density estimation for *Z. cavirostris*.

3:45

2pABb5. Preliminary evidence for rocket noise impacts on California least tern vocalizations. Megan R. McCullah-Boozer (Biology, California State Univ. Bakersfield, SCI I, 205, 9001 Stockdale Hwy., Bakersfield, CA 93311, mmccullah@csub.edu), Rachel H. Budge (Biology, California State Univ. Bakersfield, Bakersfield, CA), Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Emily Olivares Garnica, Hugo J. Zepeda (Biology, California State Univ. Bakersfield, Bakersfield, CA), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (Biology, California State Univ. Bakersfield, Bakersfield, CA), Dan Robinette, and Emily Rice (Vandenberg Field Station, Point Blue Conservation Sci., Lompoc, CA)

Rocket launches generate extreme acoustic energy that can temporarily overlap frequency bands used for animal communication. We used passive acoustic monitoring to ask whether rocket noise impacts vocalizations of the endangered California Least Tern (*Sternula antillarum browni*) at Vandenberg Space Force Base, California. Eight autonomous recorders continuously sampled during the 2024 breeding season. For 15 launch events, we quantified changes in call rate, peak frequency, peak amplitude, and average amplitude within 24-h windows before and after each launch. Initial paired pre- and post-launch comparisons revealed variability in both magnitude and direction of response variables across events, indicating that colony-level averages obscure launch-specific responses. To address this, we used mixed-effects modeling that incorporated recorder location, breeding phenology, peak overpressure of launch events, and meteorological conditions. Preliminary models indicate statistically detectable post-launch shifts in vocal signal structure, including modest increases in peak frequency and reductions in peak amplitude. Effect size and direction varied across reproductive phases and recording locations, with device-level contrasts revealing opposing frequency shifts within the colony. These findings suggest that rocket noise is associated with short-term, spatially heterogeneous modifications to tern vocalizations.

4:00

2pABb6. Surveys of endangered odontocete whales using acoustic gliders. David K. Mellinger (Coop. Inst. for Marine Ecosystems & Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Selene Fregosi (Marine Mammal Inst., Oregon State Univ., Newport, OR), Jace Marquardt (College of Earth, Ocean, and Atmospheric Sci., Oregon State Univ., Corvallis, OR), Erik Norris, and Erin Oleson (Pacific Islands Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Honolulu, HI)

Since 2022 we have used ocean gliders to conduct a series of passive acoustic surveys around the Main Hawaiian Islands (MHI) for false killer whales (FKW; *Pseudorca crassidens*). This is a species of odontocete (toothed whale) of high conservation concern both because the MHI population is endangered and also because it can become caught and drowned in fishing nets. Seagliders™, a type of deep-water ocean glider, were equipped with hydrophones and recording systems and flown between Kauai and the Island of Hawaii on zigzag tracks that brought them from near shore to over 100 km offshore. Ambient sound was recorded internally whenever the glider descended below a pre-set depth, avoiding noisy conditions at the surface, and continued down to the bottom of each dive, typically 1000 m deep, and then back up again. After recovery of the gliders at the end of their missions of up to 60 days, recordings were analyzed for the presence of false killer whales and other odontocetes. FKWs were detected on many encounters both north and south of the Hawaiian island chain on the 2022 and 2023 surveys, and results from a winter 2026 survey may be available as well. [Funding from NOAA and ONR.]

TUESDAY AFTERNOON, 12 MAY 2026 SALON A, 1:00 P.M. TO 4:15 P.M.

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Ocean Acoustic Propagation Modeling

Brian T. Hefner, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Nicholas C. Durofchalk, Cochair
Phys., Naval Postgraduate School, 833 Dyer Road, Spanagel Hall, Rm. 203, Monterey, CA 93943

Contributed Papers

1:00

2pAO1. Long-range ocean acoustic propagation across the Pacific Basin. Nicholas C. Durofchalk (Phys., Naval Postgrad. School, 833 Dyer Rd., Spanagel Hall, Rm. 203, Monterey, CA 93943, nicholas.durofchalk@nps.edu), David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA), and Kay L. Gemba (Naval Postgrad. School, Monterey, CA)

Long distance (over 2000 nmi) undersea acoustic propagation is relevant to tactical oceanography. This presentation provides an overview of ongoing efforts using repeated transmissions from the Kauai Beacon (KB) source, received at a growing network of sites including Wake Island (CTBTO), Monterey (MARS), Washington (OOI), British Columbia (ONC), and Hawaii (ALOHA). Received signals are match filtered to estimate channel impulse responses (CIRs) over this 3-year data set. Propagation modeling using ray tracing and parabolic equation (PE) methods is used to contextualize observed CIR variability. In addition, we explore how internal wave perturbations may influence travel-time fluctuations. These efforts support the long-term goal of interpreting acoustic variability in terms of dynamic ocean

processes and evaluating the potential of implementing acoustic remote sensing across basin scales into Navy models.

1:15

2pAO2. Depth-dependent Gulf Stream influence on acoustic ray and mode arrivals in the SOFAR channel. Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093, ytlin@ucsd.edu), Yongsung Park (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Arthur E. Newhall, Jennifer J. Johnson (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Matthew Dzieciuch (SIO, UCSD, La Jolla, CA)

During the New England Seamount Acoustic Experiment (NESMA), a 32-channel hydrophone vertical line array was deployed in the SOFAR channel on the top of a seamount for 6 months to receive acoustic signals transmitted from sound sources located 74 and 155 km away. The scientific objective was to investigate the influence of Gulf Stream on long-range deep-water acoustic propagation with the ultimate goal connecting acoustic variability with underlying physical oceanographic processes. Two distinct arrival types can be observed in the acoustic data: steep-angle ray arrivals traveling through subsurface and deep-water layers, and shallow-angle normal mode arrivals trapped close to the SOFAR channel axis. Vertical line array beamforming is applied to analyze ray arrivals, while normal mode filtering is used to extract modal arrivals. Comparative analysis of travel time, arrival patterns, and amplitude variability is performed to contrast responses of ray and mode arrivals due to changes of water column stratification caused by the Gulf Stream dynamics. These contrasting responses arise from the depth-dependent Gulf Stream perturbations to sound speed profiles. By separating ray-based and mode-based acoustic variability, this work can demonstrate a pathway to better understand Gulf Stream variability from sound propagation measurements.

1:30

2pAO3. Data-driven modeling of broadband modal propagation for coastal oceanic waveguides. Arthur Varon (GIPSA-Lab, 11 Rue des Mathématiques, Saint-Martin-d'Hères, Isère 38400, France, arthur.varon@gipsa-lab.grenoble-inp.fr), Julien Bonnel (Scripps Inst. of Oceanogr., San DIego, CA), Jérome Mars (GIPSA-Lab, Saint-Martin-d'Hères, France), and Rémi Emmetiere (Bouches-du-Rhône, ALSEAMAR, Rousset, France)

We propose a machine learning (ML) approach based on multi-layer perceptron (MLP) to approximate normal mode parameters in shallow-water (depth < 125 m), low-frequency (50–500 Hz) underwater acoustic waveguides. Trained on diverse range-independent environments generated with varying sound speed profiles and seabed properties, the MLPs learn a mapping from environmental parameters to modal parameters. By incorporating physics-informed preprocessing, the network is able to accurately predict the horizontal wavenumbers and modal depth functions across frequencies without prior knowledge on the number of propagating modes. The proposed approach is validated on a generated test set (environments unseen during training), showing less than 1% relative error on 99% of the environment considered in the test dataset and an accurate prediction of the modal dispersion at far range even in poor-performing environments. Once trained, the ML models computes modal parameters for 100 000 environments across 50 frequencies in approximately 3 min on a single CPU core, compared to 33 min for the numerical solver KRAKEN, demonstrating an order-of-magnitude speedup in inference time. This efficiency gain enables rapid simulations for applications requiring repeated evaluations or with limited computational resources, such as onboard prediction for autonomous underwater vehicles or inversion purposes.

1:45

2pAO4. Simulations of towed source transmissions with a statistical sound speed model. William R. Harris (AOSE, MIT-WHOI Joint Program, 77 Massachusetts Ave., Cambridge, MA 02139, harriswr@mit.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), Magdalena Andres (Physical Oceanogr., Woods Hole Oceanogr. Inst., Woods Hole, MA), and William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

During the New England Seamount acoustics (NESMA) experiment in July 2024, a towed acoustic source was deployed from *R/V Roger Revelle*. Transmissions from this source were received by a three-dimensional (3-D) hydrophone array called the 3-D acoustic telescope deployed on the summit of Kelvin Seamount. The acoustic telescope mooring contained a variety of physical oceanographic (PO) sensors. However, these PO data are only available for depths exceeding 800 m so a satellite-informed statistical sound speed model is used to estimate the upper water column sound speeds. An ensemble of simulations for these towed source transmissions will be completed using the resulting full-water column sound speed profile estimates. The simulation results will then be used to determine a probabilistic arrival time for the towed source transmissions, which will be compared to the experimentally measured timing. This work will help to validate the sound speed estimation algorithm and demonstrate its usefulness in the NESMA test region. [Work supported by the Office of Naval Research.]

2:00

2pAO5. Acoustic tomography in a stratified lake with application of the matched nonreciprocity method. John C. Wells (Civil & Environ. Eng., Ritsumeikan Univ., Noji Higashi 1-1-1, Tricea I, Civil & Env'l Eng., Kusatsu, Shiga 525-8577, Japan, jwells@se.ritsumei.ac.jp), Ernst M. Uzhansky (Phys. Dept., Naval Postgrad. School, Monterey, CA), and Naokazu Taniguchi (Graduate School of Eng., Hiroshima Univ., Higashi-Hiroshima, Japan)

We consider results of an acoustic tomography experiment conducted in a stratified lake in late November 2018. Five near-bottom transducers were deployed at bottom depths ranging from 9 to 68 m in the North Basin of Lake Biwa, Japan, with horizontal transducer separations from 6.6 to 15.2 km. Over a period of 13 days, the transducers performed quasi-simultaneous reciprocal transmissions at a 5 kHz carrier frequency every 10 min, modulated by 11-bit maximum length quasi-random sequences. Observed arrival time evolution, and ray tracing simulations, support the identification of two distinct patterns in the recorded signals: an early group predominantly traversing the warm epilimnion, and a later group propagating through the colder hypolimnion. Analysis of travel time nonreciprocity focused on the two stations, separated by 6.6 km, that communicated reciprocally for the entire 13 days. After a wind-induced excitation, the early and late arrival groups yielded an anticorrelation in their respective travel time nonreciprocities. Considering the same transect, we report the first application of the matched nonreciprocity method to quantify current-induced nonreciprocity in transmission. Previously tested only with synthetic data, this method yielded plausible estimates of depth-averaged current between the two transducers.

2:15

2pAO6. Long-duration observations of the temporal coherence of scattering from a heterogeneous seafloor. Thomas E. Blanford (Univ. of New Hampshire, Durham, NH 03824, thomas.blanford@unh.edu), Jenna Hare, Anthony P. Lyons, Gabriel R. Venegas, and Hailey L. Gilman (Univ. of New Hampshire, Durham, NH 03824)

In active sensing systems, the zero-lag temporal coherence describes the similarity of signals collected at different times from the same transmitter/receiver geometry with respect to a scattering cell. In pulse-echo sonar, the temporal coherence of echoes from the seafloor can provide an acoustic means to observe changes in the seabed over time. Temporal coherence is a measurement that includes phase and is therefore sensitive to subtle long- or short-term processes that affect the seafloor. Most measurements of temporal coherence, however, have been made at shallow-water, sandy sites over durations of weeks to a month. This presentation will describe observations of temporal coherence from a site with a mixed sediment in the Gulf of

Maine at 165 m depth. Pulse-echo measurements of a patch of seafloor were made hourly for 8 months from a custom mooring. Environmental sensors on the mooring provided concurrent observation of the water column and biological processes that could influence the seabed. The presentation will describe insights from this data into how complex seafloors change over multiple timescales, from hours to months, and the oceanographic processes that may drive this change.

2:30–2:45 Break

2:45

2pAO7. Construction of random matrices for a shallow water waveguide with a rough sea-surface. Tarun K. Chandrayadula (Dept. of Ocean Eng., Indian Inst. of Technol., Madras, Chennai, Tamil Nadu 600036, India, tkchandr@iitm.ac.in)

Random-matrix simulations are computationally fast methods to simulate acoustic propagation through a random medium. The current literature for constructing such matrices uses acoustic modes, and considers only internal-wave scattering in deep water. The simulations divide the distance between the source and receiver into range-blocks, and model the signal at the receiver using a product of unitary random matrices. This paper focuses on shallow water propagation in the presence of a rough sea-surface caused by wind. The variances for the elements of the matrices use the Pierson–Moskovitz spectrum. Each range-block is now a non-unitary matrix, which is a product of a unitary random matrix for coupling, and another diagonal matrix with elements of magnitude less than 1 to model attenuation. This paper compares Monte-Carlo statistics for mean intensities using the random matrices with parabolic equation simulations. Also, this paper presents spectral statistics for products of these non-unitary matrices at different wind speeds. At low wind-speeds, the eigenvalues show up as separate points in the complex plane with magnitudes less than 1. At high wind speeds the eigenvalues form a contiguous ring, and the eigenvectors close to the Haar distribution. These statistics are similar to the "single-ring theorem" in random matrix theory.

3:00

2pAO8. Effects of the Gulf Stream on acoustic reflections atop the Atlantis II Seamount. Emily M. Wong (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92037, emw014@ucsd.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), and Tzu-Ting Chen (Inst. of Oceanogr., National Taiwan Univ., Taipei, Taiwan)

Bottom reflections and scattering of acoustic signals can provide insights into seafloor and seabed conditions. A joint field experiment was conducted in the area of the New England Seamount Chain to investigate the effects of physical oceanography, marine geology, and biology on sound propagation and scattering. Acoustic reflection measurements of single bottom bounces were collected between a sound source mooring and a four-channel hydrophone vertical line array (VLA) mooring deployed on top of the Atlantis II Seamount at bottom depths of 1734–1839 m and separated by approximately 2.5 km. Significant intensity changes on the top channel of the VLA were observed during a period of strong Gulf Stream currents passing over the moorings. An eigenray tracing model was created to estimate single bottom-bounce reflection points and characterize the spatial extent of seafloor interaction. Multibeam, sidescan, subbottom profiler, and camera data were then examined to evaluate the conditions of the seafloor and seabed around the bottom bounce points. These comparisons are used to examine how localized seafloor properties and mesoscale oceanographic variability influence bottom-reflected acoustic intensity. This work highlights the importance of jointly accounting for environmental variability, seafloor structure, and acoustic footprint when interpreting bottom-scattered arrivals in complex deep-water environments.

3:15

2pAO9. Impacts of ocean model vertical resolution on acoustic propagation predictions in a stratified ocean. Ryan M. Munion, Tetyana Margolina (Oceanogr., Naval Postgrad. School, Monterey, CA 93943, tmargoli@nps.edu), Eva D. Regnier (Operations Res., Naval Postgrad. School, Monterey, CA), Stephen C. Upton (SEED Ctr. and Operations Res., Naval Postgrad. School, Monterey, CA), Jefferson Huang, Taylor K. Haist (Operations Res., Naval Postgrad. School, Monterey, CA), and John E. Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA)

Eddies and fronts drive the vertical structure and variability of the upper ocean at meso-and submesoscales. Associated vertical distributions of sound speed in the mixed layer and pycnocline modulate acoustic energy propagation through the stratified ocean. While modern ocean models, including HYCOM and NCOM, have increasingly high horizontal resolution and fine vertical resolution within the upper hundred meters, insufficient vertical resolution of subsurface features within the pycnocline can result in unrealistic predictions of acoustic propagation. This study examined acoustic implications of ocean model limitations in reproducing the vertical structure of the South Cyprus Eddy, a semi-permanent mesoscale feature in the eastern Mediterranean. *In situ* observations were used to characterize the eddy vertical structure during several winter-spring transitions. Observed mixed layer properties and resulting acoustic structures were compared to HYCOM and three configurations of NCOM with varying horizontal and vertical resolutions. Our results suggest limited model ability to resolve the observed double-core structure of subsurface anticyclonic eddies and mixed layer depth in surface-intensified mesoscale features. These deficiencies resulted in misrepresentation of secondary sound channels and underestimation of sonic layer depth. The study has demonstrated the importance of resolving subsurface mesoscale structure for reliable prediction of acoustic propagation through the stratified ocean.

3:30

2pAO10. Multi-timescale observations of the variability of mid-frequency sound propagation on the Washington continental shelf. Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), Dajun Tang, Guangyu Xu, Ramsey R. Harcourt, John B. Mickett, Kumar Prakash (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), William Hodgkiss, and Ying-Tsong Lin (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Extensive measurements of mid-frequency sound propagation and oceanographic variability were collected on the Washington continental shelf during a month-long experiment in June 2025. A moored sound source transmitted signals with center frequencies of 4.0 and 6.0 kHz with a bandwidth of 1000 Hz to several moored receiver arrays located 500 m to 20 km in range from the source. These signals were used in alternating sequences such that each signal was transmitted every 13 s for 2 h and every 10 min for 16 h for approximately 18 days during the experiment. These measurements enable examination of the variability in mid-frequency propagation at timescales ranging continuously from tens of seconds to several weeks. Temporal and spatial oceanographic data were also collected on five dedicated oceanographic moorings, as well as using a ship-towed profiler and a shipboard echosounder. Using the combined acoustic and oceanographic measurements and a regional ocean model, the goal is to determine the degree to which various oceanographic processes impact mid-frequency propagation. This talk presents a preliminary analysis of the temporal variability of the sound propagation over the course of the experiment. [Work supported by the Office of Naval Research under Task Force Ocean.]

3:45

2pAO11. Non-diffraction of multiplexed parametric array vortex beams for long-distance underwater communication. Jeff J. Bell (Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jjb7481@psu.edu), Jiaxin Zhong (Graduate Program in Acoust., Penn State Univ., State College, PA 16802), and Yun Jing (Acoust., Penn State Univ., State College, PA 16802)

The increasing use of Unmanned Underwater Vehicles (UUVs) for applications ranging from seafloor exploration to environmental

surveillance to national security motivates innovative methods for secure and long-distance communications. Previous work has demonstrated the feasibility of combining the parametric array phenomenon with encoding digital information within the orbital angular momentum of acoustic vortex beams to achieve both goals, at least over short ranges. A primary challenge to extending the range is the dependence of the diffraction angle of particular topological charges (TCs) on the magnitude of said TC. We present methods to reduce this variance in diffraction angle, in particular the incorporation of quasi-Bessel beams. Utilizing the split-step approach of solving the Westervelt equation, simulations of both ideal and realistic phased arrays were conducted, demonstrating marked improvement in the detection of higher-order TCs at greater distances, and thus a reduced bit error rate, due to the incorporation of quasi-Bessel beams. We also present the effects of the number of array elements, order of the TC, array arrangement among other parameters on the upper limit of the range of the non-diffraction parametric vortex beam.

4:00

2pAO12. Pilot aided channel characterization for high frequency underwater acoustic communication links. Uzma Khan (Elec. and Comput. Eng., Lehigh Univ., 19 Memorial Dr. West, Bethlehem, PA 18015, uzk222@lehigh.edu), Xiyuan Zhu, and Yahong R. Zheng (Elec. and Comput. Eng., Lehigh Univ., Bethlehem, PA)

Ocean experiment data are analyzed to characterize high frequency underwater acoustic communication channels using pilot aided signal processing. A 25 channel acoustic transmitter broadcast identical single carrier waveforms consisting of binary phase shift keying pilot symbols followed by quadrature phase shift keying payload symbols, while a four element receiver array collected data at ranges between 100 and 300 m in a coastal ocean environment. Channel impulse responses are estimated from pilot symbols using least squares methods to obtain time varying representations of multipath structure and phase dynamics at carrier frequencies of 115 and 200 kHz. Experimental results reveal pronounced frequency dependent differences in channel stability, with significantly increased temporal phase fluctuations observed at the higher carrier frequency. The effective coherence time is found to decrease with increasing carrier frequency and propagation range, indicating more stringent pilot tracking requirements at 200 kHz. Measurements collected with and without an induced multipath profile further show that additional multipath primarily increases channel estimation variance while preserving the dominant propagation delay. These results provide experimentally grounded insight into pilot design and receiver tracking constraints for high frequency underwater acoustic communication systems.

TUESDAY AFTERNOON, 12 MAY 2026 FRANKLIN HALL 9, 1:00 P.M. TO 2:35 P.M.

Session 2pBAa

Biomedical Acoustics: Biomedical Methods: Messy, Marvelous, and Meh II

Timothy J. Hall, Cochair
Univ. of Wisconsin, Madison, 1153 WIMR, UW-Medical Phys., Madison, WI 53705

Kevin J. Haworth, Cochair
Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586

Virginie Papadopoulou, Cochair
Biomedical Eng., The Univ. of North Carolina at Chapel Hill, 116 Manning Drive, 9004 Mary Ellen Jones Bldg., CB 7575, Chapel Hill, NC 27599-7575

Invited Paper

1:00

2pBAa1. Acoustic rheometry for biomaterials viscoelastic measurement. Weiping Li (Biomedical Eng., Michigan State Univ., 775 Woodlot Dr., East Lansing, MI 48823, liweipi1@msu.edu)

Soft viscoelastic biomaterials play a critical role in mechanobiology and tissue engineering, yet their mechanical characterization remains challenging due to their fragility, heterogeneity, and sensitivity to contact-based measurements. Here, we present a contactless acoustic rheometry approach that using tensile acoustic excitation and resonant surface wave analysis to rapidly probe the viscoelastic properties of soft biomaterials. By monitoring frequency-dependent acoustic responses, we identify a crossover regime in which surface wave dynamics transition from capillary-dominated to viscoelastic behavior. This transition enables quantitative extraction of material

parameters, including effective elasticity and viscosity, over a broad mechanical spectrum. The method is applicable to hydrogel-based biomaterials and biologically relevant soft tissues, providing high sensitivity without physical deformation or probe contact. Our results demonstrate that resonant acoustic responses capture both tensile and surface mechanical modes, offering complementary insights into bulk and interfacial mechanics. This acoustic rheometry framework enables rapid, non-invasive mechanical characterization of soft biomaterials and may facilitate real-time monitoring of dynamic processes such as gelation, remodeling, and blood coagulation in clinical medicine.

Contributed Papers

1:20

2pBAa2. Characterizing cavitation behavior in tissue mimicking phantoms under single-cycle histotripsy. Ruth Mtuwa (Biomedical Eng. and Mech., Virginia Tech, 325 Stanger St., Blacksburg, VA 24061, ruthmtuwa16@vt.edu), Adam D. Maxwell, and Eli Vlaisavljevich (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA)

Histotripsy produces mechanical tissue fractionation via cavitation bubble clouds. Recent studies have shown that increasing pulse repetition frequency (PRF) for single-cycle histotripsy can reduce the cavitation cloud threshold and increase treatment speed However, these studies have been limited to relatively soft tissues and 1% agarose phantoms, restricting translation to tissues with varied mechanical characteristics. This study examined the effects of PRF on cavitation initiation and cloud morphology across agarose phantoms of increasing stiffness and hybrid alginate–polyacrylamide gels. Single-cycle pulses were delivered using a 500-kHz focused transducer. PRFs of 0.1–1000 Hz and peak negative pressures of 2–35 MPa were tested, with cavitation captured using high-speed photography. In 1% agarose, cavitation onset and persistence were consistent with previously reported thresholds. Across stiffer agarose, higher pressures were required to initiate cavitation at low PRFs, while elevated PRFs promoted persistence and increased expansion. Hybrid gels exhibited less consolidated and more heterogeneous bubble clouds than agarose at matched pressures, indicating that viscoelastic properties beyond elastic modulus influence cavitation behavior. These results demonstrate that PRF effects cannot be generalized from one phantom formulation and underscore the importance of tissue mechanics in histotripsy of different tissues.

1:35

2pBAa3. Evaluation of pressure-modulated single cycle histotripsy methods for efficienct and tissue-selective ablation. Sara Elnahhas (Biomedical Eng. and Mech., Virginia Tech, 325 Stanger St., Kelly Hall, Rm. 340, Blacksburg, VA 24061, sarae@vt.edu), Alex Simon (Dept. of Eng. Mech., Virginia Tech, Blacksburg, VA), Adam D. Maxwell, and Eli Vlaisavljevich (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA)

Histotripsy is a mechanical focused ultrasound therapy, clinically validated for liver tumor ablation. Controlled cavitation dynamics are central to safe and selective ablation near critical structures or for the preservation of fine tissue components within targeted volumes. High-frequency histotripsy, due to reduced bubble expansion and lower applied strain, demonstrated in prior *ex vivo* studies the preservation of collagenous liver microstructures; however, it generates small cavitation clouds that result in clinically impractical treatment times. This study investigates the ability of pressure-modulated, low-frequency pulsing to generate large clouds of small, sustained bubbles and achieve faster volumetric ablation *in vitro*. A bubble cloud was initiated using a single 500 kHz high-pressure pulse ($p^- = 40$ MPa) and maintained using lower-pressure pulses delivered at high pulse repetition frequencies (PRFs) in red blood cell and agarose phantoms. High-speed imaging demonstrated that persistent cavitation could be sustained at peak negative pressures as low as 2 MPa across all PRFs tested (1–10 kHz), with increasing maintenance pressure leading to improved red blood cell ablation. These findings demonstrate that pressure modulation enables sustained cavitation activity under reduced pressures, providing a framework for controlling cavitation dynamics of low-frequency histotripsy with potential to reduce energy deposition and enhance ablation selectivity.

1:50

2pBAa4. Focused ultrasound targeting of the cribriform plate/skull base enhances intranasal brain delivery and glymphatic transport. Siaka Fadera (Biomedical Eng., Washington Univ. in St Louis, 4511 Forest Park Ave., St Louis, MO 63108, s.fadera@wustl.edu) and Hong Chen (Washington Univ. in St.Louis, St Louis, MO)

Intranasal drug delivery provides a noninvasive route to the CNS by bypassing the blood–brain barrier; however, its efficiency is limited by restricted transport across the nasal epithelium and into deep brain structures. Focused ultrasound (FUS) has emerged as a promising modality for modulating biological transport pathways, yet its potential to enhance intranasal-to-brain delivery and glymphatic clearance mechanisms remains underexplored. Here, we present preliminary results demonstrating that low-intensity FUS targeted to the cribriform plate/skull base enhances intranasal brain distribution of fluorescent tracers in small animals. Intranasal administration of tracer was performed with and without ultrasound exposure, followed by *ex vivo* fluorescence imaging of brain sections. Compared to intranasal delivery alone, FUS–assisted intranasal delivery produced increased tracer signal within olfactory bulb, ventral brain, and brainstem regions, with broader distribution and increased parenchymal penetration. These findings suggest that FUS targeting of the cribriform plate/skull base may provide a novel, noninvasive strategy to augment intranasal brain drug delivery while simultaneously modulating glymphatic transport. Ongoing studies are quantifying acoustic–transport relationships and regional clearance kinetics. This work establishes an acoustics-driven framework for ultrasound-enhanced nose-to-brain delivery with implications for neurodegenerative disease and therapeutic clearance strategies.

2:05

2pBAa5. Ultrasound-enhanced bacteriophage delivery results in transient bacterial reduction within complex biofilms. Sara Keller (Bioengineering, Univ. of Oregon, Botnar Res. Ctr., Windmill Rd., Headington, Oxford OX3 7LD, United Kingdom, sbkeller@uoregon.edu), Bridget Watson (Environ. and Sustainability Inst., Univ. of Exeter, Penryn, United Kingdom), Ramon Garcia Maset, Veerle A. Brans, Robin O. Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Edze Westra (Environ. and Sustainability Inst., Univ. of Exeter, Penryn, United Kingdom), and Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Bacteriophages have gained interest as antimicrobials due to their high host specificity and minimal off-target effects but face delivery challenges through biofilms. This study investigates a synergistic approach combining focused ultrasound, lytic bacteriophages, and cavitation nuclei to treat biofilm-associated infections. *Pseudomonas aeruginosa* PAO1 was cultured on *ex vivo* porcine skin for 24 h to form a biofilm. Infected skin samples were treated with 1×10^{10} bacteriophages/mL (administered either as a single phage or phage cocktail) and submicrometre proteinaceous cavitation nuclei, followed by ultrasound exposure (0.5 MHz, 2 MPa peak negative pressure, 1% duty cycle). Passive cavitation detection using a 7.5 MHz unfocused transducer confirmed acoustic emissions indicative of violent cavitation across all ultrasound exposures. Bacterial reduction and phage amplification were assessed over 24 h using colony-forming unit and plaque-forming unit assays, alongside scanning electron microscopy and histology. A 2-log reduction in bacterial burden was observed within 6 hours post-ultrasound exposure; however, bacterial levels returned to baseline by 24 h due to confirmed resistance development within the biofilm.

Preliminary results indicate that antibiotics may help sustain bacterial reduction, though further optimization is required. Overall, this work demonstrates the potential of ultrasonic cavitation to enhance bacteriophage delivery through physiologically relevant biofilms.

2:20

2pBAa6. Modeling and the search for mechanisms of sonodynamic therapy. Stephanie C. Walton (Univ. of Oxford, Flat 16, 1 Thames St., Oxford OX1 1SL, United Kingdom, worc6254@ox.ac.uk), Jovana Katrinka Mavrak, Veronica M. Lucian, Luca Bau, Eleanor P. Stride, and James Kwan (Univ. of Oxford, Oxford, United Kingdom)

Sonodynamic therapy (SDT) has been under investigation as a treatment for malignant tumours for over 30 years. SDT utilizes ultrasound in combination with a sonosensitizer, such as p SDT model, and its use in a series of SDT treatments, demonstrating the impact of isolated factors, including acoustic parameters, drug exposure, and externally applied cell stress. Additionally, a search for ROS, including singlet oxygen, hydroxyl radicals, and superoxide believed to be produced as a byproduct following the sonication of PpIX is reported at both clinically relevant and intensified acoustic parameters.type I or II reactions. Other research suggests damage to cell structure due to the ultrasound waves is the key. Understanding the required balance of chemical and physical effects for varied cancer cells is critical to optimization of the treatment. This paper presents the development of a 3D *in vitro* SDT model, and its use in a series of SDT treatments, demonstrating the impact of isolated factors, including acoustic parameters, drug exposure, and externally applied cell stress. Additionally, a search for ROS, including singlet oxygen, hydroxyl radicals and superoxide believed to be produced as a byproduct following the sonication of PpIX is reported at both clinically relevant and intensified acoustic parameters.

TUESDAY AFTERNOON, 12 MAY 2026 FRANKLIN HALL 7, 1:45 P.M. TO 5:00 P.M.

Session 2pBAb

Biomedical Acoustics and Physical Acoustics: Ultrasound Beamforming and Its Applications II

Jian-yu Lu, Chair
Bioengineering, The Univ. of Toledo, 2801 West Bancroft Street, Toledo, OH 43606

Invited Papers

1:45

2pBAb1. Pump-suppressed low-frequency beams from nonlinear acoustic vortices. Jiaxin Zhong (Graduate Program in Acoust., Pennsylvania, State College, PA), Shao-Zhe Li, Jing Lu (Inst. of Acoust., Nanjing Univ., Nanjing, China), and Yun Jing (Graduate Program in Acoust., Pennsylvania, 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu)

Generating highly directional or localized low-frequency acoustic beams is inherently difficult due to strong diffraction at long wavelengths. A common solution exploits nonlinear media, where intense high-frequency pumps generate a demodulated difference-frequency field, a principle used in directional audio, parametric sonar, and medical ultrasound. However, this approach intrinsically couples the desired low-frequency output to regions of high pump intensity, leading to unavoidable high-frequency exposure. Here, we show that nonlinear acoustic vortices can decouple this co-localization and enable topological pump rejection while maintaining efficient low-frequency beam formation. In an airborne nonlinear system, two high-frequency acoustic vortices with identical orbital angular momentum are employed. Their difference-frequency field is topologically trivial yet forms a narrow, well-collimated beam with a flat transverse "listening strip," while the pump fields retain vortex cores that suppress high-frequency pressure in this region. This effect can be interpreted as a topological redistribution of orbital angular momentum across frequency bands. We further demonstrate broadband transient operation from 500 Hz to 8 kHz, highlighting a general strategy for reducing high-frequency exposure without compromising low-frequency performance.

2:05

2pBAb2. Hybridizing conventional and adaptive beamformers. John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu) and Savas Erdim (Sanford, NC)

Adaptive beamformers balance the competing demands of suppressing strong undesired signals by creating destructive interference while attenuating uncorrelated background noise. The hybrid minimum variance distortionless response (HMVDR) beamformer for narrowband passive sonar achieves this balance by factoring its beampattern as a product of a conventional delay-and-sum beampattern and an adaptive Capon beampattern. The conventional beampattern attenuates uncorrelated background noise, while the adaptive beampattern steers nulls to cancel loud interferers. Multiplying the beampatterns corresponds to convolving two smaller weight vectors for notional subapertures. The two factors are chosen so the total number of degrees of freedom (DoFs) in the product equals the number of

sensors in the array. Universal methods allow the hybrid beamformer to adapt the partition of the DoFs between the two factors to respond to changes in the environment. Simulations and microphone array data demonstrate that the HMVDR beamformer performs better than the standard Capon adaptive beamformer or the covariance matrix taper. [Work funded by ONR 321US.]

2:25

2pBAb3. Subspace beamforming with Toeplitz rectified covariance matrices. Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, kwage@gmu.edu) and Vaibhav Chavali (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Adaptive beamformers rely on estimates of the covariance matrix to design weights that attenuate loud interferers and control ambient noise. The performance of large arrays is often constrained by snapshot availability since data required for an accurate covariance estimate scale with array size. One way to improve performance in snapshot-starved applications is to impose structure on the covariance estimate. For uniform line arrays and planewave signals, the ensemble covariance matrix has a Toeplitz structure. The sample covariance matrix (SCM) estimated from snapshots is not Toeplitz, but a Toeplitz structure can be imposed by averaging the SCM along its diagonals, a process known as Toeplitz rectification. Quijano and Zurk [JASA 2017] developed a subspace projection beamformer using a rectified covariance matrix. Motivated by the results from random matrix theory, Chavali and Wage analyzed the eigenstructure of Toeplitz rectified covariances [IEEE SSP 2025] and developed an improved subspace beamformer [IEEE UASP Workshop 2025]. The new algorithm is designed for environments with a large variation in source powers. This talk summarizes the analytical results and illustrates the performance of the existing and improved Toeplitz rectified beamformers using data from the shallow water array performance (SWAP) experiment. [Work funded by ONR 321US.]

Contributed Papers

2:45

2pBAb4. Near-real-time passive cavitation imaging via GPU-accelerated sparse frequency domain beamforming. Rahul Sheoran (Graduate Program in Acoust., The Penn State Univ., 710 S Atherton St., Apt 212, State College, PA 16801, 27sheoran@gmail.com) and Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., State College, PA)

Many applications of therapeutic ultrasound rely on passive cavitation imaging (PCI) to localize and quantify bubble activity. However, standard image reconstruction on CPUs is computationally intensive, often creating significant latency between acquisition and visualization. Here, we present a GPU-accelerated frequency domain beamforming (FDBF) algorithm to spatially map inertial cavitation in near-real time. For this study, we used a 1.1 MHz focused ultrasound (FUS) transducer to generate inertial cavitation in a polyacrylamide gel phantom inside a water tank, which was observed using a 128-element linear L7-4 transducer through the Verasonics® Vantage-128™. A GPU-accelerated sparse FDBF algorithm was programmed to process the recorded PCI data, using a spectral comb filter to remove harmonic components. The results of the proposed algorithm were compared to that of a legacy CPU based algorithm on a system having a quad-core 3 GHz CPU and an RTX 3080 Ti GPU. While the CPU-based algorithm required approximately 30 min to reconstruct a ten-frame PCI dataset, the proposed method completed the same task in approximately 0.4 s, while maintaining comparable image metrics (e.g., peak voltage 2.34×10^4 V vs 3.34×10^4 V and peak intensity 87.4 dB vs 90.5 dB for the legacy code). This algorithm displays potential to achieve near real-time visualization rates suitable for feedback during therapy. [Work supported by NIH-R01EB032860.]

3:00

2pBAb5. Cavitation generation and monitoring with conventional handheld arrays. Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford OX3 7DQ, United Kingdom, michael.gray@eng.ox.ac.uk), Yesna Yildiz, Cameron A. Smith, Brian Lyons, Eleanor P. Stride, and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom)

In cavitation-mediated focused ultrasound therapies, it is common to use separate devices for therapy beam transmission and bubble activity monitoring. In the interest of reducing complexity and cost of instrumentation, we present several examples demonstrating the use of conventional handheld arrays for all required functions: cavitation generation, passive acoustic mapping, and structural imaging. Strategies for transmit and receive beamforming, aperture use, and real time cavitation dose control are presented with three different array geometries and data sets spanning small animal *invivo* and perfused human organ deployments. We further discuss limitations, solutions, and prospects for handheld array deployments in clinical treatment scenarios.

3:15–3:30 Break

3:30

2pBAb6. Power-passive acoustic detection and dynamic differentiation: Mechanisms of subspace selection and power estimation. Nathan Caso (Bioengineering, Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, caso.n@northeastern.edu) and Tao Sun (Bioengineering, Northeastern Univ., Boston, MA)

In passive cavitation imaging for monitoring focused ultrasound therapy, passive acoustic detection and dynamic differentiation (PADAM) beamforming separates cavitation sources based on their frequency spectra. We extend this approach to power-proportional imaging and elucidate the specific mechanism of PADAM. Using isolated stable (SC) and inertial cavitation (IC) signals *in vitro*, we analyze frequency components associated with signal and noise subspaces via a spatial-covariance matrix eigenvector frequency transform. Broadband components related to IC front-load the subspaces with large eigenvalues, whereas harmonic components associated with SC appear lower in the order. PADAM therefore enables spatial separation by tuning a single subspace parameter m (low: 1–5 for IC, high: 15–50 for SC), indicating the location of SC or IC. To achieve power proportionality, we implement a hybrid beamforming approach where a PADAM image acts as a filter over delay-sum-integrate (DSI) images, yielding power weighted PADAM (p-PADAM). p-PADAM differentiates SC and IC with a contrast-to-noise ratio of 18.7 for a stably cavitating source in the presence of IC *in vitro*. The power estimation of p-PADAM scales linearly with DSI-guided frequency-domain power and independently measured PCD signal power, with standard deviations of 1.3-4.4 dB across broadband and harmonic/ultraharmonic components. These results establish p-PADAM as a robust PCI framework that dynamically isolates stable and inertial cavitation while preserving power information.

3:45

2pBAb7. Beamforming with large elements for an increased field of view in 3D ultrasound B-mode imaging. Mick Gardner (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Matthews Ave., Urbana, IL 61801, mickhg2@illinois.edu) and Michael L. Oelze (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Ultrafast three-dimensional (3-D) ultrasound imaging is an emerging topic in the field of medical ultrasound because it provides a full volume view of a region for applications such as obstetrics, abdominal, microvessel imaging, or breast cancer imaging. However, ultrasound matrix arrays have

a prohibitively high element count which limits the field of view (FOV). Our research aims to increase the FOV of 3-D ultrasonic imaging by utilizing larger elements for an increased aperture size with no increase to the element count. Then, advanced beamformers can be applied to optimize image quality. Larger elements were tested by electronically coupling elements on a Vermon 2-D matrix array. Images were beamformed using conventional Delay-and-Sum (DAS), Null Subtraction Imaging (NSI), Directional Coherence Factor (DCF), and Minimum Variance (MV). Furthermore, a positioning system was used to acquire data in sections from a virtual large aperture, four times larger than the Vermon array with the same element count. Combined with a directional MV beamformer, the virtual array increased in resolution by 3.4% compared to a dense array which used DAS. The study suggests that larger apertures can be made from larger elements with sufficient image quality when combining with appropriate beamformers.

4:00

2pBAb8. Diffraction factor of a one-dimensional rectangular linear array for quantitative ultrasound imaging. Khalid Abdalla (Dept. of Phys., Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, k1abdalla@torontomu.ca), Shivani Sharma (Phys., Toronto Metropolitan Univ., Toronto, ON, Canada), and Yuan Xu (Toronto Metropolitan Univ., Toronto, ON, Canada)

The intensity of ultrasound images of a scattering medium depends on the imaging position and probe parameters. The diffraction factor describes this dependence, an essential quantity for describing an ultrasound imaging system. Accounting for the diffraction factor is crucial for quantitative ultrasound imaging. If uncorrected, diffraction alters both the amplitude and spectral shape of echoes, leading to bias in estimates of scatterer properties and attenuation. However, there is no explicit expression so far for a linear array, the most common transducer used in clinics. In this paper, we present a closed-form expression for the diffraction factor for a linear rectangular array and its applications in quantitative ultrasound imaging.

4:15

2pBAb9. Applications of beamforming using quasi-holographic back propagation for extracting information about scattering by penetrable objects. Heather A. Moon (Phys. & Astronomy, Washington State Univ., Webster Hall, 100 Dairy Rd. Rm. 1245, Pullman, WA 99164-2814, heatherannettemoon@gmail.com) and Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA)

Using ultrasonic illumination of elastic objects in water reveals contributions that can appear to be radiated from virtual sources within the structure. Beamforming by using quasi-holographic back propagation limited to supersonic wavenumbers has been used to image enhancements to backscattering [Hefner and Marston, ARLO **2**, 55–60 (2001)] and can reveal the location in space and time for the virtual sources of the scattered sound [Baik, Dudley, and Marston, JASA **130**, 3838–3851 (2011)]. In the present research, quasi-holographic back propagation of nearly backscattered bistatic signals from underwater targets is used to process the data and to take measurements of virtual sources within the target which is achieved by observing the focusing of signal amplitude in the back propagation images. The goal of the present work is to determine information about the backscattered signal showing evidence of axial focusing by axisymmetric objects and providing measurements of the location and diameter of the corresponding virtual ring source. Additionally, as a signal processing tool, unwanted source regions are removed from the back propagated data to isolate the contributions of interest in the backscatter data [Moon and Marston, JASA **158**, 4774–4784 (2025)]. [Work supported by the Office of Naval Research.]

4:30

2pBAb10. Shear wave holography construction of a simplified source condition for focused shear wave beams in soft elastic media. Branch T. Archer (Chandra Family Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 4806 Park Ln., Austin, TX 78732, branch.t.archer@gmail.com), John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Yu-Hsuan Chao (Dept. of Bioengineering, Univ. of Pittsburgh, Pittsburgh, PA), Kang Kim (Dept. of Bioengineering and Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Shear waves are employed in medical ultrasound imaging because they reveal variations in viscoelastic properties of soft tissue. Focused shear waves produced by a longitudinally vibrating piston at several hundred hertz are investigated experimentally in soft tissue phantoms and using an analytical model for shear wave beam generation and propagation. Experiments employed a spherically concave piston shaped to focus the shear wave beam at a depth of 4 cm in a soft tissue phantom. Our initial analytical model was confined to the source plane, with zero displacement surrounding the piston. However, the actual source condition is a traction-free surface surrounding the piston, and the edges of the piston extend 9 mm into the medium. Using shear wave holography, we infer the actual source condition by backpropagation of the displacement field several cm removed from the origin to the plane at the piston edge. We find that the actual source condition for an impulse driving function is dominated by a bipolar edge wave. However, we find that adding a central "bowl wave" inspired by the curvature of the piston further refines the source condition and provides improved accuracy in the near field.

4:45

2pBAb11. Experiment for super-resolution imaging with shear wave. Jian-yu Lu (Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606, jian-yu.lu@ieee.org)

The spatial resolution of imaging based on waves such as optics and ultrasound is limited by the wavelength, as was found by Ernst Abbe in 1873. Previously, the author has proposed to modulate the imaging wave with a shear wave for ultrasound and photoacoustic imaging to overcome such a limit (super-resolution) (Lu, IEEE TUFFC Jan. 2024 and Lu, ASA POMA Nov. 2025). Because the speed of sound of shear wave in biological soft tissues is about three orders smaller than that of the longitudinal ultrasound waves, the half wavelength of the shear wave can be smaller than that of the diffraction-limited resolution of conventional images, especially, when objects to be imaged are deep in the tissues. Using a shear wave that has a short wavelength to modulate the phase of the longitudinal waves, combined with a technique to reduce sidelobes, a method to increase shear-wave amplitude (through shear-wave resonance), and the nonlinearity and spectrum of the shear wave, super-resolution images that can help to characterize tissues via their mechanical and nonlinear properties can be produced. In this study, an experiment of super-resolution imaging with shear wave was performed and the results will be presented.

Session 2pCA

Computational Acoustics: Verification, Validation, and Uncertainty Quantification

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

Jennifer Cooper, Cochair
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Gregory A. Banyay, Cochair
Appl. Res. Lab., Penn. State Univ., P.O. Box 30, State College, PA 16802

Tyler J. Flynn, Cochair
Acoustics & Electromagnetics, Johns Hopkins Appl. Phys. Lab., 1231 Beal Ave., Ann Arbor, MI 48109

Nikolai E. Maltsev, Cochair
Retired, 1467 leaftree cir, San Jose, CA 95131

Chair's Introduction—1:15

Invited Papers

1:20

2pCA1. Toward certifying trustworthy machine learning models for acoustic signal classification. Erin C. Acquesta (Sandia National Lab., 1515 Eubank Blvd SE, Albuquerque, NM 87123, eacques@sandia.gov), Michael Darling (Sandia National Lab., Livermore, CA), and Jake Nichol (Sandia National Lab., Albuquerque, NM)

There is a 30-year history and leadership in verification, validation, and uncertainty quantification (VVUQ) established by the Department of Energy (DOE) National Nuclear Security Administration (NNSA) Labs to evaluate the *credibility* of computational models used in high consequence applications. Today, the National Institute of Standards and Technology (NIST) has established the building blocks for trustworthy artificial intelligence (AI) in response to the advancements in machine learning (ML). In this presentation we describe ongoing work to certify the trustworthiness of ML models by establishing the credibility evidence for classification of acoustic signals through the lens of the broader landscape of trustworthiness criteria. Preserving the fundamental principles and properties across benchmark acoustic signal classification problems, we will demonstrate the variability in trustworthiness evidence in ML models based on their datasets, inference goals, and training algorithms. We plan to address these challenges by defining a taxonomy of AI that outline the appropriate class of algorithms that are well suited to the application context and the corresponding trustworthiness evidence needed for certification.

1:40

2pCA2. A practical application of uncertainty quantification in underwater acoustics. Sheri Martinelli (OBWB, 1100 Louisiana St., Ste. 4900, Houston, TX 77002, martinelli@obwb.com)

Uncertainty quantification (UQ) is a key component of verification and validation of computational models. UQ is also an enabler of quantitative risk, which is a framework in which model results are directly used to communicate risk information to program stakeholders such as managers or regulators in a quantitative manner. Most risk management frameworks in use today are highly subjective and based on broad categorizations into labeled bins, e.g., "red," "yellow," or "green." In this work, a method is described for incorporating results from modeling and simulation to estimate classical quantitative risk quantities, thus translating model results into a language more readily interpreted by stakeholders. Potential applications are discussed in the context of permitting under the Marine Mammal Protection Act of 1972 for, for example, underwater sound due to construction or oil and gas exploration. Challenges specific to the realm of underwater acoustic modeling are also discussed.

2:00

2pCA3. Uncertainty quantification in trans-dimensional geoacoustic inversion. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR 97201), and Julien Bonnel (Scripps Inst. of Oceanogr., San DIego, CA 92123)

This paper considers rigorous uncertainty quantification for the geoacoustic inverse problem of estimating seabed properties from observed ocean-acoustic data. Within a Bayesian formulation, this involves estimating the posterior probability density (PPD) of an assumed model of the seabed, given a model of the residual error statistics. The PPD represents the most general solution to an inverse problem, characterizing the information content of the data and prior knowledge to resolve model structure and quantify uncertainties. For nonlinear problems, the PPD can be estimated using Markov-chain Monte Carlo sampling methods. However, in addition to PPD estimation, the problem of model selection—determining parameterizations for the seabed and error models consistent with the data information—represents a challenging issue that is often neglected but is important for meaningful uncertainty quantification. Here, trans-dimensional (trans-D) inversion is applied to sample probabilistically over seabed models with differing numbers of layers and over error models of differing autoregressive orders within a generalized approach to PPD estimation. Trans-D geoacoustic inversion is illustrated for several types of acoustic data, providing a rigorous comparison of their intrinsic information content to quantify seabed structure and uncertainties.

2:20–2:35 Break

Contributed Papers

2:35

2pCA4. Information geometry techniques applied to bottom loss parameter identifiability. Aiden Edwards (Phys. and Astronomy, Brigham Young Univ., N284, Provo, UT 84602, aidencedwards@gmail.com), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Mark K. Transtrum (Cross Stream Consulting, Provo, UT)

Sound propagation in the ocean depends on the parameterization of the seabed. To quantify the information content about the seabed properties in measured data, three information geometry tools are used. These three tools are illustrated using bottom loss. Bottom loss represents the reflection coefficient of the seabed on a logarithmic scale and depends on both frequency and grazing angle. Some combinations of layer properties produce similar bottom loss results, even if the parameter values are different. An analytical, multi-layered bottom loss model is used where the seabed layers are parameterized by their sound speed and density. Our analysis uses the information geometry of a model manifold, principal component analysis, and the fisher information matrix (FIM). Analysis of model manifolds can provide a measure of the relative importance of parameters under different global conditions. Principal component analysis allows us to evaluate sources of variation in the model manifold. The FIM gives a quantitative measure of how much information a model carries about each parameter. By comparing these techniques, parameterization schemes such as the viscous grain shearing model which contains 16 base parameters for sound speed and density, can be simplified without sacrificing accuracy. [Work supported by Office of Naval Research.]

2:50

2pCA5. Uncertainty quantification of modal parameters applied to surrogate modeling for structural acoustics. Kourtney Libenow (Penn State, Graduate Program in Acoust. The Penn State Univ. 201 Appl. Sci. Bldg., University Park, PA 16802, kra5346@psu.edu) and Gregory A. Banyay (Penn State, State College, PA)

While surrogate modeling provides lower-cost methods to analyze the uncertainty of safety critical systems, high-dimensional outputs present difficulties for implementation. A system with high-dimensional outputs that acousticians commonly measure and model is the modal response of a vibrating structure, as accurate estimation of structural dynamic responses provides the necessary basis set for radiated noise calculations. In this work, we create frequency response functions and evaluate methods to estimate the modal parameters that represent a thin beam in flexure across multiple mode shapes. The methods explored entail rational function approximations and data-driven methods. Our uncertainty analysis quantifies the differences in modal parameter estimation. Error in those modal parameters propagates to error in surrogate models that aim to reconstruct frequency response functions. We close by discussing multiple methods to train and validate surrogate models for a simple vibrating structure to enable credible surrogate modeling for structural acoustic systems.

3:05

2pCA6. Uncertainty propagation for measured frequency responses and MIMO transfer functions. A. J. Lawrence (Carderock Div., Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817, ajlawrence@utexas.edu)

A rigorous measure must include a quantification of uncertainty, and this quantification can be overlooked in acoustics. Bendat and Piersol (Wiley, 2010) established analytical solutions for measurement uncertainty in a limited case of measures, such as autospectral density, cross-spectral density, and SISO transfer functions. In this talk, a framework for finding uncertainty in general acoustic and vibration measurements for non-parametric results from data, i.e., frequency response functions (FRF), using the theory of error propagation will be presented. A focus will be on finding the uncertainty in MIMO transfer functions and insight on excitation type, i.e., periodic versus random, will be discussed.

3:20–3:50 Discussion

Session 2pEA

Engineering Acoustics: The Philadelphia Acoustics Story

Gary W. Elko, Cochair
MH Acoust., 25A Summit Ave., Summit, NJ 07901

Braxton Boren, Cochair
Performing Arts, American Univ., 4400 Massachusetts Ave. NW, Washington, D.C. 20016

Hubert S. Hall, Cochair
Mech. Eng., Texas Christian Univ., 2840 W Bowie St., Fort Worth, TX 76109

Colby W. Cushing, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Chair's Introduction—1:00

Invited Papers

1:05

2pEA1. America's Liberty Bell: The sound of history. Karie Diethorn (Independence National Historical Park, National Park Service, 143 S. 3rd St., Philadelphia, PA 19106, karie_diethorn@nps.gov)

From clarion call to cultural icon, the Liberty Bell resounds prominently in our popular imagination. The Bell represents both historical events (like winning independence during the American Revolution, combating the institution of slavery, achieving women's suffrage) and contemporary experience (commercial branding, heritage affiliations, pride of place). The Bell's sound and appearance resonate with us because they are familiar. As a result of its ubiquitous presence in our history and today, the Bell is regularly taken for granted. Yet, we are often unaware of how and why the Bell was made, who used it and when. This illustrated (and audio) presentation describes the Bell's origin, its adaptation to various purposes over nearly 275 years, and its preservation by generations who have valued its survival as a reminder of shared history.

1:35

2pEA2. Benjamin Franklin's study of sound. Braxton Boren (Audio Technol., American Univ., 4400 Massachusetts Ave. NW, Washington, DC 20016, boren@american.edu)

Benjamin Franklin (1706–1790) was an American polymath and Founding Father whose early scientific achievements were critical for the fledging reputation of the American colonies in the 18th century. Franklin is most famous for his work in electromagnetism, but he also considered several questions related to acoustics during his time in Philadelphia and abroad. In addition to his famous glass 'armonica, Franklin also performed experiments considering the speed of sound in different media, free-field attenuation of sound sources, and speech intelligibility. In addition, he wrote about beating of dissonant tones and phonetic alphabet construction. This presentation gives an overview of Franklin's work in sound, including some parallels between his work in sound and electricity. The impact of Franklin's colleagues in the Philadelphia Junto as well as the American Philosophical Society (APS) are also considered. Franklin's position in the colonies, far from European universities, forced him to take a more practical and hands-on approach to these fields, but (as with other fields he contributed to) his experimentalist bent led to new theoretical advances in the growing understanding of acoustics in the 18th century.

Contributed Paper

2:05

2pEA3. Harvey Fletcher, the Philadelphia Orchestra and the first live-wire transmission of sound. Micah Shepherd (Brigham Young Univ., N249 ESC, Provo, UT 84602, mrs74@byu.edu)

Harvey Fletcher, a founding member and the first president of the ASA, made many notable contributions to acoustics, including his work with the Leopold Stokowski, the Philadelphia Orchestra, and the first live-wire transmission of sound in 1933. This talk will overview the life and career of Harvey Fletcher, starting with his undergraduate studies at Brigham Young Academy. Following graduate school and his work on the oil-drop experiment, Harvey began a career in acoustics research at Western Electric before moving to the Bell Telephone Laboratory. He eventually became director of research at Bell Labs and a professor of Electrical Engineering at Columbia University. He contributed greatly to our understanding of the perception of sound and the development of sterophonic sound. His contributions to electronic sound in conjunction with the Philadelphia Orchestra will be given particular emphasis.

2:20–2:35 Break

Invited Papers

2:35

2pEA4. Historic Philadelphia concert halls and opera houses. Jack McCarthy (12303 Wyndom Rd., Philadelphia, PA 19154, jacksnotes88@verizon.net)

Music historian and archivist Jack McCarthy traces the history of concert halls and opera houses in Philadelphia over a 225-year period, from the first purpose-built music venues of the late eighteenth and early nineteenth century, when Philadelphia was arguably the nation's premier cultural center, to the modern concert halls of the early twenty-first century. Highlights include the ornate Academy of Music, the former longtime home of the Philadelphia Orchestra, which opened in 1857 and is now the nation's oldest opera house still used as such; the Metropolitan Opera House ("the Met"), a massive venue that opened in 1908 and reopened in 2018 after a major restoration; and the Kimmel Center for the Performing Arts, the modern, multi-hall facility that opened in 2001 and is the current home of the Philadelphia Orchestra and several other music groups. In addition to these high-profile venues, the talk will feature a number of lesser-known but important eighteenth-through early twentieth-century concert halls and opera houses, most of which are no longer extant. Illustrated with archival images and photos, the talk highlights the iconic buildings that have supported one of the nation's oldest, most vibrant musical cultures.

3:05

2pEA5. Innovating higher fidelity—The radio corporation of America's legacies in audio engineering, 1927—1957. Alexander B. Magoun (History Ctr., IEEE History Ctr., 445 Hoes Ln., Piscataway, NJ 08854, a.b.magoun@ieee.org)

This paper explores the transformative contributions of the Radio Corporation of America (RCA) to acoustics and audio technology. Founded in 1919, RCA evolved from a wireless communications monopoly into a leading architect of the world's electronic, analog soundscapes. Through its laboratories and staff in Camden and Princeton, New Jersey, the company catalyzed paradigm shifts in broadcasting, recording, reproducing- and synthesizing electronic sound. Key milestones include the innovation of the velocity microphone and development of high-efficiency loudspeaker systems. Under the leadership of Harry F. Olson, RCA's applied physics and engineering set the standard for early high-fidelity sound transduction. Also examined are RCA's critical advancements in underwater acoustics during World War II and its 1940s development of the 45-rpm record system. By concluding with its work in auditorium acoustics, alongside the RCA Mark I and II Sound Synthesizers, this paper illustrates how the corporation contributed to the foundations of electronic signal processing and the global acceptance of high-fidelity sound.

3:35

2pEA6. The auditory perspective and Philadelphia. Gary W. Elko (MH Acoust., 25A Summit Ave. Summit, NJ 07901, gwe@mhacoustics.com)

Between 1929 and 1940, the Bell Telephone Laboratories and Leopold Stokowski, the conductor of the Philadelphia Orchestra, collaborated on developing audio technologies that were the origins of modern high-fidelity sound reproduction. The first public outcome of this collaboration was demonstrated in 1933 with a live high-fidelity (40Hz–15kHz, 80 dB dynamic range, and up to 100 W of acoustic power) 3-channel audio broadcast from the Academy of Music in Philadelphia to Constitution Hall in Washington DC. Bell Labs referred to the demonstrated audio reproduction system as the " and what later was called stereo. The talk will focus on the main technical challenges and the innovations that enabled this relatively unknown and historic 1933 demonstration," and what later was called stereo. The talk will focus on the main technical challenges and the innovations that enabled this relatively unknown and historic 1933 demonstration.

4:05–4:50 Discussion

Session 2pED

Education in Acoustics: General Topics in Education in Acoustics

Jill A. Linz, Cochair
Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866

John R. Buck, Cochair
Electrical and Computer Eng., UMass Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747

Contributed Papers

1:00

2pED1. Creating the acoustics education "Justice League". Liesl A. Hotaling (Univ. of Rhode Island, South Ferry Rd., Narragansett, RI 02882, lieslhotaling@yahoo.com) and Kathy Vigness-Raposa (Venterra Group, Inspire Environ., Newport, RI 02882)

Acoustics is an interdisciplinary field at the nexus of multiple science, technology, engineering and mathematics (STEM) disciplines. Due to this diversity, recruitment and retention efforts face a unique set of challenges. The growing need for expanding acoustics education and training capabilities, increasing workforce retention, and strengthening the diversity of the acoustics workforce creates an urgency to collectively address these needs as a network. This network could allow partners to bring their "superpower" to the collective and offer opportunities for learner engagement beyond what could be achieved by any one group. The discovery of sound in the sea (DOSITS) team is seeking the development of partnerships to construct examples of education and career preparation pathways; create programs and activities for undergraduate students which demonstrate the crosswalk between traditional STEM disciplines and underwater acoustics to assist with recruitment efforts; and create online and in person networking events, mentorship opportunities, and a symposium for students to share their research. To further expand recruitment and awareness, the network will create outreach opportunities for K-12 formal and informal learners and educators. While an ambitious plan, the need for acoustics skills continues to rise and we must collectively take action to respond to the clarion call.

1:15

2pED2. Minecraft education edition acoustics lab: Gamifying a lesson plan about RT60. Ethan Ashoo (Lawrence Technolog. Univ., 1451 lee wood ridge, Commerce Township, MI 48390, ethanashoo@gmail.com), Serena M. Jarbow (Bachelors of Sci., Lawrence Tech. Univ., Walled Lake, MI), and Tre DiPassio (Lawrence Technolog. Univ., Clarkston, MI)

In this session, we present a lesson plan made using the minecraft education edition to introduce students to the concepts of reverberation time (RT60) and material absorption, leveraging the game's immersive environment to create a memorable and engaging laboratory experience. The system features a "Castle Tower" tutorial level that gamifies the explanation of variables in the Sabine equation, followed by a creative sandbox where students can construct rooms in which they will calculate the RT60. A custom MakeCode JavaScript engine performs prompted geometric scanning of the player's environment to calculate the volume and absorption of the room their character is standing in. The engine then generates a simulated audio decay that matches the room's physical properties. The game's native building blocks are also assigned a requisite absorption coefficient so that students can practice "treating" their virtual room, such as by replacing concrete blocks with wool. This lesson plan demonstrates that the minecraft education edition can serve as a powerful interface for teaching foundational acoustic principles, allowing the students to visualize and experience how RT60 can be calculated with the Sabine equation.

1:30

2pED3. Interactive animations of mass-spring-dampers for mobile applications. Sean Jacobsma (Eng., Phase 1 Eng., 12330 James St., Ste H120, Holland, MI 49424, sean.jacobsma@phase1eng.com)

Within the field of acoustics, animations are often used as a teaching aid. Animations are effective as learning tools for acoustics, as many principles involve changes in space and time. Acoustic animations are typically static animations that do not take user input. This talk showcases an interactive animation for the mode shapes of mass-spring-damper systems built as an interactive teaching aid. This interactive animation was built as a mobile application due to the wide availability of smartphones and additionally benefits from the existing animation architecture used for mobile development. Mode shapes are calculated by using the equations of motion for the mass-spring-damper and solving for the eigenvalues and the row reduced echelon form of the motion matrix. The application shows accuracy in line with equivalent mode shape calculation methods. Damping is limited to underdamped stiffness-proportional viscous-damping due to computational complexity. This talk does not cover the effectiveness of learning through interactive animations but merely shows the abilities of this new tool through demonstration. As the application is still in early stages of development, additional animations and features are also discussed. This initial work was done for completion of degree requirements for Penn State University.

1:45

2pED4. Musical reinforcements of phenomena in a Physics of Music class. Jack A. Dostal (Dept. of Phys., Wake Forest Univ., P.O. Box 7507, Wake Forest University, Winston Salem, NC 27109, dostalja@wfu.edu)

I teach a Physics of Music class as a general science elective for undergraduates. Few of my students major in the sciences, but most have some experience making music. We take advantage of this familiarity to use music to introduce and reinforce acoustic concepts. Masking of instruments by other sounds, uniqueness of musical instrument timbre, sound travel time, and harmonics are some of the concepts we address with musical examples. Outdoor music ensembles such as drum and bugle corps and marching bands are complex, rich sources of material. Simpler musical examples such as individual musicians or modified recordings also help in understanding timbre, harmonics, and other concepts. Students also listen to live music and report their ability to identify these phenomena. In this talk, I will describe and demonstrate how we approach acoustic concepts using musical examples.

2:00–2:15 Break

2:15

2pED5. Music education as acoustics education. Andrea Calilhanna (Faculty of Arts, Elder Conservatorium of Music, Adelaide Univ., New South Wales, Sydney, New South Wales 2126, Australia, andrea.calilhanna@adelaide.edu.au)

Traditional acoustics education focuses on the production of sound and its effects, and informs industry standards. The approach steers students toward the instruments of the sound output and their impact on locations and product use. Yet the recent inclusion of music education on the list of acoustics careers by the Acoustical Society of America underscores the importance of music education and the role of acoustics in education. This paper discusses the role of acoustics in music education and provides details on the relationship between acoustics and music theory. Research in music education acoustics, as well as cognitive neuroscience studies involving acoustics, involves psychoacoustics—the mind and body's response to sound (music). As such, music theory requires information about the listener's response to music. The paper discusses a contemporary model of music theory (meter theory) (Cohn, 2020) in relation to teaching music through listening and skills to understand better the acoustics experienced in Ski-hill Graph Pedagogy Meter Fundamentals (Mathematical Music Theory). The paper focuses on how Cohn's ski-hill graph develops the listener's ability to acknowledge the mathematical aspects of low-meter frequencies experienced as duple or triple meter and to apply this to the performance of music.

2:30

2pED6. Teaching acoustics through comparative listening in real and auralized spaces. Kelsey Rogers (Mech. Eng., Tufts Univ., 110 8th St., Greene Bldg., Troy, NY 12180, kelsey.rogers@tufts.edu)

Comparative listening is a powerful tool for teaching acoustics to students of diverse academic backgrounds. Drawing from consulting experience, courses in the tufts music engineering program have recently expanded their use of comparative listening through "acoustic campus tours" and auralizations. Visiting campus spaces and conducting a repeatable routine of guided listening, discussion, and acoustical measurement helps students to translate and deepen their understanding of room acoustics concepts. Auralization is a useful and practical tool with fewer barriers to entry than traditional learning methods, especially for students with limited engineering and scientific background. From introducing basic concepts and developing a shared vocabulary about sound, to correlating room acoustics metrics with perception and illustrating the impact of simulation parameters, auralization has been an effective teaching tool for high school through graduate-level students. This presentation will summarize the evolution of these tools in Tufts courses, including auralization demonstrations and discussion of lessons learned to maximize educational benefit and to promote an inclusive and joyful classroom environment.

2:45

2pED7. Impact of acoustic treatment on attentional performance in informal learning environments. Semiha Yilmazer (Dept. of Architecture, Hasan Kalyoncu Univ., Gaziantep 27000, Turkey, semiha@bilkent.edu.tr), Zekiye Şahin, and Yasemin Afacan (Interior Architecture and Environ. Design, Bilkent Univ., Ankara, Turkey)

Across university campuses, informal learning environments are being increasingly designed to support academic work alongside social interaction, extending beyond traditional classrooms. Despite their growing prominence, the influence of architectural acoustic conditions on attentional performance in these socially active spaces remains insufficiently examined. This study examines the impact of acoustic modifications in campus-based social learning venues, employing a control group and two treatment groups. Sound-absorbing materials were applied to the rear wall and ceiling of one setting to create distinct reverberation times and impulse responses. These impulse responses were convolved with an anechoic background sound recording from a professional sound library to generate realistic auditory stimuli through auralization using the RAVEN software. Following the auralization, key psychoacoustic parameters were calculated using MATLAB software. Participants listened to these sounds via headphones while completing attention control tasks, allowing for controlled performance comparisons across acoustic conditions. Results reveal a non-linear relationship between reverberation time and attention control, with median performance improving at moderate levels of reverberation time before declining at higher levels. The findings highlight the significance of acoustically informed design strategies for informal, socially engaged learning environments.

3:00

2pED8. AI-assisted problem-based learning examples in electroacoustics and room acoustics. Bartlomiej Chojnacki (Mech. and Vibroacoustics, AGH Univ. of Krakow, Mickiewicza 30, Cracow 30-059, Poland, bchojnacki@agh.edu.pl)

Contemporary acoustics education increasingly requires students to work with complex datasets and extensive technical documentation. This presentation introduces an AI-assisted, problem-based teaching approach in electroacoustics and room acoustics using dedicated ChatGPT models trained exclusively on domain-specific instructional materials and reference literature. Rather than replacing theoretical instruction, these models provide structured and efficient access to large bodies of curated knowledge, allowing students to focus on practical interpretation, and physical understanding. Two workshop-based classes will be presented. In electroacoustics, loudspeaker diagnostics based on nonlinear measurements are taught using pre-prepared datasets derived from simulated loudspeaker defects. Full measurement procedures using the Klippel SCN system are demonstrated to explain data acquisition, while complete datasets—representing measurement efforts impractical within class time—are provided to students for detailed analysis. In room acoustics, students apply statistical methods to acoustic treatment design while developing custom absorber concepts, supported by a separate dedicated ChatGPT model trained on course-specific materials and established room-acoustics texts. The presentation discusses the pedagogical design of the workshops, methods for preparing realistic datasets, and challenges observed in AI-supported learnings. Educational benefits, limitations, and implications for acoustics curricula are discussed.

Session 2pMU

Musical Acoustics: General Topics in Musical Acoustics

Jonas Braasch, Chair
School of Architecture, Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180

Contributed Papers

1:30

2pMU1. Molecular sounds. Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu), Oliver Goldman, Emily Gross, and Owen Young (Phys., Skidmore College, Saratoga Springs, NY)

This paper explores methods that can be used to create scientifically accurate sonification of molecules. Initially, absorption spectra for simple molecules such as water, salt, and sugar were investigated by decomposing their continuous absorption spectra into individual spectral components using a gaussian-fit method. Once the peak wavelength and width of each gaussian was determined, they were then mapped into audible frequencies that were used to create the sounds of the molecule. These absorption spectra were tested using both numerical methods in Python and computational methods in Mathematica to find the best method to accomplish this task. Both methods gave very similar results, as well as similar sounds. While ultimately these absorption spectra are continuous functions, the methods used are discrete functions. The known spectrum of a hemoglobin molecule was then investigated using the same methods. Here, the number of gaussians used in the process was also varied. Increasing the number of gaussians resulted in the creation of quasi-gaussian wave packets that closely match the original absorption spectrum. The spectrum of coffee was then obtained using an FTIR. The methods used for hemoglobin were used with similar results, giving a promising method for the sonification of more complicated molecules.

1:45

2pMU2. Pitch-dependent vowel space adjustments and phonatory coordination in western classical singing. May Pik Yu Chan (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104-6228, pikyu@sas.upenn.edu), Jonathan Havenhill (The Univ. of Hong Kong, Hong Kong, Hong Kong), and Jianjing Kuang (Linguist, Univ. of Pennsylvania, Philadelphia, PA)

While pitch-dependent vocal tract adjustments in singing have been well documented, their functional consequences for voice quality remain less clear. This study investigates whether vowel modification strategies observed in professional singers contribute to voice quality stabilization across the pitch range, and whether systematic differences arise from singers' voice type. Fifteen western classical-trained professional singers produced [i], [ɛ], [æ], [ɑ], [u] in a randomized order at every semitone throughout their pitch range. Midsagittal ultrasound tongue images and electroglottography were recorded simultaneously. Tongue images were tracked with DeepLabCut and modeled using functional data analysis. Closed quotients were derived from the electroglottographic signal to examine the relationship between articulatory adjustments and phonatory measures. Results show that singers exhibited limited variation in voice quality, with closed quotient confined to a relatively narrow range (≈0.45–0.65). At the group level, the relationship between tongue body position and CQ is pitch-range dependent. This relationship becomes more linear when faceted by voice type, with differences in directional CQ pattern by pitch and tongue position observed between baritones, mezzo-sopranos, sopranos, and tenors. These findings suggest that articulatory–phonatory coordination strategies may be shaped by both training and voice type.

2:00

2pMU3. Classical acoustic music in North American churches and the negative impact of pandemic-related audio enhancements introduced for worship streaming. Robert G. Hall (Music, Laurentian Univ., 935 Ramsey Lake Rd., Sudbury, ON P3E 2C6, Canada, rhall@laurentian.ca)

This presentation, grounded primarily in personal post-pandemic experiences, will also draw on corroborating perspectives from church musicians across North America. It begins from the conviction that classically informed church music—performed by choirs, pianos, and organs—does not require in-house amplification and is, in fact, compromised by it. Once amplification is introduced, achieving appropriate balance and blend is an insurmountable challenge. During the pandemic, many churches began live-streaming services, most often via YouTube. To support this, they invested in upgraded audio recording systems designed primarily for broadcast. Once installed, this equipment was frequently repurposed for in-house use, leading to the amplification of choirs and pianos that had previously functioned acoustically. Post-pandemic diminishment of performing forces reinforced the perception that amplification was necessary. The goal of the presentation is to remind church musicians that choral and piano music are best left unamplified in live worship. Even musicians who recognize this principle often face pressure from well-meaning audio technicians eager to employ available technology. It is hoped that this presentation could be shared with both the Royal Canadian College of Organists and the American Guild of Organists, supported by its prior presentation to the Acoustical Society of America.

2:15

2pMU4. Influence of bow stick vibration and bow hair modes on string motion and sound output of the violin during normal and skew bowing. Christopher Fang-Yen (Biomedical Eng., Ohio State Univ., 140 W 19th Ave. Columbus, OH 43210-1110, fang-yen.1@osu.edu) and Charles Yao (Biomedical Eng., Ohio State Univ., Columbus, OH 43210-1110)

Vibrations of the bow and/or bow hair have been hypothesized to influence the sound of bowed-string instruments. To test this idea, we developed a bowing machine to control bow velocity, downward force, contact point, and skew angle while measuring bow stick vibration, bow hair movement, string motion, and radiated sound from a violin. We found that skew bowing, in which the bow is not perpendicular to the strings, leads to excitation of horizontally polarized transverse modes in the bow hair that strongly influence the string movement and radiated sound. A model based on bow hair sub-modes explains key features of the sound spectra as a function of the contact point of the string along the bow hair. Hair movement measured using a laser Doppler vibrometer showed Helmholtz-like motion, with multiple slip phases per cycle. We studied how these effects depend on string damping, skew angle, and bow hair fundamental mode frequency. We also investigate the effects of bow stick vibration and find evidence for nonlinear mixing between bow stick and string vibrations. Our results confirm that

vibrations in the bow hair and stick influence sound output, and have implications for the design of bows.

2:30

2pMU5. Evaluation of polymer oboe reed vibration using stroboscopic analysis under artificial blowing. Fumihiko Kurosawa (Graduate School of Sci. and Technol., Univ. of Tsukuba, Tennodai1-1-1, Tsukuba 3058577, Japan, kurosawa.fumihiko.24@aclab.esys.tsukuba.ac.jp), Naoto Wakatsuki, and Tadashi Ebihara (Inst. of System and Information Eng., Univ. of Tsukuba, Tsukuba, Japan)

Polymer reeds are increasingly used in double-reed instruments due to their durability, reduced humidity sensitivity, and potential allergen reduction. However, their vibrational behavior and dependence on constraints similar to those encountered during performance remain incompletely understood. This study investigated the vibration of polymer reeds under artificial blowing conditions using a stroboscope and compared them with traditional cane reeds. Reeds were blown under artificial blowing conditions, and phase-resolved images of periodic opening–closing motion were acquired using a stroboscope synchronized with the vibration. These images were used to examine vibration characteristics such as amplitude, opening–closing patterns, and symmetry. To investigate the effects of realistic playing conditions, measurements were performed under two different scenarios: (i) no constraint and (ii) applying a lip contact constraint near the reed tip. Results revealed distinct differences in motion between polymer and cane reeds, demonstrating that lip contact constraint significantly alters the vibrational behavior of both materials. These findings indicate that both material properties and constraint conditions play crucial roles in double-reed vibration, providing a fundamental basis for discussing their impact on acoustic characteristics.

2:45–3:00 Break

3:00

2pMU6. Effects of register on harmonic content and sound radiation of the alto saxophone. Hanna Pavill (Brigham Young Univ., N284 ESC, Provo, UT 84602, pavillh@gmail.com) and Micah Shepherd (Brigham Young Univ., Provo, UT 84602)

The alto saxophone is a single-reed woodwind instrument that produces sound when a thin reed vibrates, causing the air inside the instrument to resonate. The tone holes control the length of this resonating air, determining which notes are produced. Notes played in different registers require changes in fingering, which can affect both the harmonic content and the way sound radiates from the instrument. While previous research has examined the general acoustics of woodwinds, the influence of register on both harmonic structure and sound directivity has not been fully characterized. This work investigates the harmonic content and directional radiation of a B flat played across three registers. Measurements were performed using a high-resolution directivity measurement system, and the resulting data were analyzed using frequency-dependent balloon plots and spectral comparisons to illustrate how register-dependent fingering affects the sound distribution around the instrument. These observations provide insight into the relationship between register, fingering configuration, and sound radiation in the alto saxophone, contributing to a broader understanding of woodwind acoustics relevant to both musicians and acousticians.

3:15

2pMU7. Abstract withdrawn.

3:30

2pMU8. Abstract withdrawn.

3:45

2pMU9. A comparison of impulse response techniques for marimba auralization in a concert hall. Benjamin Holt (Phys., Utah Valley Univ., 800 W University Parkway, Orem, UT 84058, benjaminjholt@outlook.com), Brian D. Patchett (Phys., Utah Valley Univ., Orem, UT), and Sofina Child (Music, Utah Valley Univ., Orem, UT)

In this project, Utah Valley University's Noorda Center Concert Hall (Orem, UT) is acoustically characterized by employing slapstick impulses as well as linear, logarithmic, and stepped sine sweeps recorded in-hall to calculate the impulse response function. A Marimba One (Arcata, CA) 3100 Basso Bravo Concert Marimba is recorded by a professional musician both in the Concert Hall and in an acoustically dry practice room. The dry recordings are convolved with the hall's impulse responses to generate auralizations. These auralizations are compared quantitatively and qualitatively to the live recording of the marimba performance in the hall to determine which method most accurately reproduces the concert hall's reverberation effect and frequency response. Preliminary analysis indicates that stepped sine sweeps and slapstick impulses provide the most accurate reproduction of the hall's acoustic profile when convolved with dry marimba audio.

4:00

2pMU10. Comparison of directivity of cymbals vibrating in the linear and nonlinear regimes. Logen Bergerud (Brigham Young Univ., N284 ESC, Provo, UT 84602, bergerudl@byu.edu) and Micah Shepherd (Brigham Young Univ., Provo, UT)

Cymbals exhibit nonlinear vibration when struck using normal techniques. Previous studies [e.g., Touze *et al.*, *International Journal of Non-Linear Mechanics*, **46** (2011)] have shown that when excited with a shaker with gradually increasing amplitude, the cymbal first vibrates only at the excitation frequency (periodic regime), then harmonics and subharmonics appear in the Fourier spectrum (transitional regime), and eventually the vibration becomes chaotic (nonlinear regime). Since the harmonic content depends on the degree of nonlinearity, the directivity of a cymbal is also expected to depend on the vibration regime. This study will provide a comparison of the directivity of a cymbal in both the linear and nonlinear regimes using high-resolution directivity measurements. To achieve linear vibration, a commercial striking device (NV-Tech SAM) was used. To achieve nonlinear vibration, an in-house developed striking device was used, which has greater dynamic range than the commercial device. Balloon plots will be presented as a function of frequency to illustrate the changes in directivity for each regime.

4:15

2pMU11. Acoustic characteristics of the slapstick. Daniel Ludwigsen (Phys., Kettering Univ., 1700 University Ave. Flint, MI 48504, dludwigs@kettering.edu) and Jordan Maxgay (Phys., Kettering Univ., Flint, MI 48504)

The slapstick, or orchestral whip, is a specialized percussion instrument comprised of two slats of wood. When brought together rapidly, the resulting sound is reminiscent of the crack of a whip. This study examined the spectral content of five diverse instruments, and for one of them, the spectral content at varying dynamic levels. Third-octave spectra at varied angles were also collected to begin to understand its directionality. Results show significant timbral differences due to design and construction, while other differences are more subtle.

TUESDAY AFTERNOON, 12 MAY 2026 SALON D, 1:15 P.M. TO 2:45 P.M.

Session 2pNSa

Noise: Community Noise Testing and Analysis II

Alexandra Loubeau, Cochair
NASA Langley Res. Ctr., MS 463, Hampton, VA 23681

Kieren McCord, Cochair
Pacific Northwest National Lab., 902 Battelle Blvd., Richland, WA 99354

Contributed Papers

1:15

2pNSa1. Virtual sound level meter: Episode IV—A new hope. Ralph T. Muehleisen (Energy Systems and Infrastructure Assessment, Argonne National Lab., 9700 S. Cass Ave. Bldg 362, Lemont, IL 60439-4801, rmuehleisen@anl.gov)

A long time ago at a university not too far away … I developed a MATLAB based software package for analyzing digital recordings called virtual sound level meter or VSLM. The purpose of VSLM was simple: create a standards grade software tool for students and consultants to use in place of expensive, and fragile, standards grade instruments. VSLM provided various analyses mimicking a type 1 standards compliant, 1/3 octave band sound level meter/analyzer. VSLM was written in MATLAB, compiled to a Windows.exe with the MATLAB compiler, made publicly available, and has been used by educators, students, and consultants for nearly two decades. I have wanted to make updates and release new compiled versions, but since I left university, I no longer have access to either MATLAB or the MATLAB compiler. With the recent advent of AI-assisted software development, I took time during a break from work to use AI and convert VSLM to Python. This presentation describes the conversion to Python, the updates to the original software, and announces the release of the new Python version, PyVSLM, under a GPL license.

1:30

2pNSa2. Rural wedding venue noise assessment. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. M, Minneapolis, MN 55423, david@braslau.com)

A rural wedding venue was proposed in a rural residential area 1 h north of downtown Minneapolis. The venue building would had 2×6 walls and limited windows. A loudspeaker sound level limit of 85 dBA at 10 feet was proposed. The closest homes located only 600 feet were the greatest concern for noise impact. Initial modeling demonstrated that sound levels from the venue would be close to rural background extracted from the literature. However, residents from a Wellness Retreat 6300 feet away convinced the county board to require sound level measurements. A loudspeaker was placed in the proposer‘s garage adjacent to the venue with the same 2×6 walls proposed for the venue. A professional sound engineer played 85 dBA music at 10 feet in the garage. One-third octave band time history levels were taken 10 feet from the loudspeaker and outside within the property in four directions at distances up to 600 feet. Monitored levels compared favorably with initial predictions and published rural background curves. The Wellness Retreat never-the-less continued to object to potential noise and convinced the county board to stop the project. How useful are expensive noise studies in such situations? What could we have done better?

1:45

2pNSa3. Aerodynamic performance of acoustic louvers: Bridging ducted AMCA 500-L testing and open-environment installations. Hamid Mortazavi (Parklane Mech. Acoust., 1180 Speers Rd., Oakville, ON L6L 2X4, Canada, hamid@parklanemechanical.com), Jordan Howes (Parklane Mech. Acoust., Oakville, ON, Canada), and Viken Koukounian (Parklane Mech. Acoust., Burlington, ON, Canada)

Louvers are widely used in industrial applications—particularly in data center facilities—as noise attenuation devices, either as at-source treatments or as freestanding acoustical screens. Despite their extensive use, a significant knowledge gap remains in predicting the aerodynamic and acoustic performance of louvers when used in open environments. Current industry practice relies on the AMCA 500-L standard to evaluate louver aerodynamic performance; however, the standard is based on ducted test conditions that do not accurately represent open-environment installations. This study addresses this gap through a computational investigation of multiple louver configurations with varying free open area percentages and blade angles using computational fluid dynamics (CFD). Each configuration is analyzed under two distinct conditions: (1) the AMCA-500L test setup and (2) an open-environment configuration representative of screen applications.

Comparative analyses of the resulting pressure–velocity fields establish a quantitative mapping between AMCA 500-L pressure loss coefficients and the effective flow resistance and wake characteristics observed under open-environment conditions. This mapping enables translation of standardized aerodynamic test data into application-relevant performance metrics, providing a more physically representative basis for louver selection, aerodynamic optimization, and integrated noise mitigation design in open-environment installations.

2:00

2pNSa4. Improving noise propagation models for data centers: Analysis of source characterization on air-cooled chillers. Saira Hussain (Eng. Mech., Johnson Controls, 5000 Renaissance Dr., New Freedom, PA 17349, saira.hussain@jci.com), Paul F. Bauch (Eng. Mech., Johnson Controls, York, PA), and Sonya Thorpe (Eng. Mech., Johnson Controls, New Freedom, PA)

Driven by the rapid rise of digital infrastructure, large data centers are emerging worldwide, often resulting in increased community noise. Since air-cooled chillers are common noise sources for these facilities, accurately and efficiently quantifying sound emission characteristics is essential for implementing noise control measures that comply with federal and local regulations, as well as customer-mandated sound levels. Manufacturers currently provide overall octave-band sound power levels in accordance with AHRI Standard 370; however, improved sound models and predictions require additional source information. A sensitivity study was conducted to explore methods of improving noise propagation models by incorporating more source details. Three primary sound model approaches were considered for the unit: (a) as a single point source, (b) as individual source components, and (c) as a five-sided box of radiating planes. Noise propagation models using these source characterization approaches were applied to a measured baseline model, hypothetical data center locations, and scaled with various noise control treatments for comparison. This study provides insights for improving current industry standards through determining the most effective approach to HVAC noise propagation models for data centers when applying various noise control treatments.

2:15

2pNSa5. A systemic uncertainty study of the acoustical design of exterior façade assemblies. John LoVerde (Westside Acoust., 801 S Grand Ave. Ste. 1020, Los Angeles, CA 90017, jloverde@westsideacoustics.com), David W. Dong, Samantha Rawlings, and Sunit Girdhar (Westside Acoust., Los Angeles, CA)

The acoustical design of exterior façade assemblies involves measuring or calculating exterior sound levels and spectra, obtaining transmission loss data for proposed exterior window, wall, and door assemblies, and calculating interior sound level based on the facade dimensions and expected room absorption. The authors have previously presented studies of the variation in transmission loss data of windows (ICA 2013), measured traffic noise levels (Inter-Noise and ASA 2014 meetings), and measured traffic noise spectra (Acoustics 2017). These findings are combined and extended into a full uncertainty model of the calculation procedure, and the total uncertainty is calculated using Monte Carlo methods based on measured and assumed probability distributions. This quantifies the overall uncertainty in the calculation method and provides understanding of the relative uncertainties of the various components, quantifying which assumptions are more important and providing guidance as to where more detailed study should be focused.

2:30–2:45 Discussion

TUESDAY AFTERNOON, 12 MAY 2026 SALON D, 3:15 P.M. TO 4:40 P.M.

Session 2pNSb

Noise, Architectural Acoustics and Structural Acoustics and Vibration: Building Base Isolation

Bonnie Schnitta, Chair
SoundSense, LLC, 39 Industrial Rd., Unit 6, P.O. Box1360, Wainscott, NY 11975

Chair's Introduction—3:15

Invited Papers

3:20

2pNSb1. Human sensory systems as vibration sensors: Neural coding of environmental vibration. Kathleen E. Cullen (Biomedical Eng., Johns Hopkins Univ., 720 Rutland Ave. Traylor Res. Bldg., Baltimore, MD 21205, kathleen.cullen@jhu.edu)

Ground-borne and structure-borne vibrations are commonly considered in building design to protect equipment and reduce audible noise. Humans themselves are biological vibration sensors, and these stimuli activate neural systems that evolved to detect mechanical motion. This talk reviews neuroscience findings on sensory responses to vibration. The auditory and vestibular systems share a common evolutionary and developmental origin and function as detectors of mechanical vibration. The vestibular system represents an evolutionarily older modality specialized for encoding low-frequency motion and instability, while the auditory system emerged later as a specialization for higher-frequency vibrations. As a result, ground-borne vibrations that fall below conscious auditory perception can

nonetheless activate vestibular pathways. Experimental studies demonstrate that low-frequency vibration robustly engages vestibular pathways that encode self-motion and shape multisensory and motor processing. Neural sensitivity to vibration differs from auditory audibility and comfort criteria; vestibular responses can occur even when vibration is not perceived as sound. Sustained exposure to structured vibration alters the sensory input statistics experienced by the nervous system, leading to persistent changes in neural gain, tuning, and response variability. These observations are relevant for spaces such as hospitals, laboratories, and residential environments, where low-frequency vibration may be present despite meeting acoustic criteria.

3:40

2pNSb2. Acceptability of wind-induced vibrations in tall buildings: Impacts on occupants' work performance. Kaveh Heshmati (WSP UK Ltd., Flat 921 2 Engineers Way Wembley Park, London HA9 0SA, United Kingdom, kaveh.heshmati@wsp.com)

Wind-induced vibrations are typically narrow-band random with low frequencies, commonly within the range of 0.1 to 1.0 Hz. These vibrations can resonate with tall buildings, which inherently have low natural frequencies. If resonance occurs, it can cause discomfort—or even fear—among tall building occupants. This paper explores the acceptability of wind-induced vibrations in tall buildings and in particular, the effects of these vibrations on work performance of tall buildings' occupants. It includes a brief overview of the historical research context, current serviceability criteria applied in practice, and the emerging need for a paradigm shift in assessing occupant comfort from a serviceability perspective. Wind-induced vibrations were simulated using a unique motion simulator at University of Bath (VSimulator). A comprehensive psychological test battery was developed in PsychoPy to measure a broad range of cognitive abilities, including attention, memory, and executive function. Research participants were exposed to different motion conditions while carrying out the test battery. The results showed that wind-induced vibrations negatively affected cognitive performance of participants. The outcome of this research provides a foundation for future vibration serviceability studies where occupants' work performance is central to defining serviceability criteria.

4:00

2pNSb3. Mitigation of ground-borne/structure-borne vibration for new and existing buildings—From concept to specification and installation. Florian Sassmannshausen (GERB Vib. Control Systems, 1950 Ohio St., Lisle, IL 60532, florian@gerbusa.com) and Michael Wolf (GERB Vib. Control Systems, Lisle, IL)

Urbanization, densification, and the desire to retrofit buildings adjacent to rail infrastructure require owners and engineering firms in the built environment to explore options for protection of ground-borne/structure-borne vibration, to shield tenants from noise and vibration, or to protect sensitive equipment from external vibration. This paper explores the integrated approach between owners, engineers, and manufacturers, the material science of various isolation and damping mediums, and describes the means and methods of integration for both new construction and retrofit applications.

4:20

2pNSb4. Case studies of structureborne noise from internal building sources. Nicholas Roselli (SoundSense, LLC, 485 Madison Ave., 7th Fl., New York, NY 10022, nick@soundsense.com), Bonnie Schnitta, and Kristof Hefty (SoundSense, LLC, New York, NY)

Structureborne noise is often challenging to effectively mitigate. To minimize the risk of structureborne noise from external sources such as subway vibrations transmitting into a building's framework, the ground should ideally be treated prior to the pouring of the foundation. Even with appropriate ground treatment measures, clients may occasionally question the effectiveness of such measures if noise issues persist. Structureborne noise from internal sources of structureborne noise such as mechanical equipment can frequently be just as disruptive as external sources like subway vibrations. Therefore, it is also critical that these sources also be properly treated. This presentation outlines several challenges where structureborne noise from internal sources resulted in disruptions similar to those seen from external sources. The first are acoustic leakage points, that are easily treatable. The second is structureborne noise from the spin cycle of a washing machine that resembled subway noise and traveled over 10 floors. Finally, a tonal noise from a small sand filter pump located on a mechanical floor which resulted in significant standing waves that varied by as much as 4 dB at the tonal frequency. By accurately identifying these unrelated noises can ensure the success of noise mitigation strategies.

Session 2pPA

Physical Acoustics, Biomedical Acoustics, Noise, Education in Acoustics, and Computational Acoustics: Celebration of Allan D. Pierce's Contributions to Acoustics II

T. Douglas Mast, Cochair
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Andrew A. Piacsek, Cochair
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Invited Papers

1:00

2pPA1. Celebrating Allan Pierce's contributions—Perspective of a grateful former PhD student. Mario Zampolli (IMS/ED, Comprehensive Nuclear-Test-Ban Treaty Organization, CTBTO, Vienna 1400, Austria, Mario.ZAMPOLLI@ctbto.org)

This presentation addresses a very small subset of Allan Pierce's contributions to acoustics from the point of view of a former doctoral student whose first contact with the advisor was through Pierce's textbook, which continues to be a cornerstone for students, scientists, and practitioners. On the other hand, another immensely useful Pierce book, the monograph on "Variational formulations in acoustic radiation and scattering" (Vol. 22, Physical Acoustics, 1993), is regrettably not available anymore from the publisher. An anecdotal example of its use, a variational formulation for the acoustic field in a micro-resonator and horn transducer with strongly coupled normal modes, is presented. The Sommerfeld radiation condition at the square opening of the horn is satisfied by a Rayleigh integral using Pierce *et al.* [J. Acoust. Soc. Am., **111** (2002)], a paper which continues to be referenced in articles that deal with the efficient computation of radiation from transducers and openings. A final note pertains to the impact of Pierce's early work on atmospheric infrasound propagation, which was pivotal for the development of infrasound monitoring of atmospheric nuclear explosions: one of the four monitoring technologies constituting today's comprehensive Nuclear-Test-Ban Treaty International Monitoring System for global detection of nuclear test explosions.

1:20

2pPA2. Allan D. Pierce's three rules—Celebration of Allan D. Pierce's contributions to acoustics. Sean F. Wu (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., College of Eng. Bldg., Rm. 2133, Detroit, MI 48202, sean_wu@wayne.edu)

I consider myself extraordinarily fortunate to have been accepted as both a Master's and a Ph.D. student by Professor Allan D. Pierce at the Georgia Institute of Technology from 1983 to 1987. Recognizing the significance of this opportunity, I immersed myself immediately in my studies. My research focused on developing a new variational principle for predicting acoustic radiation and scattering from finite objects, and on demonstrating its advantages and computational efficiency relative to existing methodologies. With little prior background, I faced significant challenges. Rather than criticizing my early errors in calculations, Allan responded with concise and transformative guidance in three simple yet powerful rules: (1) *Think first, code later*; (2) *Be a snob*—being smart in going through a sea of scientific literature; and (3) *Get something done*. In hindsight, these rules provide an exceptionally powerful framework for young researchers, helping them balance rigor, efficiency, and productivity. Since joining Wayne State University in 1988, I passed Allan's three rules on to my own students and added a fourth principle inspired by his mentorship: *When in doubt, return to the fundamentals*. Together, these rules reflect not only Allan D. Pierce's intellectual legacy, but also his enduring influence on generations of acousticians.

1:40

2pPA3. Allan Pierce and ultrasound heating of brain tissue. Robin O. Cleveland (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, robin.cleveland@eng.ox.ac.uk)

Therapeutic ultrasound for use in treating disease in the brain goes back to work done by the Fry brothers in the 1950s. While at MIT Allan Pierce collaborated with Padmakar Lele who carried out extensive fundamental experiments on ultrasound lesion formation in the brain. A seminal paper by Lele and Pierce (*Proceedings of a Workshop on the Interaction of Ultrasound and Biological Tissues. FDA73-8008* (1973), pp. 121–128 discusses the mechanisms by which the lesions form and argued that thermal mechanisms were dominant; and crucially they identified that lesion formation was not a temperature threshold effect but that it was the time history of the temperature that resulted in lesion formation. Their hypothesis will put into perspective of the subsequent development of the thermal dose which accounts for the temperature history and has its roots in the Arrhenius equation for reaction rates. The relevance of Lele and Pierce's work to current applications of therapeutic ultrasound in the brain will be discussed. The talk will also reflect on the experience of being a junior faculty member at Boston University in the 1990s with Allan as Head of Department.

2:00

2pPA4. Fuzzy structures, lengthy footnotes, this thing, and the guy who married Evelyn Balfour: How Allan Pierce influenced my academic career. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

When I began my doctoral studies in the Graduate Program in Acoustics at Penn State in 1991, my advisor Vic Sparrow was the junior faculty in Pierce's research group. Part of my doctoral dissertation involved applications of the Pierce–Sparrow–Russell model of fuzzy structures. My contribution to an early publication about this model was some TeX code and several figures, a contribution resulting in the only time I heard Pierce swear. While at Penn State, I took a class on Plates and Shells from Pierce and learned to take notes without looking at what I was writing; watching him closely because he referred to every variable he wrote as "this thing" or "that thing."ll highlight these and the other ways that Allan Pierce positively influenced my own academic teaching career.'s affection for Lord Rayleigh (the subject of his Rossing Prize Lecture in 2004) has further enhanced my own appreciation of Rayleigh. And his encouragement for Education in Acoustics within ASA and JASA has guided my manuscript writing. In this talk, I'll highlight these and other ways that Allan Pierce positively influenced my own academic teaching career.

2:20–2:35 Break

2:35

2pPA5. Design of an acoustic experiment with a large number of receivers. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215-2407, barbone@bu.edu) and Alison Malcolm (Dept. of Earth Sci., Memorial Univ., St. John's, NF, Canada)

In the mid 1990's, AD Pierce urged the first author (PEB) to read Claude Shannon's classic 1948 monograph. In a 1995 report, ADP connected the Jaynes–Shannon maximum entropy principle to the modeling of structural systems about which only limited information was available. That was PEB's first exposure to information theory being applied to problems in acoustics, outside of acoustic communication. In more recent years, the authors (again) encountered information theory being used in acoustics, in this case in the seismic imaging community. There (and elsewhere) it is used to design experiments to maximize the expected information gain from a measurement. As a function of the measurement design parameters, the information gain is often both highly non-linear and non-smooth. It is therefore particularly challenging to optimize in a high number of parameter dimensions, as when using a large number of receivers, $N_r >> 1$. We examine the asymptotics of the large N_r regime and derive an easily evaluated asymptotic approximation of the information gain. We apply our results to the example of layer property inversion in an amplitude-versus-offset seismic measurement scenario. In this example, useful results can be obtained from the asymptotic approximation for N_r as low as 2.

2:55

2pPA6. Reciprocity in electroacoustic and vibrating systems by Allan D Pierce. Charles Thompson (Elec. and Comput. Eng., UMass Lowell, 1 Univ. Ave. Lowell, MA 01854, charles_thompson@uml.edu)

Equations governing physical systems often admit a reciprocity corollary. Such systems include mechanical, acoustical, vibrational, and electroacoustic systems. There is no universally applicable, succinct definition of reciprocity, although it is sometimes loosely stated that reciprocity means that "if you see it, then it can see you." A considerable number of derivations of reciprocity theorems for various acoustical and mechanical systems have been published over the years—notable names include Helmholtz, Rayleigh, Lamb, Lyamshev, Schottky, Foldy, Primakoff, Payton, de Hoop, Knopoff, Gangi, Arntsen, Carcione, and Fokkema. There appears to be no universally applicable derivation, and no universally applicable reciprocity principle. The emergence of metamaterial systems poses new challenges and opportunities regarding reciprocity. Given one such system, one naturally asks if there is some reciprocity theorem that applies. If one perceives that there is no such theorem, then one might naturally ask whether the perception is wrong, and if it is simply the case that no one has yet bothered to try to derive one or that one is not sufficiently clever to find one. It may also be that there is an analogous system for which there is a theorem, and that the theorem applies to the present system in some approximate sense. If so, how well? This paper examines several systems drawn from the recent reciprocity-theorem literature [*J. Acoust. Soc. Am.* **143**, 1946 (2018). https://doi.org/10.1121/1.5036376].

3:15

2pPA7. Allan Pierce, Leopold Felsen, and approaches to generalizing ray theory for fluid-structure interactions from the late 1980s. Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

The late 1980s commenced a growing interest in ray theory for fluid–structure interactions as evident by examples [A. D. Pierce and H. G. Kil, *J. Vibr. Acoust.* **112**, 399–408 (1990); L.B. Felsen *et al. JASA* **87**, 554–569 (1990); S. G. Kargl and P. L. Marston *JASA* **85**, 1014–1028 (1989); L. G. Zhang *et al. JASA* **91**, 1862–1874 (1992)]. This presentation summarizes aspects of the diverse approaches associated with Pierce, with Felsen, and with the present author. For example, Pierce's research on elastic cylindrical shells examined shell waves in an unbounded (unwrapped) anisotropic dispersive medium. Approaches developed by the present author and associates started with the Watson transformation and approximations for coupling with the acoustic field based on similarities with resonant scattering theory (RST). Measured scattering by objects in water was also relevant. Subsequent generalizations of approaches allowed for formulating of coupling involving Fresnel zone areas and considered anisotropies relevant to helical and meridional rays [F. J. Blonigen and P. L. Marston, *JASA* **110**, 1764–1769 (2001)] and for truncations and variable curvature. Recent applications include [B. R. Hall and P. L. Marston, *JASA Express Lett.* 2, 114001, (2022)]. Geometry is helpful for identifying dominant scattering contributions. [Work supported by ONR.]

Session 2pPP

Psychological and Physiological Acoustics: Psychological and Physiological Acoustics: Best Student Award Poster Session

Afagh Farhadi, Chair
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All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

2pPP1. Effects of wall reflections and training on sound image localization using two parametric loudspeakers. Noko Kuratomo (Ergonomics Lab., Railway Tech. Res. Inst., 2-8-38, Hikari-cho, Kokubunji-shi, Tokyo 1858540, Japan, kuratomo.noko.49@rtri.or.jp) and Kei Hoshino (Ergonomics Lab., Railway Tech. Res. Inst., Tokyo, Japan)

To implement non-wearable binaural sound presentation in indoor environments, this study examines a method using two parametric loudspeakers to create auditory images at various angles. Previous research has primarily measured localization accuracy in anechoic chambers or only after training. However, for practical indoor applications, it is essential to examine the effects of sound reflections and training. The aim of this study is to clarify how sound image localization is affected by wall reflections and training. Seventeen participants listened to natural language speech from two parametric loudspeakers under two conditions: with and without training. Sounds were presented randomly in 12 azimuth positions at 30° intervals from 0° to 330° for each condition. Participants recorded the perceived direction of the sound image on paper. Results revealed that the localization error for frontal sound image was significantly larger than for other directions, likely due to reflections from the rear wall. Furthermore, when training was provided, frontal localization accuracy improved, indicating a learning effect.

2pPP2. Effects of head covering on directional microphone benefit in hearing aids. Zahra Fekri (Commun. Sci. and Disord., James Madison Univ., 865 Port Republic Rd., 506, Harrisonburg, VA 22801, Fekrizx@dukes.jmu.edu), Mariam Mohamad (Commun. Sci. and Disord., James Madison Univ., Minneapolis, MN), and Ayasakanta Rout (Commun. Sci. and Disord., James Madison Univ., Harrisonburg, VA)

Directional microphones in hearing aids improve speech understanding in background noise. Few studies have examined the effect of various objects on directional microphone function, including ear mold venting, glasses, and hair. However, the impact of head covering has not been studied. To study the effect of a head covering on directional microphone function in receiver-in-canal (RIC) hearing aids from four manufacturers in a laboratory setting. Premium hearing aids were programmed for bilateral mild sloping to moderately severe hearing loss. Adaptive signal processing features were disabled, and two identical programs were created—fixed directional and omnidirectional. Polar plots were obtained in 30 deg intervals with and without head covering at 500, 1000, 2000, and 4000 Hz. Directivity Index was calculated at each frequency for the four conditions. The data were analyzed using a 2×2 repeated measures ANOVA. Directional microphone settings increased directivity index values at all frequencies and among all manufacturers. There was no significant main effect of head covering on the directivity index. There were also no significant interactions between head covering and manufacturer. However, manufacturer-related differences were statistically significant. Lightweight cotton head coverings such as hijabs do not significantly reduce directional microphone benefits.

2pPP3. Mapping between speech acoustics and trait impressions reveals implicit speaker-specific patterns. Qingzhi R. Zeng (Univ. of Rochester, 500 Joseph C. Wilson Blvd., Rochester, NY 14623, qzeng6@ur.rochester.edu), Nadine Lavan (Queen Mary Univ. of London, London, United Kingdom), and Elise Piazza (Univ. of Rochester, Rochester, NY)

Listeners rapidly form trait impressions from voices. Although there are established links between perceived social traits and speech acoustics, prior work typically used limited feature sets and did not examine whether these mappings are organized in a speaker-specific manner. To bridge these gaps, we sampled multiple word-length clips from naturalistic conversational speech recordings and extracted 13 acoustic features across pitch, rhythm, timbre, and voice quality. We also collected online ratings of perceived attractiveness, dominance, and trustworthiness of those clips. First, we comprehensively mapped the acoustic drivers of the three perceived traits. Attractiveness was linked to higher speaking rate, spectral brightness, and voice clarity; dominance was linked to larger pitch range, faster rhythm, and brighter but more percussive, hoarser sound; trustworthiness was linked to lower pitch, faster rhythm, and clearer voice quality. Importantly, clips that were more acoustically similar had more similar ratings across all traits, and these associations were stronger for clips from the same speaker than for clips from different speakers, even though listeners were not aware of the number of speaker identities. These results shed new light on the acoustic mechanisms underlying vocal trait impressions and their role in supporting an implicit perceptual structure linked to speaker identity.

2pPP4. The effect of interaural level differences on binaural fusion in the presence of interaural frequency dissimilarity. Itunu M. Oyebamiji (Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL, itunumo2@illinois.edu) and Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Binaural fusion is the ability to perceive sounds presented to both ears as a single auditory percept. It can be degraded by reduced interaural coherence, defined as the statistical similarity between signals at the two ears, and by interaural frequency dissimilarity, defined as differences in frequency presented to each ear. Previous work has shown that adding large interaural level differences (ILDs) can improve binaural fusion for signals with reduced interaural coherence. The present study examined whether adding large ILDs can also improve binaural fusion in the presence of interaural frequency dissimilarity. In experiment 1, 28 typical hearing listeners were tested using a single-channel vocoder with varying degrees of interaural frequency dissimilarity and either a 0 or 16 dB ILD. In experiment 2, two typical hearing listeners were tested with similar stimuli but with a wider range of ILDs. Listeners reported the number, size, and perceived location of auditory images. Results showed that increasing interaural frequency dissimilarity degraded binaural fusion. However, fusion improved when ILDs produced strongly lateralized percepts. These findings suggest that ILDs

may facilitate binaural fusion in the presence of interaural frequency dissimilarity, although perceived lateralization likely plays a role in this improvement. [Research funded by NIH/NIDCD grant R01-DC018529.]

2pPP5. Exploring low-frequency biasing effects on distortion-product otoacoustic emissions. Vikas K. Lakhmani (Mech. Eng., Indian Inst. of Technol. Bombay (IIT Bombay), Acoust. and Hearing Lab., IIT Bombay, Mumbai, Maharashtra 400076, India, lakhmani.1@iitb.ac.in), Dharmesh Verma (Ctr. for Sophisticated Instrument Facilities, Indian Inst. of Technol. Bombay (IIT Bombay), Mumbai, Maharashtra, India), and Sripriya Ramamoorthy (Mech. Eng., Indian Inst. of Technol. Bombay (IIT Bombay), Mumbai, Maharashtra, India)

Distortion-product otoacoustic emissions (DPOAEs) are widely used as a non-invasive tool to study nonlinear cochlear mechanics and outer hair cell (OHC) mediated amplification. This work presents an ongoing experimental study examining how low-frequency bias tones influence DPOAE generation in the human ear. DPOAE measurements are being conducted on normal-hearing human subjects using controlled two- and three-tone stimulation paradigms. In addition to the primary tones (f_1 and f_2), an extra low-frequency tone is introduced to bias cochlear partition, allowing the effects of bias frequency and level on the cubic distortion product ($2f_1 - f_2$) to be examined. Measurements are performed across a range of stimulus frequency ratios and levels using consistent recording protocols. The study is designed to explore how low-frequency biasing alters DPOAE amplitude and spectral characteristics, potentially reflecting changes in the nonlinear operating point of cochlear transduction. Methodological considerations and insights from this study will be presented and discussed.

2pPP6. Linking speech acoustics to listener perception and experience in dysarthria. Morgan Robertson (Speech and Hearing Sci., Utah State Univ., 1000 Old Main Hill, Logan, UT 84322, morgan.robertson@usu.edu), Annalise Fletcher, and Stephanie Borrie (Speech and Hearing Sci., Utah State Univ., Logan, UT)

Dysarthria, a motor speech disorder, is often treated as a problem of intelligibility. However, individuals with dysarthria have reported reduced communicative effectiveness even when their speech remains intelligible (Dykstra *et al*., 2015). Recent research has shown that listeners experience greater cognitive effort when processing dysarthric speech (Fletcher *et al*., 2022; Hirsch *et al*., 2025), highlighting that intelligibility alone may not capture the listener's experience. This study challenges the assumption that intelligibility alone ensures communicative success by investigating the acoustic correlates to emotional perception and listener affect in dysarthric speech. Speech samples from 32 speakers with varying types of dysarthria were acoustically analyzed and presented to 50 naïve listeners. After hearing each sample, listeners rated speech intelligibility, pleasantness, and animation and their own comfort, frustration, and fatigue while listening. Correlation analysis showed medium to large associations between scales, supporting the conclusion that listeners form global impressions that integrate both intelligibility and emotion. Among the acoustic measures examined, articulation rate, speech rate, and F2 slope emerged as the most consistent and robust predictors of ratings across scales. Overall, the convergence of listener ratings and their shared acoustic predictors suggests that intelligibility, emotion, and listener affect are not perceptually independent in dysarthric speech.

2pPP7. Spatial positioning in a complex auditory environment by listeners with varying levels of autistic traits. Alyssa D. Swann (Speech-Lang. Pathol. & Audiol., Towson Univ., 8000 York Rd., Towson, MD 21252, adolan9@students.towson.edu), Kelly Coburn, and William J. Bologna (Speech-Lang. Pathol. & Audiol., Towson Univ., Towson, MD)

Understanding speech in noisy, multitalker environments can be challenging, and these difficulties may be heightened in individuals with elevated autistic traits due to differences in sensory processing. This study examined how spatial positioning within a simulated restaurant affects speech recognition in young adults with varying levels of autistic traits. Thirty participants completed an adaptive speech-in-noise task at 12 listening positions, each reflecting different target and masker locations. Participants also selected preferred seats before and after testing. Speech-in-noise thresholds were analyzed using repeated-measures ANOVA and linear mixed-effects modeling to evaluate effects of spatial location, hearing sensitivity, and autistic traits. Results showed significant effects of table, seat, and their interaction, confirming that spatial position strongly influences speech recognition. Autistic traits did not produce a consistent main effect, but a significant interaction indicated greater performance variability among participants with higher autistic traits, with some performing better and others worse depending on position. Seating preferences did not differ between autistic-trait groups or change after testing. These findings highlight the importance of spatial factors in communication in noise and emphasize the heterogeneity of auditory performance in individuals with elevated autistic traits, supporting individualized strategies in clinical and real-world environments.

2pPP8. Speech category learning across the lifespan: Simple and complex categories. Ghazal Koohkansaadi (Psych., Univ. of New Hampshire, McConnell Hall, Durham, NH 03824, Ghazal.Koohkansaadi@unh.edu), Christopher Heffner (Dept. of Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY), Brett Gibson, and Casey L. Roark (Psych., Univ. of New Hampshire, Durham, NH)

While adults generally struggle to learn new speech categories, it is not clear how learning changes across the lifespan. For nonspeech categories, learning changes across the lifespan are directly related to executive functions which undergo substantial changes across development, improving quickly across childhood to young adulthood and declining slowly in middle to older adulthood. In the current study, we examined speech learning across a wide age range (7–69 years, N = 99, monolingual English speakers) to understand how developmentally sensitive executive functions may contribute to successful speech learning. Stimuli formed a nine-step continuum of German fricatives (/x/ to /ç/) which was either divided into three continuous categories or two disjunctive categories. Disjunctive categories require more complex rules to learn, as perceptually distinct stimuli at the ends of the continuum belong to the same category. We compared the learning trajectories of continuous and disjunctive categories across the lifespan using generalized additive models. Learning of continuous categories improved steeply across childhood, plateauing in early adulthood. In contrast, learning of disjunctive categories was relatively constant across the lifespan. Overall, these findings suggest that age-related executive functioning abilities may contribute to successful learning of speech categories that can be easily described by rules.

2pPP9. Comparing between-subject and within-subject variability in digits-in-speech. Vahid Delaram (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901, S 6th St., Champaign, IL 61820, vahidd2@illinois.edu), G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE), Emily Buss (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Human speech-in-noise measures show substantial variability across listeners. Part of this variability may be accounted for by within-subject variability, which could occur due to experimental factors, such as trial-to-trial differences in speech and masker spectral content, or due to changes in subject-related factors such as attention, learning, and fatigue. In this study, we sought to quantify this within-subject variability by measuring speech reception thresholds (SRTs) using a digits-in-speech task. Listeners completed 16 nearly identical experimental runs in an adaptive track (SRT = mean of last 6 reversals), with the only difference between runs being randomization of the target digits and masker segments on each trial. Two filtering conditions (16 runs each) were tested: full-band and low-pass filtered at 8 kHz. The average range and standard deviation of thresholds across subjects within an experimental run (across-subject variability) were 10.2 and 3.4 dB and within-subject across experimental runs (within-subject variability) were 8.8 and 2.4 dB, respectively. A repeated measures ANOVA showed significant effects of filtering on measures of within-subject variability, with increased variability when using low-pass filtered signals. The large within-subject variability across estimates of digits-in-speech recognition could limit our ability to characterize individual differences using these methods.

2pPP10. A study on perception of lexical pitch accent in Yanbian Korean. Junshu Jin (English Dept., Yanbian Univ., 162 5th St., Cambridge, MA 02141, jinjunshu@sina.com)

Based on the "prosodic feature," world languages can be classified into three major categories: stress languages, tone languages, and pitch accent languages. Typologically, pitch-accent languages stand between stress languages and tone languages, and share properties of both. Yanbian Korean, as a lexical pitch accent language, has a lexical pitch accent system, in which lexical pitch accent is an internal property of lexicon. The purpose of the present study is to identify the nature of categorical perception of lexical pitch accent, using Yanbian Korean as a case language, which has been largely overlooked in previous literature. By adopting the paradigm of identification and discrimination experiments, the primary acoustic cue of lexical pitch accent, fundamental frequency (f0), is manipulated in a nine-step continuum based on three minimal pairs of real words contrasting in pitch accent. The results indicate that the lexical pitch accents are perceived categorically, and there is a corresponding relationship between identification curve and discrimination peak. Meanwhile, the order to manipulate the stimuli between H–L and L–H contrast also have an impact on the perception of lexical pitch accent.

2pPP11. Binaural fusion and mis-localization of amplitude-modulated and unmodulated tone pairs across hemifields. Miles Carter (Univ. of Pittsburgh, Pittsburgh, PA 15260, mic419@pitt.edu), Stephen David, and Lina A. Reiss (Oregon Health and Sci. Univ., Portland, OR 97239)

Binaural fusion is the process by which sounds with similar features are integrated between both ears to form a unified percept. Previous studies have focused on binaural fusion of unmodulated (UM) tones. Here we describe the first measurements of binaural fusion of an amplitude-modulated (AM) tone paired with a UM tone, presented over different loudspeakers at ± 60 deg in normal-hearing listeners. Localization of the AM tone was also measured. Surprisingly, binaural fusion occurred for AM–UM tone pairs over similar ranges of frequency differences as for UM tones. In addition, when two sounds were heard for carrier frequency combinations, subjects consistently mis-localized the AM tone to the wrong hemifield. These novel phenomena suggest potential for deficits in envelope perception and localization with excess binaural fusion. [Funded by NIH/NIDCD grants R25GM134978, R01DC013307, and R01DC014950.]

2pPP12. Psychoacoustic forward masking Gaussian–Noise disruption measured across a range of hearing levels. Adam Svec (Audiol., Univ. of the Pacific, 448 N 2nd St., San Jose, CA 95112, adamsvec@gmail.com), David A. Cameron (Biomedical Eng., Univ. of Rochester, Rochester, NY), Marc Brennan (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, Lincoln, NE), William B. Salloom (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, West Lafayette, IN), Sebastian Ibarra, Estefany Castillo (Audiol., Univ. of the Pacific, San Francisco, CA), Cheng-Han Chou, Ella Lilleberg (Special Education and Communicative Disord., Univ. of Nebraska-Lincoln, Lincoln, NE), and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Recent work suggests that listeners with both normal hearing (NH) and sensorineural hearing loss (SNHL) exhibit more forward masking for a highly fluctuating Gaussian noise (GN) masker relative to a low-fluctuation noise (LFN) masker. This difference in masked thresholds, referred to as GN disruption, is affected by small differences in hearing sensitivity when measured using a brief (10-ms) probe signal and relatively short delays between masker offset and signal onset. We hypothesize that cochlear-gain control mediated by the medial olivocochlear (MOC) efferent system contributes to GN disruption. The current study used a behavioral forward-masking paradigm to measure GN disruption in human participants with hearing sensitivity ranging from normal to moderate SNHL. Results from one masker presentation level (80 dB SPL), two masker-signal delays (25 and 75 ms), and two masker types (GN and LFN) centered at 2000 Hz will be presented. This study is part of a series of experiments using physiological and otoacoustic-emission measures to assess GN disruption and the role of the MOC. Completion of these studies will improve our understanding of the mechanisms that contribute to excessive GN disruption in listeners with a range of hearing-sensitivity profiles.

2pPP13. Effect of suppressor Noise bandwidth on the suppressor tuning curves. Dharmesh Verma (Ctr. for Sophisticated Instrument Facilities, Indian Inst. of Technol. Bombay, Acoust. and Hearing Lab, Mech. Eng., IIT Bombay, Mumbai, Maharashtra 400076, India, dharmesh_verma@iitb.ac.in), Vikas K. Lakhmani (Mech. Eng., Indian Inst. of Technol. Bombay (IIT Bombay), Mumbai, Maharashtra, India), Preeti Rao (Dept. of Elec. Eng., Indian Inst. of Technol. Bombay, Mumbai, Maharashtra, India), and Sripriya Ramamoorthy (Mech. Eng., Indian Inst. of Technol. Bombay (IIT Bombay), Mumbai, Maharashtra, India)

The tuning characteristics of the basilar membrane can be assessed non-invasively using suppression tuning curves (STCs). STC measurements based on pure-tone stimuli are particularly suitable for human subjects and uncooperative patients, as the procedure does not depend on behavioural responses or sustained attention. In this work, STC experiments are performed to examine the effect of suppressor noise bandwidth on cochlear suppression. The study involves systematic variation of the suppressor noise bandwidth during STC measurements while maintaining fixed primary stimulus conditions. The suppressor bandwidths are selected with reference to the equivalent rectangular bandwidth (ERB) of the cochlear filter. Suppression tuning curves obtained under different bandwidth conditions are analyzed to characterize changes in the magnitude and shape of suppression. The analysis focuses on the tuning-curve tip and the asymmetry of low- and high-frequency slopes, which are associated with cochlear sensitivity and frequency selectivity. The results and insights from this ongoing study will be presented.

2pPP14. The impact of headphone choice on perceived externalization. Eli A. Faber (Performing Arts Technol., Univ. of Michigan, 1100 Baits Dr., Ann Arbor, MI 48109, elifaber@umich.edu) and Sile O'Modhrain (Performing Arts Technol., Univ. of Michigan, Ann Arbor, MI)

Most studies on binaural hearing utilize open-back headphones as the playback system. As consumer products that integrate binaural audio become popular, it is worthwhile to assess the spatial qualities of closed-back and insert headphones. In this study, participants evaluated the externalization of pink noise convolved with binaural room impulse responses (BRIRs) at varying azimuths using three pairs of headphones: open-back (least occluded), closed-back (partially occluded), and insert headphones (fully occluded). Externalization was evaluated on a continuous scale. Results indicate that headphone choice has a significant effect on perceived sound externalization, with externalization decreasing as occlusion increased. Additionally, low-pass filtering the stimuli did not significantly alter externalization ratings, suggesting that the impact of occlusion outweighs the loss of high-frequency pinna cues. These findings suggest that ear canal occlusion is a primary factor in the externalization of binaural audio.

2pPP15. Listener expectations about accent influence lexical competition. Byonne Atamna (Psychol. & Brain Sci., Washington Univ. in St. Louis, Brookings Dr., Saint Louis, MO 63130, a.byonne@wustl.edu) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

L2-accented speech is marked by systematic variation arising from features of speakers' first language(L1). English, for example, contains phonemic contrasts between tense and lax vowels that are not present in many languages. As a result, L2 speakers of English may merge /i/ and /I/ or /u/ and /ʊ/ in their production. The precise mechanisms underlying listeners' ability to reconcile such variation,however, remain underspecified. The present study considers how listener expectations about speakers' accent may modulate the dynamics of lexical competition during the processing of L2 speech. Using the visual world paradigm, we measure listeners' eye movements to tense-lax minimal pair words when listeners believe they are listening to either an L1- or and L2-accented speaker. Participants heard identical target words: American-accented words with a medial tense (experiment 1) or lax (experiment 2) vowel. We present data from two experiments. In experiment 1, the anticipation of an L2 accent resulted in targeted shifts in phonetic categories. Preliminary data from experiment 2 suggest this shift is directional.

2pPP16. Apparent source width revisited: A comparative study of acoustic measurements and perceived spatial impression. Renzhi Guo (Architecture, Univ. of Florida, 300 SW 13th St., Gainesville, FL 32611, guor1@ufl.edu) and Hassan Azad (Architecture, Univ. of Florida, Gainesville, FL)

Previous studies have found that interaural cross-correlation (IACC) and lateral energy fraction (LF) can serve as objective measures to predict apparent source width (ASW). However, there is a lack of literature regarding how well objective measurements correlate with subjective evaluation in a context-dependent scenario. Expanded upon a prior work, this study looked to examine the extent to which commonly assumed predictors of ASW remain valid when applied to real concert hall measurements across listener positions. ASW ratings were obtained through psychoacoustic tests employing both stereo loudspeakers and headphones to assess the perceived width of a symphony orchestra produced with different recording techniques and ensemble sizes. Key room acoustic parameters were calculated from impulse response measurements conducted in the EMPAC concert hall in Troy, NY, where the orchestral recordings were made. Results show that the existence and emergence of ASW is based on the opposite perceptual mappings between the two reproduction conditions, even though stereo loudspeakers resembled real hall listening more closely. The findings also suggest that assuming a fixed role of binaural decorrelation for enhancing ASW is impractical because ASW is governed by context-dependent object integration, where binaural decorrelation is only beneficial when it supports rather than disrupts object unity.

2pPP17. Addressing bias when estimating pooled d' for multiple cues or neurons. Daniel R. Guest (Dept. of Biomedical Eng., Univ. of Rochester, 601 Elmwood Ave. MC 5-6483, Rochester, NY 14620, daniel_guest@urmc.rochester.edu)

One topic of interest in sensory psychology and neuroscience is the sensitivity achieved by a hypothetical observer when task information is combined (or pooled) across multiple independent sources. For example, an observer might combine information provided by two independent sensory cues to make a judgement, or an experimenter might wish to characterize the sensitivity achieved by a population of independent neurons when pooling information across them. In the case of signal detection theory, the d' of an observer optimally combining information across independent sources is equal to the square root of the sum of squared individual d' values (optimal combination rule). However, naive application of the optimal combination rule to individual d' estimates leads to biased estimates of pooled d'. Here, we demonstrate this phenomenon with computer simulations and characterize how its severity depends on parameters of interest, such as sample size and sensitivity. We suggest an alternative unbiased estimator for the optimal combination rule that could be useful to experimenters working with small sample sizes ($< \sim 40$ trials) and/or difficult tasks (d' < 1), where the bias is most problematic. [Work supported by NIH R01 DC010813 and NIH F32 DC022143.]

2pPP18. Comparison of electroencephalography technologies on neural speech tracking in quiet and competing speech. Zenzele Thomas (Medical Eng., Univ. of South Florida, 4202 E Fowler Ave. PCD 1017, Tampa, FL 33612, thomasz16@usf.edu) and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Speech perception with competing talkers can be a major challenge, especially for listeners with hearing loss. Listeners use auditory grouping cues to selectively attend to speech of interest and ignore the irrelevant signals, and recent work has shown that neural measures can reveal this ability to follow speech in complex acoustic environments. In most cases, prior studies have used research-grade, high-density electroencephalography (EEG) equipment to demonstrate this phenomenon; however, there are economic and clinical barriers for using these systems that may be overcome with open-source systems. The present study investigated the functional benefits and uses of low-density, dry EEG technology as an affordable and useful alternative to research-grade wet caps in neural speech tracking designs. This study was performed in normal hearing listeners with both systems for direct comparison. Experiment 1 looked at neural speech tracking in quiet, and experiment 2 investigated neural speech tracking with competing speech streams. The present study provides a comprehensive assessment of a research-grade system with wet electrodes compared to an open-source system with dry electrodes, which may provide clearer implications for future advancements in real-world application, such as "neuro-steered" hearing devices.

2pPP19. Human subject testing of a PCB-based vibrotactile display prototype. Adeline Mock (Mech. Eng., Tufts Univ., 200 College Ave. Medford, MA 02155, adeline.mock@tufts.edu), Jonathan Bernstein (Draper, Cambridge, MA), Dave B. Miller (Mech. Eng., Tufts Univ., Medford, MA), Maijie Xiang (Mech. Eng., Tufts Univ., Somerville, MA), and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA)

Worldwide 2.2 billion people are visually impaired, yet Braille readers are limited to two rows of text, and graphical displays are expensive. A low-cost, electronically refreshable tactile display capable of conveying both text and images is needed. We report on a low-cost display that uses a printed circuit board (PCB) electrode array and a metallized Kapton membrane separated by a patterned foam spacer. Results for 2×3 and 4×4 arrays, both with a pitch of 3 mm, will be presented. Vibration amplitudes were 1 to 7 μm peak to peak over a bandwidth of 50 Hz to 5 kHz. This is perceptible to touch. A human subject study (Tufts IRB STUDY5839) was conducted with 10 unskilled participants each conducting 20 trials on a discrimination task for two different vibration patterns. Eight of the ten participants achieved an accuracy greater than 70% indicating the patterns can be discriminated by untrained participants ($N_T = 200$, $X^2(1) = 8.99$, $p = 0.0027$). The prototype demonstrates the feasibility of this approach and is scalable to large area displays at low cost. Future work will investigate the influence of applied finger load on device dynamics and scale to larger areas and finer pitch.

2pPP20. Stability in the speech system: How neural encoding consistency shapes behavioral categorization and language function. Ege Gur (Psychol. and Brain Sci., Univ. of Iowa, 340 Iowa Ave. Iowa City, IA 52242, ege-gur@uiowa.edu), Hyoju Kim, Mariah Lees, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Speech perception requires listeners to contend with highly variable acoustic input to construct stable linguistic categories. Recent work has shown that listeners differ in their *categorization consistency*—the degree to which categorization responses vary for the same stimuli across different trials. Crucially, differences in consistency predict a wide range of real-world outcomes, including language and reading ability in children and adults. However, the mechanisms underlying these differences remain poorly understood. Here, we evaluate the hypothesis that inconsistent categorization is driven by inconsistent encoding of the acoustic information at the neural level. Using electroencephalography, we measured the consistency of the N1 and P3 components in an oddball task. The auditory N1 captures early, pre-attentional encoding processes, while the P3 reflects higher level categorization judgements in the oddball design. Listeners heard steps along speech continua involving bilabial and velar stop consonants while EEG recording took place. We then predicted listeners' categorization consistency at the behavioral level using the visual analog scaling task from the consistency of their neural responses. Preliminary results suggest a relationship between the consistency of the N1 and the behavioral response, bolstering the idea bottom up noise at the neural level is responsible for differences in categorization consistency.

2pPP21. Perception of novel words in noise by cochlear implant users. Michelle Kapolowicz (Commun. Sci. & Disord., Univ. of South Florida, 4202 E Fowler Ave. Tampa, FL 33620-9951, kapolowicz@usf.edu), Vahid Montazeri (Qualcomm Inc., San Diego, CA), Dimitri Brunelle, Christopher Brown, Robert A. Lutfi (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), and Peter F. Assmann (Psych., Univ. of Texas at Dallas, Richardson, TX)

When listeners are exposed to complex auditory scenes with unfamiliar sounds, repetition provides a basis for segregating a stable source from dynamic competing sounds (McDermott *et al.*, 2011; Montazeri *et al.*, 2021) This process enables robust target recognition that persists even when novel distractors are introduced (Kapolowicz *et al.*, 2025). Here, we

investigate whether this perceptual learning mechanism is preserved in cochlear implant (CI) users, who rely on a significantly degraded auditory signal. Our results indicate that, on average, CI users perform comparably to normal-hearing listeners in these tasks. Specifically, CI users successfully extract subtle acoustic cues from repeating target sounds to distinguish them from varying distractor sounds, maintaining robust recognition as the acoustic environment changes. These findings support a generalized model of auditory scene analysis where statistical regularities in an environment allow listeners to construct stable object representations and demonstrate that this fundamental grouping process remains resilient even when the acoustic input is significantly degraded and processed through a cochlear implant.

2pPP22. Modeling interaural-time-difference lateralization perception. Kiet Tran (Elec. and Comput. Eng., Univ. of Maryland-College Park, College Park, MD) and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., 0141 Lefrak Hall, College Park, MD 20742, goupell@umd.edu)

Binaural hearing allows humans to localize sound sources in the horizontal plane using interaural time differences (ITDs) and the interaural level difference (ILDs). For complex sounds, binaural information must be processed across different frequency bands, which helps resolve ambiguous locations. One of the most discussed models that explains across-frequency ITD processing consists of three components: straightness (greater weighting of consistency of ITDs across frequency), centrality (greater ITD weighting toward the center of the head), and frequency-weighting (greater weighting near 700 Hz). This project sought to understand the different contributions of these components in explaining human perceptual data in lateralization tasks. Intracranial lateralization was measured in normal-hearing adults using headphones that presented physiological plausible and implausible ($>$750 μs) ITDs. The listeners were presented with sound sources at single pure tones, complex tones, and narrowband noise around 500 Hz. Preliminary results suggest that much of perception can be accounted for by the frequency weighting.

2pPP23. Explaining contralateral interference with selective-attention abilities in cochlear-implant and normal-hearing listeners. Milana Klopouh, Jillian Keng (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), Michael A. Johns (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Silver Spring, MD), Kylie Kelleher, Erin Catob (Hearing and Speech Sci., Univ. of Maryland-College Park, College Park, MD), Sandeep A. Phatak, Joshua G. Bernstein (National Military Audiol. & Speech Pathol. Ctr., Walter Reed National Military Medical Ctr., Bethesda, MD), and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland-College Park, 7251 Preinkert Dr., 0141 Lefrak Hall, College Park, MD 20742, goupell@umd.edu)

Bilateral (BI) or single-sided-deafness (SSD) cochlear implants (CIs) are increasingly common, producing binaural benefits like improved spatial release from masking. However, these benefits vary widely, possibly due to "bilateral speech interference," whereby speech in the second ear hinders target-ear speech recognition. We hypothesized that impaired selective attention, compounded by functional ear asymmetry, limits binaural benefits, with some CI listeners struggling to attend to degraded target speech while ignoring a less degraded masker. Seventeen BI-CI, 15 SSD-CI, and 15 normal-hearing listeners reported keywords from the coordinate response measure corpus. Binaural benefit was assessed via contralateral unmasking: target speech to one ear, same-sex masker to the same ear versus both ears. This was compared to the multi-masker penalty (performance decrease with increasing number of maskers) and to dichotic listening (performance decrease for identifying monaural target speech alone versus with a contralateral masker). The multi-masker penalty and dichotic deficit explained independent variance in contralateral unmasking, suggesting that bilateral speech interference reflects poor binaural fusion (and thus extra voices perceived in the mixture) and impaired selective attention. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy of the Department of War or U.S. Government.]

2pPP24. Listening quality perspectives from musicians who wear hearing aids. Chad Bullard (Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, charbull@iu.edu) and Jennifer Lentz (Indiana Univ., Bloomington, IN)

Many studies have explored participants' perceptions of sound and music quality through the processing of their hearing aids, however, relatively few studies have targeted the musician population with these same questions. The purpose of this study is to supplement existing work on music perception in hearing aids. We recruited musician participants with more than 5 years of music experience who wear hearing aids to participate in an interview about their perceptions of music with hearing aids and their listening habits. Preliminary qualitative thematic analysis has revealed some interesting contrasts, with differing perspectives among musicians. Some musicians reported a preference to always wear their hearing aids, whereas others only wear them for speech due to poor music processing. Similarly, some participants find their hearing aids greatly improve all music listening, but others perceive that their hearing aids distort their ability to perceive musical features like intonation and clarity. This work highlights the diversity of perspectives on hearing aids among musicians and the need for additional inquiry into the results of hearing aid processing on music perception, particularly as it pertains to quality of life for older musicians.

2pPP25. The ability of deep-learning-based noise reduction to generalize to accented-English speech. Kian Fallah (The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fallah.10@buckeyemail.osu.edu), Sarah E. Yoho, and Eric W. Healy (The Ohio State Univ., Columbus, OH 43210)

Deep learning has revolutionized noise reduction and has been implemented in the latest hearing aids. Examined here is the ability of a modern deep learning noise-reduction network to generalize to a type of speech not encountered during network training. Whereas training primarily involved native-English speech, the ability to remove background noise from speech produced by non-native speakers of English was tested. Accented-English speech from 63 individuals representing 7 non-English native languages was mixed with shaped noise at an SNR of 0 dB. The attentive recurrent network was used to remove noise, and performance was assessed using objective measures of intelligibility and sound quality as well as human intelligibility before and after noise reduction. It was found that generalization was robust overall, despite considerable acoustic differences that exist across native and accented speech. Objective measures revealed that the algorithm output was slightly more accurate when speech was less strongly accented. But these small acoustic differences were not clearly reflected in human intelligibility, suggesting that human listeners may not be sensitive to such disruptions to the signal. These results are promising for noise-reduction systems that need to operate effectively across a variety of real-world communication situations. [Work supported by NIH.]

2pPP26. Cross-dataset training for auditory attention decoding: A comparative study of canonical correlation analysis and convolutional neural network-based localization. Pooja Yakkala (College of Computing and Information Sci., Rochester Inst. of Technol., 1 Lomb Memorial Dr., Rochester, NY 14623, py9363@rit.edu), Sungyoung Kim (Meta AI Research Group, New York Univ., Daejeon, Korea), Inyong Choi (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), and Hwan Shim (Dept. of Elec. and Comput. Eng. Technol., Rochester Inst. of Technol., Rochester, NY)

Electroencephalography (EEG)-based auditory attention decoding (AAD) is a vital component of neuro-steered hearing devices designed to address the "cocktail party" problem. A significant challenge remains the effective utilization of data across diverse recording protocols and datasets. In this work, we evaluate and compare two baseline AAD approaches, canonical correlation analysis (CCA) and a convolutional neural network-based localization (CNN-Loc) method, using the public KU Leuven and DTU datasets. Notably, the CNN-Loc architecture employed in this study incorporates both spatial and temporal processing layers, differing from its original formulation. To investigate the feasibility of cross-dataset training,

we constructed a combined dataset using a unified preprocessing pipeline. This included multi-channel Wiener filtering and shared EEG channel alignment. Our findings indicate that models trained on the combined data perform comparably to or better than, those trained on individual datasets. This suggests that aggregating heterogeneous data enhance model robustness. The CNN-Loc approach also showed strong results on the DTU dataset, which is surprising given earlier reports about the limited generalization of nonlinear methods in AAD. Our results suggest that combining diverse datasets is a beneficial strategy for the field. Future work will continue exploring correlation-based decoding to improve performance across diverse recordings.

2pPP27. Relating binaural temporal fine-structure sensitivity to spatial release from speech-on-speech masking: Interactions with working memory and executive function. Amanda Bass (Speech Lang. Pathol. and Audiol., Towson Univ., 6135 Wooded Run Dr. Columbia, MD 21044, Towson, MD 210204, abass7@students.towson.edu), Allison Vanisko, Diana Maughan, Samantha Garofalo (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD), Nirmal Kumar Srinivasan (Audiol., Speech-Lang. Pathol., and Deaf Studies, Towson Univ., Towson, MD), and Chhayakanta Patro (Speech Lang. Pathol. and Audiol., Towson Univ., Towson, MD)

Children derive significantly less benefit from target-masker spatial separation than adults, a persistent deficit in spatial release from masking (SRM) despite normal audiometric thresholds. In adults, SRM is supported by suprathreshold binaural temporal fine structure (TFS) processing, yet factors underlying this developmental lag remain poorly defined. We hypothesized that immature TFS processing and top-down cognitive resources parallel and constrain pediatric speech-on-speech perception. To evaluate this, we assessed 24 children (ages 5–14) using the TFS-LF test. Sensitivity was quantified as IPD thresholds at 250, 500, and 750 Hz. SRM was calculated from speech reception thresholds in co-located and spatially separated conditions 45 deg). Cognitive profiles were established using Digit Span and the trail making test (TMT). Preliminary analyses suggest TFS sensitivity and SRM were only significantly associated in children with high executive control (low TMT scores). For children with poorer executive flexibility, TFS processing failed to translate into a spatial advantage. Additional analyses are underway to examine the contribution of working memory to this auditory-cognitive interaction. These findings will determine whether the functional integration of suprathreshold auditory processing and cognitive capacity serves as a robust metric for quantifying perceptual variability and characterizing the developmental spatial hearing gap in children.

TUESDAY AFTERNOON, 12 MAY 2026 SALON B, 1:00 P.M. TO 3:40 P.M.

Session 2pSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustic Metamaterials II

Christina Naify, Cochair
App. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Alexey Titovich, Cochair
Carderock Div., Naval Undersea Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700

Bogdan-Ioan Popa, Cochair
Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Chair's Introduction—1:00

Invited Paper

1:05

2pSA1. Nonstationary synthetic acoustic holography. Lea Beilkin (Mech. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, leabeilkin@tauex.tau.ac.il)

We address synthetic acoustic holography for the reproduction of prescribed target sound fields within designated regions of three-dimensional space using a set of spatially dispersed control sources. The objective is to synthesize driving signals such that the resulting acoustic field matches that of a maneuvering, sound-emitting target or the acoustic scattering from an object, as observed within the region of interest. Both target fields known in advance and target fields acquired in real time are considered. Email address:

1:25

2pSA2. Energy landscape in the context of a multi-tuned vibration absorber system. David Feit (4601 N Park Ave., Apt. 213, Chevy Chase, MD 20815, dvdfeit@yahoo.com)

The "Energy Landscape" framework provides a powerful way to understand the dynamics of a primary oscillator coupled to multiple absorber oscillators. The energy landscape exists in a $2(N+1)$-dimensional phase space, with position and momentum coordinates for each oscillator. The landscape has a single global minimum at its origin where all oscillators are at equilibrium. Unlike complex systems with multiple minima, this landscape is convex, but the coupling creates interesting directional features. When the primary oscillator is excited, the coupled dynamics cause the system to traverse the landscape, redistributing energy among all oscillators. In engineering applications, the energy landscape picture explains how tuned absorbers (absorber frequencies close to the primary) create efficient energy transfer channels, rapidly draining energy away from the primary structure. If time permits, a similar picture will be shown for a force excited vibrating plate radiating into an acoustic half-space.

1:40

2pSA3. Local anisotropy control of effective coupling for tunable dispersion in acoustic lattices. Taehwa Lee (Toyota Res. Inst. of North America, 1555 Woodridge Ave., ERD/TRINA, Ann Arbor, MI 48105, taehwa.lee@toyota.com)

Tunable acoustic lattices are highly desirable for controlling wave propagation and exploring emerging wave phenomena. Here, we introduce a versatile strategy for dispersion engineering based on local anisotropy control, which enables continuous modulation of effective inter-site coupling without altering the underlying physical connectivity of the lattice. By constraining the local wave field (or displacement) at each lattice site to a prescribed orientation, a site-dependent directional field vector is defined. The relative orientation between neighboring sites governs the strength and sign of their effective coupling. Because the local field orientation can be continuously tuned via a tilting angle, this approach provides broad and continuous control over the effective coupling landscape, allowing access to a wide range of dispersion relations within a single lattice architecture. Furthermore, the same framework naturally enables long-range effective coupling beyond nearest neighbors, offering a powerful platform to investigate diverse wave-physics phenomena, including topological phases, non-local interactions, and exotic band structures. This work establishes local anisotropy as a general and robust design principle for reconfigurable acoustic metamaterials.

1:55

2pSA4. Benchmarking compact language models for porous acoustic metamaterials. Jennifer L. Celeste (Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, jenniferlceleste@gmail.com) and Bhisham Sharma (Mech. and Aerosp. Eng., Michigan Tech Univ., Houghton, MI)

Following the development of compact, domain-specific language models for assisting research in porous acoustic metamaterials (presented at the ASA Fall 2025 meeting), this work focuses on establishing an evaluative framework to quantify their performance relative to general-purpose large language models (LLMs). While domain-specific models aim to bridge knowledge gaps in niche scientific domains, they often face challenges related to limited training data and the absence of standardized evaluation benchmarks. This study explores multiple evaluation strategies tailored to acoustics research, including LLM-as-a-judge approaches for automated scoring, expert human evaluation to capture scientific nuance, and standardized datasets modeled after foundational acoustics examinations. These methods are applied to tasks relevant to porous acoustic metamaterial research, such as theoretical understanding, material modeling, and interpretation of acoustic phenomena. The objective is to characterize trade-offs between model size and domain accuracy, and to identify conditions under which small-scale, domain-specific LLMs can reliably support research in porous acoustic metamaterials.

2:10

2pSA5. Prediction of acoustic transmission and kinetic energy partition in chiral metamaterials using deep neural networks. William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 9767, Austin, TX 78766-9767, william.willis@utexas.edu), Samuel P. Wallen, and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Acoustic metamaterials (AMMs) can be designed for the control of sound wave scattering and propagation for specific frequencies and angles of incidence to achieve a desired performance. Chiral AMMs, whose geometry breaks mirror symmetry, are of interest to control acoustic transmission for their ability to convert energy between translational and rotational motion. This work first considers the effects of geometry and material properties of a chiral AMM by modeling acoustic reflection and transmission through an AMM layer using finite element analysis (FEA). FEA enables analysis of the energy distribution within the AMM unit cell to determine the relative magnitudes of translational and rotational kinetic energy components to elucidate the role of translational and rotational energy partition in acoustic transmission. However, performing FEA for all possible variations in design parameters can incur prohibitively high computational costs. To address this bottleneck, the present study develops fast surrogate models to enable efficient design space exploration for chiral AMMs. Specifically, deep neural networks, which can be trained on limited FEA results, are utilized to predict transmission and kinetic energy components as functions of frequency, incidence angle, and design parameters. Necessary constraints on FEA training data are discussed, as well as associated computational expenses.

2:25–2:40 Break

2:40

2pSA6. Broadband wide-angle acoustic lens design using exact pressure gradients. Feruza Amirkulova (Mech. Engieering, San Jose State Univ., San Jose, CA), Samer Gerges (San Jose State Univ., San Jose, CA), Nathan Wesling (San Jose State Univ., 287 Mesquite Rd., Bishop, CA 93514, nathan.wesling@sjsu.edu), and Mikhail Korobkov (San Jose State Univ., San Jose, CA)

This work presents a rigorous inverse design method for broadband acoustic lenses composed of nonuniform planar configurations of cylindrical scatterers. By applying multiple scattering theory, we calculate the gradient of the absolute pressure at a predetermined focal point, effectively mapping the sensitivity of the acoustic field to variations in scatterer geometry and placement. Unlike prior heuristic optimization approaches, this method integrates these exact gradient formulas into a gradient-based optimization (GBO) algorithm to simultaneously adjust the size and location of every scatterer. The approach maximizes sound amplification at the focal point while adhering to nonlinear constraints that prevent scatterer overlap within a confined region. We demonstrate the efficacy of this method through numerical examples of acoustic lenses submerged in water, consisting of rigid, fluid, or elastic solid cylinders. The results highlight significant improvements in pressure amplification and robust focusing performance across broad frequency ranges for point source excitations from varying locations. This semi-analytical approach offers a computationally efficient pathway for designing high-performance acoustic metamaterials for sound localization and medical imaging applications.

2:55

2pSA7. Flexural Friedrich–Wintgen BICs in open elastic plate resonators. Yu-Tong Wang (Acoust., Penn State, 201 Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, ybw5392@psu.edu), Mourad Oudich (Institut Jean Lamour, Universite' de Lorraine, Nancy, France), and Yun Jing (Acoust., Penn State, State College, PA)

Bound states in the continuum (BICs) are resonant modes that remain decoupled from the surrounding radiation continuum, theoretically offering

infinite quality factors (Q). However, realizing BICs in elastic structures remains challenging due to the modal complexity of elastic waves. In this work, we present a physics-based framework for realizing elastic Friedrich–Wintgen (FW) BICs in an open plate resonator, consisting of a macroscopic rectangular aluminum plate supporting flexural waves and coupled to the surrounding elastic continuum through a opening. We show that FW BICs emerge from the suppression of radiative leakage via two mechanisms: (i) geometrical tuning of the resonator opening to coincide with a vanishing elastic displacement field at the opening, corresponding to zero effective impedance, and (ii) destructive interference between asymmetric coupled resonance modes, whose hybridization produces a non-radiating state. By varying the resonator aspect ratio, $\alpha = L/W$, these mechanisms can be continuously accessed. Experimental validation using laser Doppler vibrometry (LDV) across two frequency regimes confirms the existence of these states. In the 3-kHz range, a BIC is observed at $\alpha = 1.115$ with $Q = 591$, while in the 5-kHz range a transition at $\alpha = 0.773$ yields $Q = 1578$. The measurements agree well with numerical simulations. These results provide a robust route toward tailoring high-Q elastic resonances in open systems for narrow-band filtering and sensing applications.

3:10

2pSA8. Reproducing the page curve in classical mechanics through time varying entropy of entanglement. Abrar N. Faiaz (Mech. Eng., Wayne State Univ., Detroit, MI), Donte J. McMullen (Mech. Eng., Wayne State Univ., 30419 Brush St., Madison Heights, MI 48071, dontemcmull@gmail.com), Kazi Tahsin Mahmood, and M Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI)

Entanglement entropy is a key diagnostic of information flow in interacting many-body systems, and, although its behavior is usually regarded as inherently quantum, it is not known whether similar information-flow dynamics can arise in purely classical systems. An instance when this phenomena is in effect is reflected through the Page curve, which describes how entanglement entropy first grows and then decreases during black-hole evaporation. Here we show that a minimal, classically mechanical system can reproduce Page-curve-like behavior. Our system consists of two epoxy-bonded granular dumbbells each acting like an elastic bit, a classical version of a qubit, with its own normal modes. Each dumbbell is coupled by Hertzian contact, driven harmonically and monitored with a laser-Doppler vibrometer. We find that the measured time-dependent entropy of the two subsystems exhibits a cyclic rise and fall closely resembling the Page curve, with the amplitude and timing controlled by the drive frequency, driving amplitude, and contact geometry. This demonstrates that time-dependent entropy of entanglement does not require quantum mechanics but can emerge from deterministic nonlinear dynamics. Our results establish granular mechanical networks as a fully observable, decoherence-free platform for studying information scrambling and retrieval, and provide a tangible analogue for phenomena normally associated with black holes and quantum entanglement.

3:25

2pSA9. A dynamic homogenization framework for elastic metamaterials. Emadoddin Abdoli (Mech. & Mater. Eng., Univ. of Cincinnati, 2600 Clifton Ave. Cincinnati, OH 45221, abdolien@mail.uc.edu) and Ahmed Allam (Mech. & Mater. Eng., Univ. of Cincinnati, Cincinnati, OH)

Elastic metamaterials, including architected materials and lattice structures, are increasingly used to engineer novel material properties through geometry and topology. While significant progress has been made in their static characterization, their extension to dynamic problems, such as wave and vibration control, is limited by the prohibitive computational cost of simulating millions of unit cells in large structures. This work develops a technique to replace complex subscale geometric features with an equivalent continuum description to accelerate dynamic simulations. Using a finite-element model based on periodic unit-cell theory and Bloch wave analysis, we systematically extract full frequency-dependent stiffness and density tensors. The method maps the eigenfrequencies and vibration modes of a repeating unit cell into macroscopic properties suitable for large-scale simulations. The technique is applied to cubic lattice structures with varying geometric configurations. The results reveal pronounced frequency-dependent dispersion and anisotropy that are overlooked by static homogenization, which fails to account for micro-scale periodicity as the wavelength approaches the unit-cell size. It is then validated in large-scale simulations against high-fidelity, fully resolved geometric models of the metamaterials. Ultimately, this method enables fast and accurate modeling of ultrasonic wave propagation and paves the way for efficient, lightweight designs in vibration isolation and impact mitigation.

Session 2pSC

Speech Communication: Technology, Methods, and Computational Modeling (Poster Session)

Benjamin Munson, Chair
Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN 55455

All posters wll be on display from 1:00 p.m. to 4:30 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:45 p.m. and authors of even-numbered papers will be at their posters from 2:45 p.m. to 4:30 p.m.

Contributed Papers

2pSC1. Evaluation of source separation algorithms on infant–caregiver speech in autism and typical development. Ashlyn Hall (Grand Canyon Univ., Newnan, GA) and Gordon Ramsay (Dept. of Pediatrics, Emory Univ., Marcus Autism Ctr., 1920 Briarcliff Rd NE, Atlanta, GA 30329, gordon.ramsay@emory.edu)

Speech separation, isolating individual vocal sources from a mixed audio signal, remains a difficult challenge for artificial intelligence. Unlike machines, the human auditory system can cope with the "cocktail party effect," filtering relevant voices from background noise using a combination of spatial cues, frequency sensitivity, and contextual awareness. Despite progress in deep learning, achieving robust separation in real-world conditions continues to fall short, particularly for unexplored domains such as vocal development, where infant and caregiver vocalizations regularly overlap, or communication disorders, which exhibit atypical voice properties. In this study, addressing an important problem in vocal biomarker development, we investigated the performance of source separation algorithms for isolating infant and caregiver vocalizations, in autism and typical development. Using home audio recordings collected every month from 0–24 months from 45 infants diagnosed with autism and 100 typically developing controls, we trained existing architectures based on DCCRNet, ConvTasNet, DP-RNN, and DPTNet to separate infant and caregiver speech, and evaluated each model based on the scale-invariant signal-to-distortion ratio (SI-SDR). The ConvTasNet architecture gave the best performance, but all algorithms failed relative to adult speech, in interesting ways that suggest new biologically motivated directions for source separation research.

2pSC2. Extending unsupervised phonetic segmentation with self-supervised speech embeddings. Hahn Koo (San Jose State Univ., One Washington Square, San Jose, CA 95192-0093, hahn.koo@sjsu.edu)

Unsupervised phonetic segmentation provides a bottom-up alternative to forced alignment, enabling phonetic analyses without predefined segment labels and supporting a range of speech research scenarios where detailed phonetic annotation is unavailable or unnecessary. In previous work, I introduced a lightweight Praat-based implementation that segments speech into phone-sized units by computing framewise differences on traditional acoustic representations such as mel spectrograms and MFCCs, achieving competitive performance on benchmark data while remaining easy to install and use. In this paper, I extend that framework by incorporating self-supervised speech embeddings derived from recent deep learning models (e.g., wav2vec 2.0, HuBERT, WavLM, and Whisper) as alternative input representations. These embeddings are learned without phonetic supervision yet are known to encode contextualized phonetic structure and invariances that may not be readily captured by low-level acoustic features. This study extends prior work by comparing self-supervised embeddings with traditional acoustic representations using the same windowed difference measures and peak-based boundary detection procedures. Evaluation against manually annotated phonetic boundaries examines segmentation errors across phonetic boundary types, identifying conditions under which self-supervised embeddings exhibit different behavior while maintaining the simplicity of the original bottom-up approach.

2pSC3. A double-reed sound source for physical models of the human vocal tract. Takayuki Arai (Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

For a long time, a single-reed sound source has been used as part of our set of physical models for demonstrating human speech production. However, this type of sound source is unable to show asymmetrical patterns of vibrations as a phonation source demonstrating the voice qualities of roughness and breathiness, for instance. Physically simulating a rough or breathy voice in voice disorders is especially important for observing the source and filter interaction. Here, we designed a double-reed mechanism for our sound source settings. A simple solution to achieve this goal was to join two "tables" for reeds back to back. This allowed us to stably vibrate two sets of reeds synchronously by blowing a steady air flow into the mouthpiece. Furthermore, we could use the source to simulate an asymmetrical vocal-fold vibration by adding a small weight only on one side of two reeds or a bump on one side of the two tables. We confirmed that outcomes from a vocal-tract model with such asymmetrical double-reed sound sources had voice qualities of roughness and/or breathiness. [This work was supported by JSPS KAKENHI Grant No. 24K06423 and Sophia University Special Grant for Academic Research (Research in Priority Areas).]

2pSC4. Non invasive acoustic parameters for real time voice authentication in secure communications. Gilberto A. Perez (IT, PBO, Avenida Libertad #5, La Romana, La Romana 22000, Dominican Republic, gaperez@pbo.org.do)

Voice-based communication systems are increasingly exposed to spoofing attacks driven by synthetic speech and voice cloning technologies. Conventional voice biometrics rely on storing speech recordings, raising concerns related to privacy, scalability, and forensic handling of sensitive data. This work explores a non invasive acoustic authentication framework based on the extraction and comparison of short-term acoustic parameters without retaining raw voice content. The proposed approach analyzes brief speech segments captured during live communications and derives a compact set of acoustic descriptors, including formant distribution patterns, fundamental frequency behavior, temporal pause characteristics, and spectral timbre indicators. These parameters are encoded as lightweight textual representations and compared against adaptive reference profiles associated with communication endpoints. Unlike traditional speaker recognition systems, the method focuses on dynamic acoustic consistency rather than explicit identity attribution. Preliminary evaluations indicate that the framework can differentiate natural speech from replayed and synthetically generated voice signals under controlled conditions while maintaining low computational overhead. The results suggest that parameter based acoustic validation can function as an additional trust layer for secure voice

communications, forensic screening, and real-time risk assessment in environments affected by emerging voice manipulation technologies.

2pSC5. Using machine learning modeling to analyze neural development of music processing in children with different language backgrounds. Elliot S. Blin (The Bronx High School of Sci., 2077 5th Ave. Apt. 5C, New York City, NY 10035, elliotblin2509@gmail.com), Anita Hyseni, Michael Yushavaiev, Benjamin Lee, Elizabeth Dickey (St. John's Univ., New York City, NY), and Yan H. Yu (Commun. Sci. & Disord., St. John's Univ., Bayside, NY)

Both music experience and language experience influence auditory processing in the developing brain. However, it remains unclear how bilingual experience specifically affects music processing and its underlying neural mechanisms. The primary objective of this study was to examine the neural network involved in music processing in children and to explore how music and language experience interact at the cortical level during development. Children between the ages of five and ten from Mandarin, Spanish, French, Standard American English, and African American English speaking households participated in a passive listening oddball task that included six types of musical changes, including intensity, pitch, rhythm, timbre, slide, and location. Neural responses were recorded using event-related potentials from 65 scalp sites. Brain connectivity was analyzed across frequency bands between 1 and 40 Hz. Functional connectivity measures were used as input features for machine learning classification models to distinguish children by language background. Support vector machine, random forest, and elastic net methods were applied, and model performance was compared across acoustic features. The results showed that machine learning approaches successfully captured differences in neural connectivity patterns, providing insight into how language background shapes music processing networks in the developing brain.

2pSC6. Neural networks for speech separation in infants. Ashlyn Hall (Emory Univ., 17 Forest Cir, APT 606, Newnan, GA 30265, ashlynhall1204@gmail.com) and Gordon Ramsay (Dept. of Pediatrics, Emory Univ., Atlanta, GA)

Speech separation, isolating individual vocal sources from a mixed audio signal, remains one of the hardest challenges in artificial intelligence. Despite progress through deep learning and signal processing, achieving robust separation in real-world conditions continues to fall short, particularly for infant vocalizations that may provide insights into early identification of autism spectrum disorder (ASD). Unlike machines, the human auditory system can filter relevant voices from chaotic noise using a combination of spatial cues, frequency sensitivity, and can go for contextual awareness. This is known as the "cocktail party effect," so we seek to explore new ways to model sound separation in this study to see if it can be potential use for applications in infant monitoring ASD. First is to use existing architectures such as DCCRNet, ConvTasNet, DP-RNN, and DPTNet to find which is better in separate overlapping sources and suppress background noise. These models were trained on recordings of infants from Marcus Autism Center and seeing if these models can get a high positive scale-invariant signal-to-distortion ratio (SI-SDR) of the accuracy of distinct babbling or more speech-like. With the next work on improving performance by building on the ConvTasNet architecture, which had the best results and may help reveal meaningful acoustic differences between typical and disordered speech.

2pSC7. Effect of formant tuning on the voice source and vocal fold contact pressure in a computational model of voice production. Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California Los Angeles, 1000 Veteran Ave. 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Source–filter interaction, in general, has small effect on vocal fold vibration and vocal fold contact pressure in the frequency range typical of speech. However, this effect may become noticeable during formant tuning when a vocal tract resonance becomes close in frequency to either the first or second harmonic of the voice source spectrum, as often in classical or music theater singing. In this study, the effect of formant tuning on the voice source was investigated in a three-dimensional computational model of voice production. Simulations were performed with parametric variations in both vocal fold and vocal tract configurations. Conditions of strong formant tuning to the first or second harmonic were identified. Acoustic, aerodynamic, and vibratory measures of the voice source and peak vocal fold contact pressure at these conditions were then compared to those at conditions of weak formant tuning. The results are discussed with implications to vocal health and voice training. [Work supported by NIH.]

2pSC8. Optimizing linear prediction order for formant analysis of speech. Joshua Sims (Linguist, Indiana Univ., 2461 S Woolery Mill Dr., Bloomington, IN 47403, jodasims@iu.edu)

Linear predictive coding (LPC) is used in phonetics to measure formant frequencies of speech, but these formant readings are prone to error. Measured formants are attracted to F0 harmonics (Klatt, 1986; Monsen and Engebretson, 1983). Furthermore, different speakers or segments may require different analysis parameters (Escudero *et al*., 2009), due in part to changes in vocal tract length owing to physiology, larynx height, and lip rounding. Common methods for minimizing error in formant readings include adjusting parameters ad hoc, discarding outliers, finding parameters which minimize variation (Escudero *et al*., 2009), produce the smoothest contours (Barreda, 2021), or realistic formant and bandwidth values (Nearey *et al*., 2002). In autoregressive methods, an LPC of order *n* is computed from a filter of order *n-1* to minimize prediction error (see Vallabha and Tuller 2002). The trend in this error over successive orders yields a heuristic for LPC order fit. In this paper, I propose that the optimal LPC order for a given frame is that which optimally reduces prediction error over its inferior order. I evaluate this proposal with formant measurements of vowels produced by Klatt synthesis (Klatt & Klatt, 1990).

2pSC9. Implementing the UCLA voice synthesizer as a jsPsych plugin for web-based voice perception research. June M. Contreras (Speech Lang. and Hearing Sci., Univ. of Texas at El Paso, 3532 Keltner, El Paso, TX 79904, jmcontreras2@miners.utep.edu)

The UCLA Voice Synthesizer serves as an important tool in voice research, offering precise control over acoustic parameters to model both normal and pathological voices. The synthesizer's architecture enables precise manipulation of acoustic parameters—including glottal source, noise, pitch, loudness, and vocal tract resonances—supporting both research and clinical applications such as gender-affirming voice therapy. However, its current implementation is constrained by limited flexibility for browser-based experimentation on online platforms such as Prolific. This abstract proposes a port of the UCLA Voice Synthesizer as a jsPsych plugin that loads the UCLA Voice Synthesizer modules compiled as WebAssembly. jsPsych's plugin framework allows researchers to design custom plugins for use in online experiments; this project compiles the original C/C++ UCLA Voice Synthesizer code to WebAssembly using Emscripten, loading it with jsPsych as a plugin. The poster will include a QR code to a web demo of the plugin for participants to experiment with.

2pSC10. Exploring (psycho)linguistically informed approaches to ASR dataset development for minoritized language varieties. Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT), Shannon Barrios (Kahlert School of Computing, Univ. of Utah, 255 S. Central Campus Dr., LNCO Bldg., Ste. 2300, Salt Lake City, UT 84112, s.barrios@utah.edu), Neal Patwari, and Maneesha Rani Saha (Kahlert School of Computing, Univ. of Utah, Salt Lake City, UT)

Automatic speech recognition (ASR) systems have become ubiquitous, and they provide an increasingly unavoidable means for human–computer interaction. Unfortunately, they typically perform better for speakers of historically favored languages (Sprugnoli *et al*., 2017) and accents (Markl, 2022), with particularly poor performance for so-called "non-native" accents (DiChristofano *et al*. 2024). Our goal is to mitigate the biases that permeate ASR by exploring psycholinguisticallyinformed methods for dataset construction by examining the generalizability of listener and task factors that are known to modulate transcription accuracy in laboratory studies in contexts designed to resemble crowdsourced ASR dataset development. Over several experiments we have found mixed support for the hypotheses

that transcription accuracy will be higher when: (1) talkers and listeners have similar linguistic backgrounds; (2) listeners self-identify as familiar with the talker*et al*. 2024). Our goal is to mitigate the biases that permeate ASR by exploring psycholinguistically-informed methods for dataset construction by examining the generalizability of listener and task factors that are known to modulate transcription accuracy in laboratory studies in contexts designed to resemble crowdsourced ASR dataset development. Over several experiments we have found mixed support for the hypotheses that transcription accuracy will be higher when: (1) talkers and listeners have similar linguistic backgrounds; (2) listeners self-identify as familiar with the talker's accent; and (3) audio recordings are presented in single-talker versus mixed-talker blocks.

2pSC11. Personal synthetic voices for children using disentangled speaker embeddings. Jason Lilley (Ctr. for Pediatric Auditory & Speech Sci., Nemours Biomedical Res., 1701 Rockland Rd., P.O. Box 269, Wilmington, DE 19803, lilley@asel.udel.edu) and H. Timothy Bunnell (Res., Nemours Children's Health, Wilmington, DE)

Methods now exist for creating natural-sounding synthetic "voices" that mimic an articulate individual's voice from audio samples. Based on this technology, creation of personal synthetic voices for use in speech generating devices (SGDs) by people with neurodegenerative conditions has become commonplace; services now exist to help clients record their speech before it becomes heavily degraded. Children who must use SGDs for communication may also benefit from a personal synthetic voice. However, such children typically cannot produce the articulate samples needed to train the underlying models for text-to-speech systems. In previous work, we adapted a multi-speaker model trained on articulate speech to the speech of dysarthric children. The approach suffered from the fact that the model tended to adapt not only the desired properties of the target's speech (timbre) but also undesired properties (articulation and prosody). For this work, we develop new models that uses disentangled speaker embeddings to model timbre separately from undesired properties. After training this model on a diverse set of articulate speakers, we produce synthetic voices for children with speech disorders using their own vocalizations as adaptation material. We present preliminary results in terms of intelligibility, naturalness, and resemblance to the target speaker using samples evaluated by listeners.

2pSC12. Automatic scoring of speech intelligibility in children with cleft palate. Jason Lilley (Ctr. for Pediatric Auditory & Speech Sci., Nemours Biomedical Res., 1701 Rockland Rd., P.O. Box 269, Wilmington, DE 19803, lilley@asel.udel.edu), Linda Vallino, and H. Timothy Bunnell (Ctr. for Pediatric Auditory & Speech Sci., Nemours Biomedical Res., Wilmington, DE 19803)

Recent technology has the potential to standardize and quantify speech intelligibility ratings. Automatic speech recognition provides an objective means by which a standard speech test can be administered and automatically evaluated. Our aims were to (1) automate scoring of the Children's Speech Intelligibility Measure (Wilcox and Morris, 1999), a single-word task, using ASR technology and (2) compare the accuracy of two ASR technologies: hidden Markov models (HMMs) and deep neural networks (DNNs), both adapted to assess intelligibility by identifying a response word from 12 alternatives as a human listener would. The measure was administered to 66 school-aged children with repaired cleft palate (CP) and 12 controls. Each child's responses were judged by 3 adult listeners unfamiliar with CP speech (39 listeners total). A response was deemed "intelligible" if at least 2 of the 3 judges correctly identified the stimulus word; the child's final score was the number of "intelligible" words. Separately, we trained HMM and DNN systems to identify word intelligibility. The HMM and DNN systems were 83.0% and 84.2% accurate, respectively, at agreeing with the human consensus of a response's intelligibility. Correlations between judges' final scores and ASR systems were 0.853 (HMMs) and 0.895 (DNNs). [Work funded by ASHF.]

2pSC13. Detecting coda cluster epenthesis in dialectal Arabic with forced alignment. Rachel Meyer (Linguist, Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611, rmeyer2@ufl.edu)

In modern standard Arabic (MSA), any root consonants may surface as a cluster, regardless of sonority. However, many Arabic dialects require cluster epenthesis to prevent word-final sonority rises. Although this process has been described for many dialects (Almihmadi, 2023), this project analyzes cluster environments phonetically through a customizable forced alignment model. Data were taken from a corpus of subtitled YouTube videos across a broad range of Arabic dialects, labeled by country (Al-Fetyani *et al*., 2023). A pronunciation dictionary was generated by compiling pronunciations from CAMeLTools, a Python package for NLP in Arabic (Obeid *et al*., 2020). The dictionary was then modified in Python to account for phoneme–grapheme correspondence differences between MSA and dialectal Arabic (Habash *et al*., 2018). Additionally, case, definiteness, and mood suffixes used in MSA are largely lost in dialectal varieties and can be filtered out of the dictionary accordingly (Haddad, 2023). Once the dictionary has been generated, an acoustic model will be trained using the Montreal Forced Aligner's *train* function and then applied to the data (McAuliffe *et al*., 2017). We predict that although CC# (cluster) vs. CVC# (epenthesis) is often seen as binary, actual epenthesis rates are variable and dependent on the specific cluster.

2pSC14. Comparison of time-domain vocal tract acoustics models. Michael H. Krane (Appl. Res. Lab., Penn State Univ., University Park, PA 16802, mhk5@psu.edu)

Given the inherent nonlinearities in the larynx, voice modeling is best performed in the time domain. However, vocal tract acoustic behavior is most concisely and conveniently described in the frequency domain. Two time-domain vocal tract acoustic models are tested and compared. The first uses a "wave reflection" description, coupled to a time-domain mouth reflection/transmission model. This model is described by a difference equation relating mouth volume flow and right-and left-running wave pressures at the mouth. The second model uses a combination of a) pole-fitting to derive a time-domain filter for the vocal tract acoustic response, and b) recursive time-updating. The two models are tested by exciting each model with identical time domain volume flows, then computing the frequency response. This response is then compared in the frequency domain to the analytical expression for the frequency response. Then, both models are coupled to a low-order vocal fold aeroelastic-aeroacoustic model, and differences in vocal fold vibration and acoustic output are reported.

2pSC15. Exponential averaging of Noisy embeddings for phoneme-aware speech enhancement. Shuubham Ojha (Elec. and Comput. Eng., Univ. of Maryland College Park, 3180, AV Williams Bldg., 8223 Paint Branch Dr., College Park, MD 20742, sojha1@umd.edu) and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, College park, MD)

We present a phoneme-aware speech enhancement (SE) approach that boosts perceptual quality and intelligibility in adverse noise by leveraging temporally smoothed speech embeddings. Frame-level features from a pretrained self-supervised model are aggregated via exponential smoothing. This enhances robustness of diffusion-based SE model to harsh noise while preserving phonetic structure by biasing models toward linguistically plausible reconstructions. Smoothed embeddings are injected at the U-Net bottleneck via feature-wise modulation, enabling global phonetic context to guide enhancement without disrupting local time–frequency processing. Conditioning at encoder layers led to overly aggressive speech suppression, motivating bottleneck-only integration. Evaluation used noisy LibriSpeech across SNRs -3 to 12 dB and multiple noise types. Compared to an unconditioned baseline, the method consistently improves perceptual quality (PESQ) and intelligibility (STOI). For a 128-channel configuration, PESQ rises from 1.26 to 1.79 at -3 dB and 1.86 to 2.77 at 12 dB; STOI increases 0.73 to 0.88 and 0.95 to 0.98, respectively. Similar trends hold for compact models. For a 32-channel configuration, PESQ rises from 1.32 to 1.37 at -3 dB and 2.24 to 2.40 at 12 dB. Results show that temporally smoothing phonetic embeddings is an effective, computationally efficient strategy for robust diffusion-based SE in challenging acoustic conditions.

2pSC16. Signal properties of amplified speech outdoors. Matthew C. Kelley (English, George Mason Univ., George Mason University, 4400 University Dr., Fairfax, VA 22030, mkelle21@gmu.edu), Adam Cutter, Susan Kelly, Jordan Kidd, Giulia M. Soldati, and Janalyn A. Miklas (English, George Mason Univ., Fairfax, VA)

Phonetics is replete with studies about speech in quiet indoor environments. However, many speech events occur outdoors. Additionally, in the modern era, they are frequently amplified. The present study examines signal properties of amplified speech in an outdoor setting. Four speakers read a subset of the Harvard sentences over a loudspeaker using a wireless handheld microphone in a quiet soccer field outdoors in late summer of 2025 on a sunny day. Recordings for each speaker were taken simultaneously at 4, 8, and 16 m away from the loudspeaker, in addition to a recording directly from the microphone. Intensity measurements were taken using a sound-level meter several times at each distance during each session. Loudspeaker volume was adjusted for each speaker to produce an average intensity of approximately 95 dB measured at 1 m away. The overall intensity drop as distance increases will be measured for each sentence, which is predicted by the inverse-square law to be approximately 6 dB each time the distance doubles. The signal-to-noise ratio will also be estimated for each sentence. These and other quantities will be interpreted in relation to speech intelligibility in outdoor settings.

2pSC17. Noise robustness of self-supervised speech models compared to human listeners. Joselyn Rodriguez (Univ. of Maryland, 1407 D Marie Mount Hall, College Park, MD 20742, jrodri20@umd.edu), Ahmed A. Attia, and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, College park, MD 20742)

Modern self-supervised learning models have shown great advancement in applications in automatic speech recognition. However, they still struggle in noisy environments. A landmark work by Miller and Nicely investigated the errors of human perception under different levels and kinds of noise, shaping our understanding of human perception. In this work, we investigate how close machine perception is to human perception by recreating this study with two self-supervised speech recognition models, Wav2Vec 2.0 and WavLM. We compare the results of the models and humans qualitatively through classification accuracy scores and hierarchical clustering models. In a classification task, we find that models perform worse at phone classification relative to human listeners across signal-to-noise ratios (SNR). Notably, the models' performance lags behind humans by $\sim$12dB illustrating the gap in performance between humans and models. In examining the pattern of errors for specific consonants, we found that voiceless stop confusions are similar across humans and models—primarily across place, rather than voicing. We also find high accuracy for nasals across SNRs relative to other phones, consistent with human findings as well. However, there are still substantial differences in performance between human and model classification.

2pSC18. A hybrid WavLM-SLM architecture for speaker recognition and linguistic reasoning in sylheti: A low-resource approach. Pooja Gambhir (Amity Inst. of Information Technol., Amity Univ., AIIT Dept. I1 Block third Fl., Amity University Noida, Noida, Uttar Pradesh 201301, India, pgambhir@amity.edu), Shyam S. Agrawal (Electronics and Commun., KIIT Group of Colleges, Gurugram, Haryana, India), and Amita Dev (AKGEC, Ghaziabad, Uttar Pradesh, India)

Sylheti is an under-resourced Indo-Aryan language with distinct phonological features that differentiate it from standard Bengali, yet it lacks robust speech processing tools. This study presents a speech–language–reasoning framework designed to optimize speaker recognition and contextual interpretation using a limited dataset of 16 native speakers. The methodology employs a two-stage hybrid architecture. First, a pre-trained WavLM-Base+ model is fine-tuned to extract speaker-specific acoustical embeddings, capturing nuanced phonetic shifts characteristic of Sylheti. Second, these embeddings are integrated with a small language model (SLM), utilizing LoRA-finetuning to process paired audio-transcript data. This approach allows the system to not only identify the speaker but also to reason over the linguistic context of the utterance. Preliminary evaluation yields a speaker identification accuracy of 92.4%. The integration of the SLM resulted in a WER of 22.1%, significantly outperforming baseline multilingual models. Furthermore, the model demonstrates high semantic fidelity, accurately categorizing speaker intent in 89% of test cases through zero-shot reasoning. These results demonstrate that leveraging cross-modal transfer learning from high-resource languages significantly mitigates the constraints of small-scale datasets. This research provides a scalable template for developing sophisticated speech systems for other endangered/low-resource languages, contributing to the broader field of phonetics and speech communication.

2pSC19. Acoustic analysis of a vocal-tract model actuated by a robotic arm: Relationship between vocal-tract configurations and formants. Kazuha Fukunaga (Graduate School of Sci. and Technol., Sophia Univ., 7-1, Kioicho, Tokyo Chiyoda 102-8544, Japan, k-fukunaga-7s4@eagle.sophia.ac.jp), Shinya Tsuji (Information and Commun. Sci., Sophia Univ., Chiyoda-ku, Tokyo, Japan), Ryohei Suzuki (Graduate School of Sci. and Technol., Sophia Univ., Tokyo, Japan), Takayuki Arai (Information and Commun. Sci., Sophia Univ., Chiyoda-ku, Tokyo, Japan), and Shunichi Ishihara (College of Asia and the Pacific, School of Culture, History & Lang., Dept. of Linguist, Australian National Univ., Canberra, Australian Capital Territory, Australia)

A physical model of the vocal tract is utilized for investigating speech-production mechanisms. Although some vocal-tract configurations can be manipulated, such operation is limited in terms of the quantity, reproducibility, and precision of controls and measurements. Here, we developed an automated system incorporating a robotic arm to manipulate a model consisting of six blocks and conducted exhaustive acoustic measurements. Our aim was to quantify the relationship between vocal-tract configurations and formant frequencies through collecting vowel samples generated using a single fixed-source signal. We systematically displaced each block in 6-mm increments, generating four configurations per block, totaling 4096 configurations, and crucially including configurations unattainable by humans, and extracted the first two formant frequencies (F1, F2). We evaluated the model's ability to reproduce human vowels and the influence of block displacement on formants. The results show that the model covers most of the human vowel space on the F1–F2 plane. The results of this study quantitatively characterize the complex mapping between physical geometry and acoustic output, providing a foundational dataset regarding the mechanical constraints and capabilities of the vocal-tract model. [This work was supported by JSPS KAKENHI Grant No. 24K06423 and Sophia University Special Grant for Academic Research (Research in Priority Areas).]

2pSC20. A hybrid learning framework for speaker verification in elementary classroom environments. Saba Tabatabaee (Univ. of Maryland, AV Williams Bldg, 8223 Paint Branch Dr., College Park, MD 20740, sabatb@umd.edu), Jing Liu (Univ. of Maryland, College Park, MD), and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, College park, MD)

Speaker verification in elementary classrooms is challenging due to children's speech characteristics, non-stationary background noise, and overlapping speech. This study investigates the development of a robust speaker verification model based on real-world classroom recordings of students in grades 4 through 6. We analyze a dataset collected from 18 mathematics classrooms, comprising between 6 and 40 lessons per classroom, with approximately 200 h of audio and more than 55 000 utterances. Of this corpus, approximately 188 h lack speaker identity annotations, while only 12 h are labeled, consisting of one annotated lesson per classroom. In addition, speaker enrollment was conducted using 30–60 s of read speech from both students and teachers, yielding approximately 4 h of enrollment data. To address limited labeled data, we propose a hybrid speaker verification framework that first leverages self-supervised learning on unlabeled classroom recordings to learn speaker representations, and then fine-tunes these embeddings with supervised learning on the labeled subset. We evaluate the proposed hybrid approach against embedding models trained using supervised-only and self-supervised-only methods,

and further compare the more traditional ECAPA-TDNN model with the WavLM-Large-based embedding model under realistic classroom conditions. [Work supported by Grand Challenge Award at the University of Maryland.]

2pSC21. Do large Language model-based automatic speech recognition models outperform domain adapted Wav2vec models under low resources? Ahmed A. Attia (Inst. for System's Res., Univ. of Maryland, A.V. Williams Bldg, 8223 Paint Branch Dr., College Park, MD 20740, ahmadadelattia@gmail.com) and Carol Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, College park, MD)

LLM-centric approaches have gained attention in automatic speech recognition (ASR), motivated by their scale, world knowledge, and ability to natively incorporate external context through prompting. This raises an important question for low-resource domain-specific ASR: can LLM-based ASR systems outperform domain-adapted ASR models in low-resource environments like classrooms? We conduct an exploratory comparison between a modern multimodal LLM and a domain-adapted Wav2vec2.0 model for classroom speech recognition. We evaluate off-the-shelf and parameter-efficiently fine-tuned LLM-ASR using Qwen–Omni2.5, and test contextual prompting using classroom metadata. Performance is compared against a Wav2vec2.0 model adapted through continued pretraining, weakly supervised pretraining, and supervised fine-tuning. On a standard test set of transcribed classroom recordings, the domain-adapted Wav2vec2.0 model achieves a word error rate (WER) of 13.51%, outperforming the LLM-based system under all configurations (22.4% with LoRA and contextual biasing, 25.1% with LoRA only, 27.02% with contextual biasing only, and 33.2% off-the-shelf). These results suggest that, while contextual prompting is a promising and cheap domain adaptation method, domain-specific adaptation remains critical for low-resource settings. Ongoing work explores how such adaptation strategies can be integrated into future LLM-centric ASR systems. [Work supported by the Grand Challenge Award at the University of Maryland and the Bill and Melinda Gates Foundation.]

2pSC22. Comparing methods for measuring average speaking fundamental frequency. Line Lloy (Commun. Sci. and Disord., New York Univ., 261 23rd St., Unit 2, Brooklyn, NY 11215, mal9792@nyu.edu) and Susannah Levi (Commun. Sci. and Disord., New York Univ., New York, NY)

An individual's average speaking fundamental frequency (F0) has important social significance, as listeners rely on F0 to infer assumed talker gender. As such, average F0 is widely analyzed across different fields. However, there is a lack of consistency in how average F0 is measured and reported. The current study explores two approaches to measuring F0 within a dataset of men ($n = 44$), women ($n = 50$), and non-binary ($n = 71$) talkers. The first approach is common within the linguistics and clinical speech science literature, easilyimplemented, and takes measurements from Praat using an automatic pitch extraction method. The second approach is novel and aims to maximize accuracy by leveraging the advantages of multiple pitch algorithms (Praat, Snack, and Straight), but requires additional data cleaning. In brief, each algorithm simultaneously extracts pitch estimates every 5 ms. Then all points estimated at 0 Hz (for any algorithm) and the top and bottom 5% of estimates are removed as a way to eliminate tracking errors. We will discuss the extent to which these different methods differ and the pros and cons of each. Our goal is to suggest best practices for accurately and reliably measuring average F0 in large datasets.

2pSC23. Abstract withdrawn.

2pSC24. Leveraging automatic forced alignment of reading passages in dysarthria. Thea Knowles (Communicative Sci. and Disord., Michigan State Univ., 103D Cary Hall, South Campus, University at Buffalo, Buffalo, NY 14215, theaknow@buffalo.edu), Maura Philippone (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Maria Cuervo Cano (Steinhardt Dept. of Communicative Sci. and Disord., New York Univ., New York City, NY), Hung-Shao Cheng (Univ. of Wisconsin-Madison, Madison, WI), Jeffery Berry (Speech Pathol. and Audiol., Marquette Univ., Milwaukee, WI), and Adam Buchwald (Steinhardt Dept. of Communicative Sci. and Disord., New York Univ., New York, NY)

Manual acoustic-phonetic segmentation of dysarthric speech is challenging due to the degraded nature of the acoustic signal. The Montreal Forced Aligner (MFA) is a gold-standard approach for automatic segmentation in non-disordered speech, but its efficacy for speakers with dysarthria has not been evaluated. We evaluate the MFA's performance on vowel segmentation in a standardized reading passage produced by five talkers with dysarthria and five healthy controls. MFA input includes speech audio and an orthographic transcript and uses pre-trained acoustic models and pronunciation dictionaries to time-align word and phone boundaries. We manipulate three alignment conditions that differ in the amount of pre-processing a researcher may choose to perform on their input transcripts, namely (1) the full text of the reading passage, (2) the passage segmented into utterances, and (3) the passage segmented into utterances, with speech errors corrected in the text. We compare the force aligned output with manually segmented vowels as a function of condition, speaker group, and vowel. Alignment across all conditions was high ($> 90\%$ accuracy) for controls, but varied widely for dysarthric speech. Alignment accuracy for dysarthric speakers ranged from 20% (full-text) to 76.4% (segmented-corrected-text). Findings will guide best-practices for leveraging automatic acoustic tools in disordered speech research.

2pSC25. Using accent variability to probe the performance of the Whisper automatic speech recognition system. Yumeng Shen (Cognit. Sci. Dept., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, yus099@ucsd.edu), Oishani Bandopadhyay (Computational Social Sci. Dept., Univ. of California, San Diego, La Jolla, CA), and Sarah Creel (Cognit. Sci. Dept., Univ. of California, San Diego, La Jolla, CA)

Automatic speech recognition (ASR) systems often achieve high accuracy for native speech, yet remain less reliable for non-native (L2) accented speech. This gap raises a question about why ASR performs so well on L1 speech. When acoustic cues diverge from expectation, does ASR accommodate L2 speech acoustics, or does it rely on semantic predictability to infer likely words? To separate acoustic sensitivity from semantic inference, this study uses Whisper to probe how model size, semantic context, and talker-level variability influence transcription accuracy under accent-related variability. Five Whisper models (tiny, base, small, medium, and large-v3) were used to transcribe 200 read sentences, half high-predictability and half low-predictability, produced by 24 L1 Mandarin speakers of English and 24 L1 American-accented English speakers. We characterize each talker's vocabulary knowledge, accent exposure, and perceived accentedness. Mixed-effects analyses of transcription accuracy will test three predictions. First, increasing model size will predict higher accuracy for both L1 and L2 speech, with an outsized effect for L2 speakers. Second, high-predictability

sentence context will predict higher target-word accuracy, with a larger benefit for L2 speech. Third, continuous measures of lexical proficiency, prior accent exposure, and perceived accentedness will explain transcription accuracy beyond a categorical L1–L2 distinction.

2pSC26. Interpreting mel frequency cepstral coefficients for frication on the Turkish rhotic. Janalyn A. Miklas (English, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, jmiklas@gmu.edu) and Matthew C. Kelley (English, George Mason Univ., Fairfax, VA)

Canonically, the "r" in Turkish is produced as an alveolar tap; however, word-finally, this tap has been described as devoiced and fricated. Previous work found that mel frequency cepstral coefficients (MFCCs) show spectral differences between canonical and fricated taps in Turkish. Furthermore, work on other languages found MFCCs to be better at discriminating places of articulation for fricatives than spectral moments. Although MFCCs are useful in speech recognition and acoustic research, their phonetic interpretations are still unclear. Current work has been done to interpret MFCCs for vowels in American English [Iskarous and Vietti, J. Phonetics **112** (2025)]; however, more work is needed to interpret MFCCs for consonants. MFCCs can be interpreted as providing information about the concentrations of energy in the spectrum, which is useful for identifying fricatives. Using the Turkish dataset from the Mozilla Common Voice corpus, this study takes a closer look at the significant MFCCs for distinguishing fricated and non-fricated rhotics in Turkish. The results will be discussed with respect to how MFCCs might be phonetically interpreted for frication, with a specific focus on the Turkish rhotic.

2pSC27. Evaluating speech-to-text systems with PennSound. Jonathan Wright (Linguistic Data Consortium, Univ. of Pennsylvania, 3600 Market St., Ste. 810, Philadelphia, PA 19104, jdwright@ldc.upenn.edu), Mark Liberman, Neville Ryant, and James Fiumara (Linguistic Data Consortium, Univ. of Pennsylvania, Philadelphia, PA)

A random sample of nearly 10 h of speech from PennSound, the world's largest online collection of poetry readings and discussions, was used as a benchmark to evaluate several commercial and open-source speech-to-text systems. PennSound's wide variation in recording conditions and speech styles makes it a good representative for many other untranscribed audio collections. Reference transcripts were created by trained annotators, and system transcripts were produced from AWS, Azure, Google, IBM, NeMo, Rev.ai, Whisper, and Whisper.cpp. Based on word error rate, Rev.ai was the top performer, and Whisper was the top open source performer (as long as hallucinations were avoided). AWS had the best diarization error rates among three systems. However, WER and DER differences were slim, and various tradeoffs may motivate choosing different systems for different end users. We also examine the issue of hallucinations in Whisper. Users of Whisper should be cautioned to be aware of runtime options, and whether the speed versus accuracy trade off is acceptable.

2pSC28. A Bayesian account of perceptual asymmetries in confusion matrix data. John McGahay (Linguist, UCLA, 335 Portola Plaza, Los Angeles, CA 90095, jmcgahay@g.ucla.edu)

Confusion matrices encoding the rate at which listeners confuse phonemes for each other in experimental settings exhibit perceptual asymmetries, e.g., English speakers misperceive /θ/ as /f/ at a higher rate than they misperceive /f/ as /θ/ (Miller and Nicely, 1955). A Bayesian account of phonetic categorization (cf. Clayards *et al.*, 2008) taking into acount prior probabilities and acoustic likelihoods suggests one explanation for such asymmetries; since /f/ is more probable than /θ/ in English, English speakers should demonstrate a prior bias favoring perception of /f/. However, many observed perceptual asymmetries appear inconsistent with such an account, e.g., /s/ is more often misperceived as /ʃ/ than vice versa despite its higher probability in English. I will present results testing the hypothesis that an acoustic bias toward /ʃ/ results from the fact that white noise or multi-speaker babble added to generate confusion matrices includes mid-range frequencies more strongly present in /ʃ/ than /s/. Such results suggest that a Bayesian account can be salvaged; perceptual asymmetries may result from either prior bias or asymmetrically distributed acoustic likelihoods. This work has the implication that one should take caution in applying phonetic similarity measures derived from confusion matrices to contexts with lower signal-to-noise ratios.

2pSC29. Analysis of the radiation characteristics of vowel /a/ produced by a vocal-tract model. Shiori Matsumoto (Sophia Univ., 7-1, Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, s-matsumoto-3g3@eagle.sophia.ac.jp), Runa Tanaka, Shinya Tsuji, Takayuki Arai (Sophia Univ., Chiyoda-kuTokyo, Japan), Takehiro Sugimoto, and Kotaro Kinoshita (NHK, Setagaya-ku, Tokyo, Japan)

Achieving realistic auditory experiences in "immersive" content requires modeling the radiation characteristics of human speech. However, measurements of human speakers are affected by inter-individual differences and variations within individual speakers during speech production. In this study, an experimental setup consisting of a vocal-tract model with a simulated human head and torso was used. The three-dimensional spatial radiation characteristics of the vowel /a/ at fundamental frequencies of 130 and 170 Hz were measured using a 186-channel microphone array. In addition, we measured a male speaker producing the vowel /a/ at the same fundamental frequencies using the same equipment. As a result, the vocal-tract model showed that intensity levels decreased toward the rear in the two-dimensional horizontal plane, with greater attenuation at higher frequencies. Analysis of the three-dimensional spatial radiation characteristics revealed a downward radiation tendency in 500–800 Hz and an upward radiation tendency in 1000–1600 Hz. These trends were consistent with those observed in the speaker. In contrast, above 2000 Hz, the vocal-tract model exhibited an upward radiation, whereas the speaker showed a downward radiation tendency. [This work was supported by JSPS KAKENHI Grant No. 24K06423 and Sophia University Special Grant for Academic Research (Research in Priority Areas).]

2pSC30. Forced alignment of Istro-Venetian speech via neural versus non-neural models. Austin Jones (Dept. of Linguist, Univ. of Georgia, Athens, GA), Massimo Daul (New York Univ., New York, NY), Margaret E. Renwick (Dept. of Cognit. Sci., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, mrenwick@jhu.edu), Zvjezdana Vrzic (Linguist Dept., New York Univ., New York, NY), John Hale (Dept. of Cognit. Sci., Johns Hopkins Univ., Baltimore, MD), and Keith Langston (Dept. of Linguist, Univ. of Georgia, Athens, GA)

Forced aligners are an essential tool to facilitate access to the word and phone level of speech data, but they rely on acoustic models, pronunciation dictionaries, and complete word-level transcriptions that may not exist for low-resource languages. Alternatively, word-level alignments can be extracted from neural network-based tools for automatic speech recognition like Wav2Vec-BERT 2.0, avoiding the need to create bespoke acoustic models and dictionaries. This paper compares the ability of the Montreal Forced Aligner (MFA) and Wav2Vec-BERT 2.0 to produce word-level alignments for the low-resource variety Istro-Venetian, spoken in Croatia. Results indicate that MFA, which uses Gaussian mixture models with hidden Markov models, significantly outperforms Wav2Vec-BERT 2.0 on forced alignment. However, we show that Wav2Vec-BERT 2.0 can produce transcriptions with a word error rate of only 14.4%. Neural models can thus aid research on low-resource languages by creating transcriptions that can be used as input to forced aligners, and with sufficient data they may also perform acceptably on forced alignment.

2pSC31. Neural network analysis of phonological variation across Istro-Venetian fricatives. Austin Jones (Dept. of Linguist, Univ. of Georgia, Athens, GA 30602, adj10085@uga.edu), Margaret E. Renwick (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD), Zvjezdana Vrzic (Linguist Dept., New York Univ., New York, NY), Massimo Daul (New York Univ., New York, NY), and Keith Langston (Dept. of Linguist, Univ. of Georgia, Athens, GA)

Istro-Venetian and Fiuman are endangered linguistic descendants of Venetian Italian spoken in the Istria and Kvarner regions of Croatia. Varieties

of Venetian have fricatives /s/ and /z/, whose realizations are known to vary from dental to postalveolar, but research on their acoustic characteristics or sociolinguistic patterning is vanishingly scarce. With a corpus of 21 sociolinguistic interviews, we conduct and inter- and intra-speaker analysis of these sibilants. We apply a neural network toolkit trained to identify acoustic characteristics corresponding to phonological distinctive features in speech. We analyze the toolkit's output of posterior probabilities for the phonological features [coronal], [strident], [distributed], [palatal], and [dorsal], drawn from Istro-Venetian fricatives. We hypothesize that these values reflect place of articulation, with higher [coronal] values for dental productions, but increased values for [palatal] and [dorsal] during postalveolar realizations. Variation in sibilant production as captured by deep learning is treated using linear mixed effects to identify linguistic and socio-demographic factors (place, gender, and age) that influence degree of retraction. This analysis represents a step toward investigating phonetic variation without phone-level alignments, and provides a novel sociolinguistic analysis of sibilant production in Istro-Venetian and Fiuman.

2pSC32. Abstract withdrawn.

TUESDAY AFTERNOON, 12 MAY 2026 FRANKLIN HALL 8, 1:00 P.M. TO 4:30 P.M.

Session 2pSP

Signal Processing in Acoustics: Signal Processing Potpourri I

Contributed Papers

1:00

2pSP1. Multi-event impulse response onset detection using lower envelope analysis of RMS derivatives. Trevor Jerome (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., BLDG 3 #329, West Bethesda, MD 20817, trevor.w.jerome@navy.mil)

Traditional impulse onset detection methods struggle with overlapping or closely-spaced impulse responses, often producing false positives or missing secondary events. This work presents an enhanced multi-event onset detection framework that extends the previously successful approach by enabling identification of multiple overlapping impulses based on intrinsic signal characteristics rather than user-defined thresholds. The method employs multi-scale constant-start RMS derivatives to identify candidates by detecting regions where the lower envelope maintains negative slope. Batches containing multiple non-positive indices with prominent negative values near the beginning are treated as separate impulses. Work presented previously demonstrated superior performance over traditional Akaike information criterion (AIC) and short-term average/long-term average (STA/LTA) methods for single impulse detection. This enhanced framework maintains these performance advantages while successfully handling multiple overlapping excitations, achieving superior accuracy and low false-positive rates across multiple SNR ranges. The approach adapts automatically to varying impulse characteristics across diverse signal environments.

1:15

2pSP2. Studying water pipeline acoustic dataset augmentation with generative adversarial networks. Ruth Willet (Penn State, 14 Eng. Collaborative Res. and Education Bldg., 556 White Course Dr., University Park, PA 16802, raw5930@psu.edu) and Karl Reichard (Penn State, State College, PA)

Prognostic health management (PHM) datasets are notoriously biased toward data measured when an engineering system is functioning properly rather than when it has failed. One example of this can be found in water pipeline acoustic monitoring systems. While supervised machine learning classifiers can still identify pipe leak data from a variety of time and frequency domain features in unbalanced water pipeline datasets, it is important to study how data classification can be improved with more balanced datasets. This work is investigating the process of creating several generative adversarial networks (GANs) and using them to generate synthetic acoustic leak data to augment measured acoustic data. GANs are most commonly trained with images, so exploring their strengths and weaknesses with acoustic data is important to further understanding of these models. Comparing results from the validation and training datasets should highlight similarities and differences in the data generation and provide insights into what features are important for useful synthetic data. This work hopes to show the differences between synthetic data from several types of GANs, compare results from GAN generated synthetic data with other standard augmentation methods, and build on current literature to improve the implementation of GANs for acoustic signal processing.

1:30

2pSP3. Explainability of large language models using the rational inattention theory: A case study in hate-speech detection. Yuan Zhao (Elec. and Comput. Eng., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, yz847@njit.edu) and Ali Abdi (Elec. and Comput. Eng., New Jersey Inst. of Technol., Newark, NJ)

Hate-speech detection is important for maintaining healthy communities. Large language models (LLMs) have demonstrated good performance on text classification, where the text is generated by transcribing an audio content. However, the confidence and decision strategy behind a classification are often unclear unless the LLM generates an explanation, which increases inference costs, especially at platform scales such as social media. Since processing signals and updating beliefs to make decisions are costly, the rational inattention (RI) theory is developed as a powerful approach to balance performance and cost, by rationally ignoring some signals. To understand how LLMs are successful in text classification, without costly generation of explanations, we introduce a new RI model where LLM classifiers are envisioned as decision makers. Using a hate-speech dataset

constructed across multiple environments with different noise levels, we estimate the parameters of the new RI model that quantify and reveal LLMs' attention strategies when they make decisions. We also demonstrate how LLMs' correct decision probabilities and their amount of attention change across environments. Overall, the results show that this new rational inattention model enables us to explain LLMs' classification outcomes in terms of confidence and decision strategy, without prompting the LLMs to generate computationally expensive explanations.

1:45

2pSP4. Multi-channel blind passive detection of weak signals: A learning-based approach using multiple vector and scalar underwater field components. Behrooz Mosallaei (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, bm636@njit.edu), Ali Abdi (Elec. Comput. Eng. Dept., New Jersey Inst. of Technol., Newark, NJ), and Zoi-Heleni Michalopoulou (Mathematical Sci. Dept., New Jersey Inst. of Technol., Newark, NJ)

Due to multipath and channel uncertainties, detection of weak signals in underwater environments is a challenging task. Furthermore, the distribution of signal energy among multiple propagation paths makes the weak signal easier to be masked by noise. The lack of prior knowledge of the signal waveform—convolved with an unknown channel impulse response—requires a blind detection approach. In this work, we introduce a multi-channel dictionary-learning blind and passive detection method. The dictionary-learning formulation allows for estimating the unknown signal waveform, as well as deconvolving it from the unknown channel response. As for the multi-channel reception formulation in order to enhance SNR, we utilize the vector and scalar components of the underwater acoustic field, all measured simultaneously by a compact vector sensor. While a scalar array of hydrophones can also be used, employing a vector sensor allows for the blind detector to be deployed in small platforms that have size constraints and cannot accommodate hydrophone arrays. Using simulations and underwater experiments, we demonstrate how multiple vector and scalar components of the field can be jointly used to achieve high detection probabilities in scenarios where matched filtering is not feasible because of not knowing the signal waveform and the channel response.

2:00

2pSP5. Information-guided angular inspection in echolocation-based target discrimination. Aditya Krishna (Elec. and Comput. Eng., Univ. of Washington, 1013 NE 40th St., Henderson Hall (APL) Rm. No. 459, Seattle, WA 98105, adkris@uw.edu) and Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Echolocating toothed whales and bats use self-generated acoustic energy to perform active sensing, a closed-loop process of acquiring sensory information by planning and adjusting their interaction with the environment. While many models have been developed to describe echolocating animals' modulation of outgoing signals based on received echoes, few models have explored the role of movement for interrogating objects from sequentially selected inspection angles. Here, we develop a model for an echolocating agent performing a go/no-go target discrimination task where the agent must select its acoustic views of an object to conclude its identity as either the target or a non-target. After each echolocation ping, the agent can choose to maintain or alter its subsequent inspection angle by remaining still or stepping in either clockwise or counter-clockwise directions based on an information-greedy strategy known as "infotaxis." We show that the infotaxis strategy generates angular movement sequences reminiscent of echolocating animals' behavior when performing target discrimination, and that maintaining memory of previous inspection angles and returning echoes improves the performance. We further investigate how the agent's internal target and non-target models influence its inspection angle choices, which is crucial for connecting this model to experimental observations.

2:15

2pSP6. Phased array measurements of scaled propellers through a wind tunnel shear layer. Mary L. Houston (NASA Langley Res. Ctr., NASA Langley Res. Ctr., M.S. 461, Hampton, VA 23601, Mary.L.Houston@NASA.gov) and Nikolas S. Zawodny (NASA Langley Res. Ctr., Hampton, VA)

As part of the second phase of the vertical lift propeller noise test (VLPNT), several propeller/proprotor geometries were measured in the NASA Langley 14- by 22-foot subsonic tunnel (14×22) in its semi-open jet, acoustically treated configuration. Blade geometries were chosen to be representative of those used in the emerging future vertical lift (FVL) and urban air mobility (UAM) industries. Performance and acoustic measurements were taken of the propellers at high incidence angles, simulated descent conditions, and for dynamic pitching maneuvers intended to emulate portions of a tilting rotor's transition envelope. A 54-microphone phased array was deployed to isolate targeted acoustic sources while rejecting background noise. A time-domain beamforming algorithm, the ROtating Source Identifier (ROSI), was used to track and acoustically image the rotating blades. Advantages and limitations of the ROSI method are discussed. In the final submission, shear layer correction methods will be applied to the data. Additionally, advanced beamforming algorithms will be explored to improve resolution and enhance source localization.

2:30

2pSP7. Performance comparison of time difference of arrival and matched field processing for source localization in lieu of environmental mismatch. Isabelle Brandicourt (Oceanogr., Univ. of Washington, 1501 NE Boat St., #206, Seattle, WA 98195, imcbrandicourt@gmail.com) and Shima Abadi (Univ. of Washington, Seattle, WA)

Knowledge of underwater sound source locations is fundamental to many scientific and engineering problems in underwater acoustics. Two of the most common methods for underwater source localization are time difference of arrival (TDOA) and matched field processing (MFP). The performance of each method depends on array geometry, environmental characterization, and source position, resulting in distinct sensitivities to environmental mismatch. Here, we compare theoretical performance limits of each method through derivations of the cramer rao lower Bounds (CRLBs) and the associated variances in source position estimation. Building on prior work by Blanc–Benon (1995) examining the impact of sound-speed errors, we extend the comparison to evaluate robustness under additional sources of mismatch, including signal-to-noise ratio, receiver positioning errors arising from vertical mooring tilt, and source location. Monte Carlo simulations are performed for a limited-element, vertically moored array in a shallow-water waveguide representative of coastal environments. These simulations quantify the comparative effects of increasing and compounding uncertainties on TDOA- and MFP-based localization performance.

2:45–3:00 Break

3:00

2pSP8. Analysis of cable strum characteristics in the PhilSea10 experiment. Emily H. Zhu (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, ezhu3@gmu.edu), Jeff Tucker, and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Sensor arrays in the deep ocean are used for tasks such as characterization of ambient noise and estimation of the time-evolving ocean state. Since these arrays often span multiple kilometers, they are particularly vulnerable to cable strum. The PhilSea10 experiment was conducted in the Philippine Sea in 2010 and 2011. It included a 5-km long distributed vertical line array with 150 hydrophones. The array data contain intermittent cable strum that is often loud enough to obscure the cone of distant shipping noise. Wage and Buck used the strum as an interferer of opportunity and mitigated it with a universal adaptive filter [IEEE SAM, 2018]. Tucker and Wage conducted a similar study using a performance weighted combination of spatial filters designed from a statistical characterization of the strum signal [IEEE SAM, 2024]. While these studies suggest methods for rejecting interference, there has not been a detailed study of the strum signal itself. This talk

presents analyses of array receptions from a 30-sensor segment, as well as current profiler data and mooring navigation data to investigate the harmonic and spatial structure of the strum and its relationship to environmental factors. Work supported by ONR.

3:15

2pSP9. Detection, localization, and discrimination of impulsive acoustic sources using ground-coupled airwaves in urban environments. Samba Gaye (SEAS, Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, samba.gaye@udc.edu)

This work examines the use of ground-coupled airwaves for the detection, localization, discrimination, and classification of impulsive acoustic sources in complex urban environments. Ground-coupled airwaves are generated when high-energy acoustic events, such as gunshots or explosions, interact with the ground surface and couple into the subsurface as seismic waves. These signals are recorded using a distributed seismic sensor array and are observed to propagate at velocities close to the speed of sound in air. Source localization is performed using array-based techniques, including beamforming and time-difference-of-arrival (TDOA) analysis, which exploit the coherent propagation of ground-coupled airwaves across the array. Time–frequency analysis is applied to extract salient waveform features, which are subsequently used in machine learning frameworks to discriminate and classify different impulsive source types. Results demonstrate that ground-coupled airwaves preserve both spatial and spectral information sufficient to support accurate localization and reliable discrimination, even in the presence of strong reflections, scattering, and multipath effects characteristic of urban terrain. The combined use of seismic sensing, array signal processing, and data-driven classification highlights the potential of ground-coupled airwaves as a robust modality for urban acoustic monitoring, with applications in threat detection, situational awareness, and environmental acoustics.

3:30

2pSP10. Suppressing benign ultrasonic non-destructive evaluation artefacts to maximize the energy and shape of defect signatures. Sergio Cantero-Chinchilla (School of Elec., Electron. and Mech. Eng., Univ. of Bristol, University of Bristol, Queen's Bldg., University Walk, Bristol BS8 1TR, United Kingdom, sergio.canterochinchilla@bristol.ac.uk), Paul Wilcox, and Anthony Croxford (School of Elec., Electron. and Mech. Eng., Univ. of Bristol, Bristol, United Kingdom)

The presence of structural artefacts in most non-destructive evaluation modalities, such as ultrasonic array imaging, guided waves, or x-ray, poses a challenge to the detection and characterization of defects. In ultrasonic array inspections, these benign artefacts manifest as reflections from different structural features. These artefacts pose a risk in the detection of defects as they obscure defects' signatures if these are close to structural features (e.g., bottom crack tip), potentially leading to false negatives. Previous efforts have relied on statistical and data-driven techniques to mitigate this problem. However, an accurate reconstruction of the defect signatures when these interact with benign artefacts remains open. This work explores a subtraction approach based on a series of basis functions learned from each benign artefact in a predefined array inspection. These functions are then used to suppress the benign artefacts while maximizing the shape and energy of the defect signatures. Multiple suppression approaches are compared, including dictionary learning and weighted mean subtraction. The proposed framework is illustrated in simulated and experimental data, highlighting the remaining challenges. This approach is less computationally intensive and more transparent than the state-of-the-art and it aims to generalize and maximize the performance of artefact suppression in arbitrary geometries.

3:45

2pSP11. Error analysis of noise directionality estimation using a towed line array. John J. Lyons (Acoust., Penn State Univ., 47 Warwick Way, Centerville, MA 02632, jjl7288@psu.edu), Chad M. Smith (Acoust., Penn State Univ., State College, PA), and Geoff Goehle (Penn State Univ., State College, PA)

This talk characterizes the accuracy of undersea noise field directionality estimates produced by the Wagstaff method [J. Acoust. Soc. Am. **63**, 863–869 (1978)] using data from a towed line array. This method, which reconstructs the mean directional noise field using repeated measurements, aims to minimize the error (disagreement) between measured and reconstructed beam noise levels. Sources of error present in the method include but are not limited to aliasing, ambiguity, and array heading uncertainty, with estimation of the error in noise field reconstruction being a gap in the literature. We present techniques correlating error in the noise field reconstruction to error in reconstructed beam noise levels, presenting uncertainty in this relationship statistically via prediction intervals applied to this relationship using a large set of synthetic noise directionalities. The relationship between the two errors depends on the array geometry and on the set of array headings used in data collection. We show that a shorter array using more measurement headings performs better with regards to this error relationship than a longer array and discuss the implications of this result for experimental design in a practical scenario.

4:00

2pSP12. Experimental analysis on the direction of arrival accuracy of ambisonic microphones. Steven J. Todd (The Penn State Univ., 201 Old Main, University Park, PA 16802, svt5691@psu.edu) and Michael Roan (ME, Penn State, State College, PA)

This presentation will show an experimental campaign conducted to determine the direction- of-arrival (DoA) accuracy of soundfield microphones. Soundfield microphones are most com- monly used for virtual audio used in entertainment, but have found growing use in scientific studies. As such, quantifying expected performance of these microphones is important in understanding the overall error in scientific studies. Two comparisons between three ambisonic microphones of first, second, and third order will be presented. First, the DoA accuracy of each ambisonic microphone will be calculated using solely the first order spherical harmonics which provides a base layer of comparison as all microphones used are able to process first order spherical harmonics. Then, the DoA accuracy of each ambisonic microphone will be calculated using as many spherical harmonics as is available to them which shows the expected improvement as one moves to higher order microphones. Additionally, multiple DoA algorithms will be used. This includes directional audio coding (DirAC) and high angular resolution plane wave expansion (HARPEX). These array and algorithm combinations will provide an experimental baseline for expected DoA accuracy of these ambisonic microphones.

4:15

2pSP13. Adaptive data-driven audio network for speech deepfake detection. Poline Xian (Comput. Sci., Hong Kong Baptist Univ., 224 Waterloo Rd., Kowloon Tong, Hong Kong 999077, Hong Kong, poline3939@gmail.com), Cheer Cheung (Hong Kong Baptist Univ., Hong Kong), Jiaxi Ying (Mathematics, Hong Kong Univ. of Sci. and Technol., Hong Kong), and Jun Qi (Comput. Sci., Hong Kong Baptist Univ., Hong Kong)

Automatic speaker verification (ASV) systems are often vulnerable to spoofed attacks, particularly unseen attacks. Among the audio deepfake detection algorithms, AASIST and RawNet2 are primarily used. These methods employ the Sinc filters as encoders to learn an efficient filter bank. The key limitations include the model's lack of generalization to different datasets, and the computational cost is high compared to simpler feature models. In this work, we propose the adaptive data-driven audio network. This network identifies the sparse coefficients for the input audio patches and updates the filters to better align with the data. It allows the model to extract more relevant and complex features of audio signals. The network reduces computational time to one-third of the network algorithms with Sinc filters. Experimental results on English, Chinese, and Cantonese audio deepfake detection demonstrate that the network outperforms other typical algorithms, such as AASIST and RawNet2. The network offers greater stability and robustness.

Session 3aAA

Architectural Acoustics and Noise: Interesting Measurement Techniques and Results Part I

Michael Raley, Cochair
PAC International, 2000 4th Ave., Canby, OR 97013

Joseph Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, STE 209, Woodbridge, NJ 07095

Chair's Introduction—8:35

Invited Paper

8:40

3aAA1. Virtual Goniometer (VIRGO) predicts the scattering and diffusion coefficients of any topology from a CAD file. Peter D'Antonio (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), Luiz A. Alvim, and Jeffrey S. Madison (RPG Acoust. Systems LLC, Passaic, NJ 07055)

Experimental determination of scattering and diffusion coefficients in accordance with ISO 17497-1 and ISO 17497-2 is resource-intensive, requiring multiple physical samples, access to anechoic or large reflection-free environments, and strict far-field measurement conditions. As a result, these measurements are rarely available from commercial laboratories and are impractical during early design development. To overcome these limitations, this work introduces a Virtual Goniometer based on wave-based numerical simulation. The method eliminates the need for scale models or full-scale prototypes by computing scattering and diffusion coefficients directly from a 3-D CAD file. A finite element method (FEM) combined with perfectly matched layers (PML) is used to emulate free-field conditions and extract directional and random-incidence coefficients consistent with ISO standards. The approach has been validated through extensive comparison with experimentally measured topologies, demonstrating strong agreement across a wide range of geometries. It accurately models complex scattering topologies, including hybrid reflecting/absorbing configurations. This Virtual Goniometer process enables acousticians to design, evaluate, and optimize project-specific scattering surfaces without the delays and costs associated with physical prototyping. It significantly accelerates the architectural design development process and provides a robust means of verifying the acoustic performance of architecturally proposed geometries.

Contributed Papers

9:00

3aAA2. Variation in absorber data on room performance study. Kevin Herreman (Owens Corning, 2790 Columbus Rd., B75, Granville, OH 43023, USA, kevin.herreman@owenscorning.com)

It has been shown that testing the same material at different laboratories can result in reproducibility variation in the data. A fact that is published in the sound absorption test standards. By comparing the calculated background levels and reverberation times for various spaces using the measured results from multiple laboratories a general understanding of the impact of the variation can be understood. The intent of this study is to compare results from various laboratories on the performance of the spaces the material could be used.

9:15

3aAA3. Experimental and numerical analysis methods for improved material characterization using a standing wave impedance tube: Demonstration on experimental data. Erin Nguyen (Graduate Program in Acoust., The Penn State Univ., 414 Eng. Collaborative Res. and Education Bldg., 556 White Course Dr., University Park, PA 16802, een5106@psu.edu), Sean Collier (Penn State Appl. Res. Lab., University Park, PA 16802), and Nicholas Vlajic (Penn State Appl. Res. Lab., State College, PA 16801)

Acoustic properties of materials can be characterized using standing wave impedance tubes, but measurements are often limited in frequency bandwidth due to the dimensions of the setup. To increase the bandwidth and improve impedance tube measurements, we present various ways that standard testing methods, such as the four-mic method and two-load method, can be expanded upon. This presentation will discuss the six-mic as well as the multi-load methods that extend the measurement bandwidth, smooth out experimental noise, and assist in taking measurements when anechoic terminations are difficult to achieve. At ASA/ASJ Honolulu, we compared two new optimization techniques to the traditional four-mic method using synthetic data. As a continuation of our previous presentation, we apply these various testing and processing techniques to experimental data. By using the six-mic, multi-load, and optimization methods, we show an increase in frequency bandwidth and improved experimental results, particularly when solving for the speed of sound through the testing sample.

9:30

3aAA4. Low-frequency absorption methods: Quantifying bias, uncertainty, and repeatability in 20–550 Hz impedance-tube measurements. Daniel H. Robinson (Sublime, P.O. Box 25816, St. Paul, MN 55125-0816, daniel.robinson@sublime.llc)

This work presents advances in low-frequency impedance tube measurements using a custom-built 8 ft. long, 12 in. wide tube designed for a practical range of 20–550 Hz. The tube employs three 1/4″ microphone positions and a 10″ subwoofer source, with calibration performed via channel interchange and single-mic piecewise methods to mitigate phase mismatch errors. A calibration approach places the microphone in close proximity to the rigid termination, improving accuracy at very low frequencies. Key challenges addressed include tube wall vibration, strong resonances, and uncertainty in generating sufficient energy below 40 Hz. Parameter estimation and dissipation modeling, inspired by recent research, further enhance measurement reliability. Results are reported in accordance with ASTM E1050 and ISO 10534-2 standards, with a focus on uncertainty and repeatability. This approach enables more accurate characterization of acoustic materials at low frequencies and provides practical insights for future measurement and calibration techniques.

9:50

3aAA5. Digital wireless microphone system for long distance acoustics measurements. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Steph Harrington (DLR Group, Lincoln, NE 68508)

DLR Group created a field measurement kit which utilizes digital wireless microphone transmitters and receivers to transmit measurement signal over long distances. The presentation will show the assembly of the kit and discuss challenges and successful deployments.

10:10

3aAA6. The case of the dirt bike bandit. Eric Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

The rider of a 2-stroke dirtbike was charged with felony elusion after failing to pull over while being pursued by police. Was is plausible, as he claimed, that he was unable to hear the siren of the cruiser over the noise from the bike, filtered through his helmet? The author, serving as an expert for the defense, reconstructed the incident using in-ear microphones, the bike, the subpoenaed police cruiser, data gleaned from the dashcam video, and various other elements. Tune in for the exciting conclusion!

10:30–10:45 Break

10:45

3aAA7. Using sound intensity for environmental source localization. Matt Slaughter (NTi Audio, 1804 NE 14th Ave., Apt 4, Portland, OR 97212, mattslaughter6@gmail.com)

Sound intensity is traditionally used to measure sound power and energy flow, but can also provide directional information that is useful for source localization. A compact array of eight MEMS microphones was deployed in environments where conventional microphone arrays would be impractical due to the quantity of microphones needed, price, and space constraints. Measurements were conducted over source–receiver distances spanning centimeters to kilometers and frequencies from 63 Hz to 8 kHz to evaluate the accuracy of the spatial information. Azimuth and elevation information was derived for the dominant source. Results show that intensity-based localization can provide reliable directional information. Observed limitations related to low-frequency performance, elevation, and localization uncertainty are discussed. The measurements demonstrate that sound intensity instrumentation, when applied beyond its typical use case, can yield meaningful results and offers a practical alternative for spatial acoustic analysis in challenging measurement environments.

11:05

3aAA8. Searching for corona noise in an electrical utility right-of-way. Joseph Quagraine (Ostergaard Acoust. Assoc., 1460 Rte. 9N, Ste. 209, Woodbridge, NJ 07095, jquagraine@acousticalconsultant.com) and Joseph Keefe (Ostergaard Acoust. Assoc., Woodbridge, NJ)

Ostergaard Acoustical Associates was engaged by an engineering firm to carry out a noise study of existing and future sound emissions from an electrical utility right-of-way. Specifically, the concern was possible corona discharge noise: a crackling, hissing, or buzzing sound caused by the ionization of air surrounding high-voltage electrical conductors. This presentation will discuss the challenges of physically accessing needed measurement locations, measuring sound that is sporadic and only prevalent under certain weather conditions, and correlating long-term unattended measurement data with weather data acquired via a deployed weather station. A discussion of IEEE 656: Standard for the Measurement of Audible Noise From Overhead Transmission Lines, including its useful content and practical limitations, will be included.

Contributed Paper

11:25

3aAA9. Disproving the validity of the environmental adjustment factor in standard AHRI 885. Derrick P. Knight (Trane Technologies, 2313 20th St. South, La Crosse, WI 54601, Derrick.Knight@TraneTechnologies.com) and Victoria Knott (Purdue Univ., West Lafayette, IN)

The Environmental Adjustment Factor (EAF) is a theoretical correction prescribed by standard AHRI 885 to account for what was believed to be a bias in sound power measurements of HVAC equipment in a free field. The effect was originally estimated by AHRI standard committee members who referenced work published by Vorländer and Raabe. ASHRAE research study RP 755 performed additional measurements to justify reducing the original estimates of this effect. The work presented in this current study was used to show that that effect was never justifiable from the source material. Additionally, new measurements were made and comparisons of data across decades showed that the EAF was a bias in uncertainty related to measuring sound power in a reverberant field using the direct method. The EAF is in the process of being removed from standard AHRI 885 because of this work.

WEDNESDAY MORNING, 13 MAY 2026 FRANKLIN HALL 5, 7:55 A.M. TO 10:25 A.M.

Session 3aAB

Animal Bioacoustics, Computational Acoustics and Acoustical Oceanography: Machine Learning Approaches for Analyzing Large Acoustic Datasets I

Irina Tolkova, Cochair
K. Lisa Yang Center for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850

John Allen, Cochair

3a WED. AM

Chair's Introduction—7:55

Invited Papers

8:00

3aAB1. Individual identification of songbirds based on vocal signature: Successes, failures, and potential. Justin Kitzes (Univ. of Pittsburgh, Fifth and Ruskin Ave., Pittsburgh, PA 15260, justin.kitzes@pitt.edu), Lauren Chronister, Samuel Lapp, and Leonardo Viotti (Univ. of Pittsburgh, Pittsburgh, PA)

Automated sensors and machine learning models are transforming ecology and conservation by enabling surveys of species in the field at unprecedented spatial and temporal scales. Today, the vast majority of such surveys are designed to detect species presence by recording, and then classifying, the vocalizations made by adults of a particular species. Moving forward, new methods such as transfer learning, few-shot learning, and active learning will provide opportunities to go beyond sounds that indicate simple species presence to identify sounds that indicate an individual's state, behavior, or unique identity. Here, I describe progress that we have made toward using large pre-trained acoustic foundation models to identify individual birds within species based on their unique vocal signatures. I describe examples of three attempts to perform individual identification with three distinct species (Ovenbird, Song Sparrow, and 'Oma'o), with varying degrees of success related to the song system of each species, classifier performance, and recording method. I highlight how the interaction of species biology and technical methods combine to determine potential research questions (e.g., abundance, movement, yearly survival) that can be asked and answered using such deep learning approaches to individual identification.

8:20

3aAB2. What matters for bioacoustic encoding. Marius Miron, David Robinson, Milad Alizadeh, Ellen Gilsenan-McMahon, Gagan Narula, Benjamin Hoffman (Earth Species Project, West Hollywood, CA), Olivier Pietquin (Earth Species Project, 8605 Santa Monica Blvd # 78666, West Hollywood, CA 90069, olivier@earthspecies.org), and Matthieu Geist (Earth Species Project, West Hollywood, CA)

Applications of machine learning to bioacoustics often suffer from limited annotated data, highlighting the need for a general-purpose bioacoustic encoder capable of extracting useful representations for diverse downstream tasks. Such encoders have been proposed before, but are often limited in scope due to a focus on a narrow range of species (typically birds), and a reliance on a single model architecture or training paradigm. Moreover, they are usually evaluated on a small set of tasks and datasets. We present a large-scale empirical study that considers training data diversity and scale, model architectures and training recipes, and the breadth of evaluation tasks and datasets. We obtain encoders that are state-of-the-art on existing and newly proposed benchmarks. We also identify what matters for training these encoders, such that this work can be extended when more data are available or better architectures are proposed. Specifically, across 26 datasets with tasks including species classification, detection, individual ID, and vocal repertoire discovery, we find self-supervised pre-training followed by supervised post-training on a mixed bioacoustics + general-audio corpus yields the strongest in- and out-of-distribution performance. We show the importance of data diversity in both stages.

8:40

3aAB3. From blue jays to blue whales: Leveraging bioacoustics foundational models, pretrained on birds, for long-term marine mammal passive acoustic monitoring. Lauren Harrell (Google, 2930 Pearl St., Boulder, CO 80301, laurenharrell@google.com), Andrea Burns (Google, New York, NY), Bart van Merriënboer, Vincent Dumoulin (Google, Montreal, QC, Canada), Jenny Hamer (Google, New York, NY), Ștefan Istrate (Google, London, United Kingdom), and Tom Denton (Google, San Francisco, CA)

Using transfer learning from pretrained bioacoustics embedding models can drastically reduce the amount of time, computation, labeled data, and experimentation needed for researchers to develop a high-performing custom classifier for a targeted species or sound set with limited (or no) annotated data. For example, our latest supervised bioacoustics foundation model, Perch 2.0, was pretrained on over 14 000 terrestrial species, including birds, mammals, amphibians, and insects. Despite having no underwater marine mammal examples in the training data, Perch 2.0 had state-of-the-art performance for transfer learning to underwater acoustics classification tasks including baleen whale species classification and distinguishing between killer whale ecotypes from passive acoustic monitoring data. Given a high-performing embedding model such as Perch 2.0, we share how to use agile modeling to embed passive acoustic data (terrestrial and/or underwater) and efficiently create a new classifier from unlabeled data by searching for target sounds. We also discuss additional considerations such as computation time and metadata for applications to large-scale passive acoustic data and generating downstream insights.

9:00–9:15 Break

Contributed Papers

9:15

3aAB4. Exploring context relevance for echogram classification tasks using class activation map methods. Caesar Tuguinay (Appl. Phys. Lab. and Dept. of Elec. and Comput. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98015, ctuguina@uw.edu) and Wu Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Echograms are images commonly used to visualize and classify scattering sources in water column sonar data. We developed a convolutional neural network-based binary semantic segmentation model that classifies regions of echogram as either Pacific hake, a trophically important semi-pelagic groundfish, or background. Interpretation of the model's reasoning in making decisions was difficult for two reasons. First, the model setup does not account for variability within human-annotated hake aggregations, which include identifiable subtypes. Second, heterogeneous echogram context, including unlabeled non-hake biological aggregations and bathymetry features, exists in the background. To better understand if and how echogram context influences the model behavior across different hake aggregation subtypes and environments, we explored class activation map (CAM) methods, which highlight regions that most influence a model's decisions. Traditionally, CAM methods identify which parts within an object a model uses to predict its class. While this within-object viewpoint is useful, our analysis using gradient-based and eigendecomposition CAMs suggested that the model also utilizes echogram features outside specific objects, or the "context" in making decisions. We will conclude by discussing how CAM analysis may be used as a tool to design or select model architectures that promote desired model behaviors.

9:30

3aAB5. Spatiotemporal variation in Narwhal echolocation within Canadian arctic marine conservation areas. Juliana Rodrigues Moron (School of Earth and Ocean Sci., Univ. of Victoria, 3117 Tillicum Rd., Victoria, BC V9A 2B4, Canada, jrodriguesmoron@uvic.ca), William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), Nigel Hussey (Univ. of Windsor, Windsor, ON, Canada), Marianne Marcoux (Univ. of MB, Winnipeg, MB, Canada), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Canadian Arctic marine conservation areas (CAMCAs) are intended to safeguard rapidly changing Arctic ecosystems, yet many lack the ecological baselines needed to evaluate their effectiveness. In southern Baffin Bay, two CAMCAs—the Disko Fan and Davis Strait—overlap with key habitat for narwhals (*Monodon monoceros*), deep-diving whales that depend on offshore pack ice and undertake long annual migrations. To assess whether narwhals use these areas seasonally or year-round, we manually analyzed duty-cycled (5 min every hour) broadband passive acoustic recordings collected between September 2020 and September 2021 at four sites within CAMCAs and one reference site near Qikiqtarjuaq outside CAMCAs boundaries. Semi-automated click-detection and parameter-extraction methods in PAMGuard and R were used to identify narwhal echolocation clicks in a transparent and replicable way and to quantify temporal and spatial patterns in acoustic activity. These curated detections are being used to train supervised machine learning models, such as Random Forest classifiers applied to spectrogram-based features, to automate narwhal click detection in large datasets collected from complex soundscapes. Model outputs will be used to compare seasonal and regional patterns in narwhal habitat use across conservation and reference sites, providing a scalable basis for assessing the ecological effectiveness of Arctic marine conservation areas.

9:45

3aAB6. Comparing machine learning techniques with spectral methods for identification of species-specific sounds for follow-up analysis. Mark W. Sprague (Phys., East Carolina Univ., 1000 E 5th St., C209 Howell Sci Cmplx, MS 562, Greenville, NC 27858, spraguem@ecu.edu) and Joseph J. Luczkovich (Biology, East Carolina Univ., Greenville, NC)

Acoustical methods can be used to identify habitat use by soniferous animals, and these methods are particularly useful in environments where we cannot see them. But recordings of acoustically active locations capture many sounds in addition to those of the target species. Analyzing these recordings in detail wastes time and resources on nontarget sounds. Therefore, we use machine learning (ML) models and spectral techniques to identify sounds for further analysis. We deployed a seven-channel hydrophone array in a saltwater creek and recorded sound continuously for 24 h. Localizing sound sources using the hydrophone positions showed how the locations of sound-producing striped blenny *Chasmodes bosquianus*, oyster toadfish *Opsanus tau*, Atlantic croaker *Micropogonias undulatus*, and spotted seatrout *Cynoscion nebulosis* varied throughout the day. We used three techniques for the identification of sounds for localization analysis: a YOLOv5 model trained on images of 3 s spectrograms, a ResNet-18 convolutional neural network trained on 3 s spectrogram arrays, and power spectral band sum (PSB) analysis. Comparing the results of these analysis techniques we find the ML models are useful for identifying short-time sounds, but PSB analysis is faster and easier to use for identification of long-time aggregation sounds like those of the striped blenny.

10:05

3aAB7. Rethinking segmentation metrics for echogram analysis. Wu Jung Lee (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, leewj@uw.edu), Caesar Tuguinay, and Valentina Staneva (Univ. of Washington, Seattle, WA)

High-frequency echosounders are widely used to map and quantify fish distribution and abundance in fisheries and ecological studies. A key step in echo analysis is classifying echogram features into different scatterer classes, a process that has traditionally been performed manually or semi-manually and does not scale with growing data volumes. Deep learning-based image segmentation has emerged as a promising alternative. However, echogram labels differ fundamentally from those of standard images. Depending on species, fish aggregations may lack well-defined boundaries, and expert annotations often include low-backscatter "empty" water, particularly along region edges, that contribute minimally to downstream acoustically derived biological quantities. This ambiguity is further compounded by the logarithmic scaling commonly used in echogram visualization, which is not proportional to acoustic energy. Using data from an acoustic-trawl survey targeting Pacific hake, a species that forms both loosely connected mid-water schools and diffuse near-seafloor aggregations, we examine how commonly used segmentation metrics fail to reflect these acoustic-specific considerations. We then explore evaluation metrics that emphasize echo strength within segmented regions over precise label boundary agreement. Together, these investigations motivate the development of acoustically meaningful loss functions to improve the robustness and biological relevance of deep learning models for echogram analysis.

Session 3aBAa

Biomedical Acoustics and Physical Acoustics: Vortex Ultrasound

Chengzhi Shi, Cochair
Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109

Likun Zhang, Cochair
National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38655

Chair's Introduction—8:00

Invited Papers

8:05

3aBAa1. Observation of rotational Doppler effect on a MEMS diaphragm in a deep subwavelength regime. Wei Wang (Dept. of Elec. and Comput. Eng., the Univ. of Texas at Austin, 2501 Speedway, Austin, TX 78712, weiwang@austin.utexas.edu), Zihuan Liu, Yen-Chen Wang, and Neal A. Hall (Dept. of Elec. and Comput. Eng., the Univ. of Texas at Austin, Austin, TX)

The rotational Doppler effect associated with waves carrying angular momentum in rotating systems has attracted increasing attention due to its potential applications in sensing and communication. While widely studied in optical and acoustic wave propagation, its manifestation in mechanical systems remains largely unexplored. Here, we report the experimental observation of the rotational Doppler effect in a microelectromechanical system (MEMS) operating in a deep subwavelength regime. The system consists of a circular MEMS diaphragm driven by four spatially distributed actuation ports. Phase-delayed excitation signals corresponding to a helical phase profile selectively excite rotating vibrational modes on the diaphragm. The MEMS device is mounted on a rotating platform, and its vibrational response is measured using laser Doppler vibrometry (LDV). A frequency shift proportional to the platform's angular velocity is observed, providing direct evidence of the rotational Doppler effect in mechanical vibration. Importantly, the same frequency shift is transferred to the acoustic waves radiated by the diaphragm and can be detected using a microphone. Owing to the compact MEMS architecture, the effect is observed at a deeply subwavelength scale on the order of $\lambda/100$, enabling a compact platform for studying rotational Doppler phenomena in both acoustics and mechanics.

8:30

3aBAa2. Array and phase-ramp approaches to acoustic vortex beam generation and applications to radiation torque, communications, and scattering. Philip L. Marston (Washington State Univ., Phys. & Astronomy Dept., Pullman, WA 99164-2814, marston@wsu.edu)

Motivated in part by discussions of helicoidal waves [J. F. Nye and M. V. Berry, *Proc. Roy. Soc.* (1973)] and angular momentum transport and angular momentum transport [P. L. Marston and J. H. Crichton, *Phys. Rev. A* (1984)], Hefner and Marston tested approaches to generating acoustic vortex beams (AVBs) in water [*JASA* (1999)]. By raster scanning a hydrophone it was possible to verify that array and phase-ramp approaches radiated diverging waves having the expected helicoidal phase distribution. Furthermore, quasi-holographic back propagation displayed anticipated aspects of AVB spatial evolution. Paraxial analysis of a simplified model (Laguerre–Gauss beams) gave approximations for torque on absorbing objects and angular momentum transport. A simple approach was subsequently demonstrated for rapidly reversing the topological charge enabling AVB-based communication [Marston and Marston, abstract, *JASA* (2010); Bollen *et al.*, *POMA* (2013)]. When symmetric arrays are used to detect the scattering of AVBs by objects, helicity dependent projections of the detected scattering reveal phase information sensitive to the object's location [Bollen and Marston, *JASA-EL* (2020)]. Generalizations of the analysis of angular momentum transport are available [Zhang and Marston, *Phys. Rev. E* (2011)] as well as for vortex Bessel beams predictions for scattering and radiation forces [Marston, *JASA* (2008, 2009)]. [Work supported by ONR.]

8:55

3aBAa3. Analytical expressions of second harmonics in the far fields of vortex beams. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 9767, Austin, TX 78766-9767, cgokani@arlut.utexas.edu), Randall P. Williams, and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78766-9767)

Second harmonics generated by acoustic vortex beams are difficult to describe analytically due to mathematical complications introduced by the dependence of the field on the azimuthal angle ϕ. However, for a spherically spreading wave with wavenumber k, directivity D, and velocity amplitude u_0 at radius $r = a$, the complex pressure of the second harmonic for $kr \gg 1$ and $|\nabla D| \ll k$ is $p_2 = i\beta\rho_0 {u_0}^2 k a^2 \ln(r/a) D^2 e^{-2i\omega\tau}/2r$, where β is the coefficient of nonlinearity, ρ_0 is the ambient fluid density, ω is angular frequency, and τ is retarded time [Westervelt and Radue, *JASA* **33**, 1672 (1961)]. For a Gaussian velocity source with orbital number $\ell > -2$, $D(\theta, \phi) = \pi^{1/2}(ka)^3 \sin(\theta) e^{-\chi} [I_{(\ell-1)/2}(\chi) - I_{(\ell+1)/2}(\chi)] e^{i\ell(\phi-\pi/2)}/8$, where I_ν is the νth-order modified Bessel function, $\chi = (ka \sin\theta)^2/8$, and θ is the polar angle. For a circular piston with the same phase dependence, $D(\theta, \phi) = F_\ell(ka \sin\theta) e^{i\ell(\phi-\pi}/2)/\sin^2\theta$, where F_ℓ is given by Eq. (7) of Gokani *et al.*, *JASA-EL* **4**, 124001 (2024). Comparison with numerical solutions of the Westervelt equation shows that p_2 accurately predicts the vortex ring amplitudes and radii of the second harmonics for $z \gtrsim 2z\mathrm{R}$, where $z_\mathrm{R} = ka^2/2$, despite the presence of near-field effects in that region. The expressions become increasingly accurate for $\ln(z/z_\mathrm{R}) \gg 1$.

9:10

3aBAa4. Multi-feature acoustic vortex characterization and automated quality classification via physics-informed ensemble model. Chadi Ellouzi (Dept. of Mech. Eng., Rowan Univ., 201 Mullica Hill Rd., Rowan Hall 212, Glassboro, NJ 08028, chadie17@rowan.edu), Farhood Aghdasi, and Chen Shen (Dept. of Mech. Eng., Rowan Univ., Glassboro, NJ 08028)

Acoustic vortices carrying orbital angular momentum are central to emerging technologies in particle manipulation, acoustic trapping, imaging, and high-capacity communication. However, objective and repeatable evaluation of vortex quality remains an open challenge, with current practices relying on visual inspection or isolated metrics that are highly sensitive to fabrication defects and measurement noise. This work presents a physics-informed, automated framework for the quantitative assessment of acoustic vortex quality. A compact six-parameter descriptor set is introduced to capture vortex geometry, symmetry, phase continuity, and angular momentum distribution in a physically interpretable manner. Using a combination of simulated and experimentally measured fields, a statistically consistent synthetic dataset comprising 2000 vortex realizations is generated to span a wide range of quality conditions. A supervised random forest model trained on this dataset achieves 98.45% classification accuracy across four defined vortex-quality categories. Validation using independent experimental measurements demonstrates strong robustness to noise and structural asymmetries. The proposed framework provides a standardized and scalable solution for acoustic vortex metrology, enabling real-time diagnostics and quality control in vortex-based manipulation, imaging, and orbital angular momentum communication systems.

9:25–9:40 Break

Invited Papers

9:40

3aBAa5. Preliminary results of kidney stone comminution using acoustic vortices. Sergio Maldonado-Ortega (Instituto de Instrumentación para Imagen Molecular (i3M), Universitat Politècnica de València (UPV)—Consejo Superior de Investigaciones Científicas (CSIC), Camino de Vera S/N, València 46022, Spain, smalort@upv.es), Enrique González-Mateo, Alejandro Cebrecos (Instituto de Instrumentación para Imagen Molecular (i3M), Universitat Politècnica de València (UPV)—Consejo Superior de Investigaciones Científicas (CSIC), Camino de Vera S/N, 46022, València, Spain), Brenda Morant-Ferrando (Instituto de Investigación Sanitaria La Fe (IIS La Fe), València, Spain), Cesar D. Vera-Donoso (Universidad Católica San Vicente Mártir, València, Spain), Alicia Carrión, Francisco Camarena, and Noé Jiménez (Instituto de Instrumentación para Imagen Molecular (i3M), Universitat Politècnica de València (UPV)—Consejo Superior de Investigaciones Científicas (CSIC), Camino de Vera S/N, 46022, Valencia, Spain)

Ultrasound-based therapies such as extracorporeal shock wave lithotripsy (ESWL) or burst wave lithotripsy (BWL) are used for kidney stone comminution. However, these techniques employ conventional focused beams of mechanical waves. In this work, we explore the use of focused acoustic vortex beams for kidney stone fragmentation. Unlike conventional focused beams, acoustic vortices exhibit helical phase distribution with embedded phase dislocations that carry pseudo-orbital angular momentum. These fields can be generated with spherical phased-array systems, enabling beam steering and control of the vortex topological charge. An experimental study was conducted in a water tank using a 128-channel therapeutic phased-array transducer operating at 1.1 MHz. Excitation consisted of short sinusoidal bursts of 20 cycles delivered at a repetition rate of 200 Hz. Real-time targeting and monitoring were achieved using a confocal 128-element imaging probe, while a camera recorded the fragmentation process. Both artificial stone phantoms (BegoStone) and human calculi were tested. Fragmentation outcomes obtained with single-topological-charge vortices were compared with those produced by standard BWL under conditions of equal peak rarefaction pressure and equal acoustic energy. The vortex-based approach consistently reduced fragmentation time by at least half for all stone types. Furthermore, the distributed focal structure of vortex beams resulted in a lower mechanical index, suggesting a reduced likelihood of damage to surrounding tissue.

10:05

3aBAa6. Small aperture vortex ultrasound transducers for intravenous sonothrombolysis. Xiaoning Jiang (NC State, 911 Oval Dr., Raleigh, NC 27695, xjiang5@ncsu.edu), Jing Wang, Huaiyu Wu (NC State, Raleigh, NC 27695), John Kim (Mech. Eng., Univ. of Michigan, Ann arbor, MI 48109), and Chengzhi Shi (Univ. of Michigan, Ann Arbor, MI 48109)

Vortex ultrasound has been demonstrated *in vitro* for intravenous sonothrombolysis. In this paper, several small-aperture piezoelectric vortex ultrasound transducers were designed and fabricated, followed by acoustic field characterization and cavitation experiments, to further investigate vortex ultrasound mediated thrombolysis. Specifically, controlled acoustic phase differences among transducer elements, implemented through both mechanical and electronic delays, were shown to generate vortex acoustic fields with sufficient peak negative pressure to promote cavitation. Cavitation studies were conducted using microbubbles and nanodroplets over a range of acoustic pressures. The findings suggest that vortex-enhanced cavitation offers significant potential for sonothrombolysis and other advanced therapeutic ultrasound applications.

Contributed Paper

10:30

3aBAa7. Acoustic vortices stimulation on hPSC-derived cardiac organoids. Chenzhe Wang (Mech. Eng., Univ. of Michigan, G.G. Brown Bldg., 2350 Hayward St., Rm. 2651, Ann Arbor, MI 48105, czwang@umich.edu), Shiyu Sun (Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48105), Huaiyu Wu (North Carolina State Univ., Raleigh, NC 27695), Yan Deng (Mech. Eng., The Univ. of Michigan, Ann Arbor, MI 48105), and Chengzhi Shi (Univ. of Michigan, Ann Arbor, MI 48105)

Three-dimensional human cardioids were used to investigate how acoustic vortices applied from an intraluminal device can reshape emergent cardiac-like activity. We integrated a miniaturized vortex-generating piezoelectric transducer at the distal tip of a model catheter to produce focused beams carrying orbital angular momentum. Individual cardioids were positioned in the vortex field and continuously monitored by bright-field videography with motion tracking. Baseline recordings showed stable, self-organized beating patterns. When pulsed acoustic vortices were applied, the beating rhythm was consistently modified. We observed changes in beating rate, alterations in beat-to-beat regularity, with beats becoming transiently entrained to the vortex pulse sequence. These rhythm changes were reversible upon cessation of stimulation, with cardioids gradually returning toward their original beating pattern. By demonstrating that catheter-delivered acoustic vortices can dynamically and reversibly tune the spontaneous rhythm of human cardioids, this work outlines a new, noncontact modality for local rhythm control. The approach establishes a benchtop platform to design and screen vortex-based pulse schemes and suggests future minimally invasive strategies where patterned acoustic vortices modulate regional cardiac function without direct electrical leads or systemic drugs.

Invited Papers

10:45

3aBAa8. Iso-propagation vortex waves for robust long-range underwater acoustics. Jianyu Hua (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, elihua@umich.edu) and Chengzhi Shi (Univ. of Michigan, Ann Arbor, MI 48109)

Acoustic vortex waves carrying orbital angular momentum offer an additional degree of freedom for multiplexing and wavefield control, but their practical use in underwater environments is limited by topological-charge-dependent diffraction, beam spreading, and sensitivity to multipath and turbulence. We investigate acoustic iso-propagation vortex (IPV) waves, a class of structured fields engineered so that their propagation envelope is effectively independent of topological charge while preserving the helical phase structure required for modal multiplexing. IPVs are synthesized through controlled superposition of vortex modes using phased transducer arrays and are evaluated in representative shallow-water environments using ray-based propagation modeling. By combining ray tracing with coherent field reconstruction, we assess beam width, phase stability, and modal integrity over range in the presence of refraction, boundary interactions, and multipath. Results show that, unlike conventional vortex beams, IPVs maintain nearly constant beam size and divergence across modes, leading to improved robustness against environmental distortion and reduced inter-modal coupling. These properties enable more uniform reception and reliable demultiplexing without increasing aperture size. The findings suggest that IPV waves provide a robust foundation for underwater acoustics, with implications for high-capacity communication, sensing, and adaptive field control in complex ocean environments.

11:10

3aBAa9. Vortex ultrasound in drug delivery applications. Wei-Chen Lo (Dept. of Biomedical Eng. and Environ. Sci., National Tsing Hua Univ., Hsinchu, Taiwan), Ching-Hsiang Fan (Dept. of Biomedical Eng., National Cheng Kung Univ., Tainan, Taiwan), Elaine Huang (Dept. of Biomedical Eng. and Environ. Sci., National Tsing Hua Univ., Hsinchu, Taiwan), and Chih-Kuang Yeh (Dept. of Biomedical Eng. and Environ. Sci., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu 30013, Taiwan, ckyeh@mx.nthu.edu.tw)

Acoustic vortex tweezers (AVT) enable contactless trapping and manipulation of gas microbubbles using helical ultrasound wavefronts that carry orbital angular momentum and generate a toroidal acoustic potential. In a tornado-inspired implementation, the vortex beam forms a stable 3-D trap that captures individual microbubbles against buoyancy and flow, and then transports, rotates, merges, or releases them with programmable trajectories. This deterministic control creates a new microbubble robotics modality that decouples positioning from destructive insonation. Leveraging this capability, AVT can intentionally cluster microbubbles to amplify cavitation activity at a designated site, thereby enhancing drug delivery while maintaining spatial confinement. Vortex-driven clustering concentrates acoustic nuclei, promotes repeatable cavitation events, and increases local permeability and payload transport relative to dispersed

bubbles. By mapping operational regimes of trapping, clustering, radiation-force action, and bubble destruction, the combined studies provide design rules for selecting frequency, pressure, duty cycle, and vortex order for either gentle manipulation or strong cavitation-assisted delivery. Together, these results position AVT as a programmable ultrasound platform for precision microbubble control and cavitation-enhanced therapeutics, including targeted vascular delivery.

Session 3aBAb

Biomedical Acoustics: General Topics in Biomedical Acoustics: Quantitative Ultrasound and Doppler

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

Contributed Papers

8:00

3aBAb1. Quantitative lung ultrasound spectroscopy versus computed tomography: A comparative study to differentiate pneumonia and cardiogenic pulmonary edema. Mattia Perpenti (Dept. of Information and Commun. Eng., Univ. of Trento, Via Sommarive, 9, 38123 Povo TN, Trento 38123, Italy, mattia.perpenti@unitn.it), Eleonora Balzani (Univ. of Trento, Trento, Italy), Federico Mento (Dept. of Information and Commun. Eng., Univ. of Trento, Povo, Trento, Trento (TN), Italy), Claudia Marinaro (Emergency Dept., Humanitas Gavazzeni, Bergamo, Italy), Giacomo Bellani (Univ. of Trento, Trento, Italy), Tiziano Perrone (Emergency Dept., Humanitas Gavazzeni, Bergamo, Italy), and Libertario Demi (Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italia, Italy)

Pneumonia (PNE) and cardiogenic pulmonary edema (CPE) share reduced air–space dimensions and pulmonary edema, making their differentiation with conventional imaging challenging. Standard lung ultrasound (S-LUS) is widely used but relies on subjective interpretation of vertical artifacts (VAs), resulting poorly reproducible and lacking in diagnostic specificity. Quantitative lung ultrasound (Q-LUS) was developed to mitigate these limitations through objective measurements of imaging patterns; however, direct comparisons between Q-LUS and gold-standard imaging are limited. Here, a comparison between computed tomography (CT), S-LUS, and Q-LUS is presented. Data were acquired from 55 patients (28 PNE and 27 CPE). CT and S-LUS images were evaluated to assign semi-quantitative scores. Q-LUS consists of multifrequency radiofrequency (RF) images obtained using convex and linear probes. VAs appearing in RF images were segmented and quantified into three spectral features: native frequency, bandwidth, and total intensity. These parameters were statistically analyzed to derive 15 Q-LUS features per patient (for each spectral feature: min, quartile 25%, 50%, 75%, and max). The diagnostic significance of each imaging modality is evaluated using generalized estimating equation model. Results demonstrated highest area under the curve of 80% for Q-LUS using linear probe, highlighting Q-LUS as the most effective modality for differentiating between PNE and CPE.

8:15

3aBAb2. Quantitative ultrasound of cortical bone: Joint assessment of anatomy, tissue quality, and blood flow. Gabrielle Laloy-Borgna (Dept. of Imaging Phys., Delft Univ. of Technol., 151 Cours Albert Thomas, 69428, Lyon 69003, France, g.laloy-borgna@tudelft.nl), Nastassia Navasiolava (Clinical Res. Ctr., CHU Angers, Angers, France), Pim Hutting (Dept. of Imaging Phys., Delft Univ. of Technol., Delft, Netherlands), Andréa Bertona (Clinical Res. Ctr., CHU Angers, Angers, France), Hervé Locrelle (INSERM, Université Jean Monnet, Mines Saint-Etienne, St-Etienne, France), Jacques-Olivier Fortrat (Clinical Res. Ctr., CHU Angers, Angers, France), Laurence Vico (INSERM, Université Jean Monnet, Mines Saint-Etienne, St-Etienne, France), Marc-Antoine Custaud (Clinical Res. Ctr., CHU Angers, Angers, France), and Guillaume Renaud (Dept. of Imaging Phys., Delft Univ. of Technol., Delft, Netherlands)

We present a quantitative ultrasound framework that enables the joint assessment of bone anatomy, tissue quality, and intra-osseous blood flow from a single acquisition. The scanned region contains different tissue types and is described with multiple homogeneous layers with different wave speed values. Ray tracing modelling refraction and wave-speed anisotropy in cortical bone provides accurate travel times and therefore anatomically accurate image reconstruction. *In vivo* imaging of the human tibial diaphysis demonstrates agreement with peripheral quantitative computed tomography (pQCT). Beyond imaging, the method estimates ultrasound wave-speeds in radial and axial directions, capturing porosity and structural anisotropy of cortical bone, and yielding significant correlations with bone mineral density measured with pQCT ($R^2 \approx 0.45$–0.49). To access vascular dynamics, we introduce a ray-selection strategy that suppresses specular interface echoes and enhances diffuse scattering from moving blood cells by promoting oblique incidence at the bone cortex, enabling power Doppler imaging within the bone cortex and marrow. Arterial occlusion produces reproducible changes in Doppler signal and reveals reactive hyperemia within cortical bone. This work demonstrates how advanced beamforming, anisotropic propagation modelling, and tailored Doppler processing extend ultrasound imaging to the anatomical, structural, and vascular characterization of highly aberrating tissues such as bone.

8:30

3aBAb3. Noninvasive assessment of intracranial pressure using Nakagami-based quantitative ultrasound. Ladan Yazdani (Dept. of Radiology, Children's Hospital of Philadelphia, 3615 Civic Ctr. Blvd., Fl. 4, Rm. 409G, Philadelphia, PA 19104, yazdanil@chop.edu), Yibo Wu (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218), Sophie Haddad, Aref Ghaderi, Katherine Luo, Todd J. Kilbaugh (Dept. of Radiology, Children's Hospital of Philadelphia, Philadelphia, PA 19104), Joseph Katz (Dept. of Mech. Eng., Johns Hopkins Univ., Baltimore, MD 21218), and Misun Hwang (Dept. of Radiology, Children's Hospital of Philadelphia, Philadelphia, PA 19104)

Noninvasive assessment of intracranial pressure (ICP) remains a significant clinical challenge. This study investigates quantitative ultrasound based on the Nakagami statistical model as an acoustic biomarker of ICP changes. Ultrasound radiofrequency data were acquired from brain tissue in three controlled preclinical porcine models under baseline conditions and following elevated ICP. Each animal underwent a baseline acquisition and a single elevated ICP condition (20, 30, or 40 mmHg), enabling evaluation across multiple ICP levels. Envelope statistics were analyzed using a sliding-window Nakagami model to estimate the shape (m) and scale factor (Ω) parameters, which characterize the statistical properties of ultrasonic backscatter from tissue microstructure. Nakagami parameter maps were spatially averaged within defined regions of interest and correlated with invasively measured ICP values. Both parameters demonstrated systematic decreases with increasing ICP. Spearman correlation analysis revealed significant negative correlations between ICP and the shape parameter ($\rho = -0.83$, $p < 0.01$) and the scale factor parameter ($\rho = -0.93$, $p < 0.01$). Groupwise comparisons further showed clear separation between baseline and elevated ICP conditions. These findings support Nakagami-based quantitative ultrasound as an acoustics-driven, noninvasive approach for monitoring ICP. [Work was supported by NIH R01NS119473-05 and R01NS126469-03].

8:45

3aBAb4. Multicenter clinical study on pleural sliding quantitative characterization by means of multifrequency lung ultrasound. Chiara Zangrandi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Povo, TN 38123, Italy, chiara.zangrandi@unitn.it), Federico Mento (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Povo, Trento, Trento (TN), Italy), Mattia Perpenti (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy), Giovanni Pierro (Rome, Italy), Alessandro Perrotta (Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Rome, Italy), Tiziano Perrone (Dept. of Internal Medicine, Humanitas Gavezzeni, Bergamo, Italy), Andrea Smargiassi (Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Gemelli, Italy), Riccardo Inchingolo (Fondazione Policlinico Universitario Agostino Gemelli IRCCS, Gemelli, Italy), and Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy)

Quantitative lung ultrasound (LUS) offers a way to overcome the visual interpretation of imaging patterns. A well-known imaging pattern is pleural sliding, defined as the lateral motion of the pleural line in LUS videos. To quantify this pattern, we proposed a methodology to estimate the velocity of pleural sliding (*SlV*), inspired by functional ultrasound (fUS). In this study, we apply this methodology to two datasets from different centers and assess the differentiating potential of *SlV* through binary classification. Data were acquired with a multifrequency approach using a linear probe and the Ultrasound Advanced Open Platform (ULA-OP). The datasets comprise 536 multifrequency LUS videos from 79 patients: 13 affected by pneumonia, 15 by cardiogenic pulmonary edema (CPE), 19 by chronic obstructive pulmonary disease (COPD), 20 by fibrosis, and 12 by fibrosis exacerbated. The resulting SlV values were within the physiological range. On average, COPD is characterized by the lowest SlV and pneumonia by the highest. These differences are also reflected in binary classification performance, reaching accuracies up to 92.48% when differentiating between COPD and pneumonia. These results demonstrate the potential of this methodology to extract clinically valuable information that can be implemented in future diagnostic processes.

9:00

3aBAb5. Comparing the color Doppler ultrasound twinkling artifact on *ex vivo* kidney stones between Verasonics Vantage and NXT research ultrasound systems. Andrea Castillo Manas (Mech. Eng., Stony Brook Univeristy, 100 Nicolls Rd., Stony Brook, NY 11794, acastillomanas@gmail.com), Jacob C. Elliott (The Penn State Univ., University Park, PA 16802), and Julianna C. Simon (Biomedical Acoust., Penn State Univ., State College, PA 16802)

The color Doppler ultrasound twinkling artifact improves kidney stone detection compared to traditional B-mode ultrasound imaging. Currently, the twinkling artifact is found on about 60% of kidney stones and may depend on stone composition and imaging machine settings. Here, we compare the color Doppler twinkling artifact on *ex vivo* kidney stones between the Verasonics Vantage® NXT and the Vantage®-128 ultrasound machines. Seven *ex vivo* kidney stones of varying compositions were imaged three times in flash mode Doppler using a Philips/ATL L7-4 transducer (5.2 MHz) in degassed water. The Doppler power within a stone's region-of-interest was calculated and normalized with a depth-matched background region of interest. A two-tailed paired t-test was used to evaluate statistical significance. Across all stones, twinkling was 1.3 ± 0.20 with the Vantage compared to 2.1 ± 0.2 with NXT ($p < 0.001$). As we found frame numbers were not quite equal, we repeated the scans and found more similar, but still statistically different, results with twinkling of 1.8 ± 0.3 with the Vantage compared to 2.1 ± 0.5 with the NXT ($p = 0.05$). These results suggest that the NXT produces more twinkling compared to the Vantage −128. [Work supported by NSF CAREER 1943937 and ASA SURIEA].

9:15

3aBAb6. Passive cavitation detection analysis of color Doppler twinkling materials with varying levels of peak negative pressure. Benjamin Wood (Dept. of Radiology, Mayo Clinic, 200 1st St. SW, Rochester, MN 55902, wood.benjamin@mayo.edu), Md Aktharuzzaman, Christine Lee, and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN 55902)

The color Doppler twinkling artifact has been theorized to be attributed to existing microbubbles or cavitation occurring on an object. Twinkling has been reported for objects with rough surfaces like kidney stones, sandpaper, and polymethyl methacrylate (PMMA) bone cement. We used passive cavitation detection (PCD) as a method for evaluating the presence of ultrasound-induced bubble oscillation while varying peak negative pressures (PNP). We evaluated a twinkling PMMA sample, a PMMA sample with intentionally lower twinkling (PMMALT), and a nontwinkling metal rod (M). Ultrasound scans were conducted with a Verasonics L11-4v probe on a V1 system from $f_0 = 3$–6 MHz Doppler transmit frequency and 5–35 V Doppler transmit voltage. A Sonic Concepts Y-107 PCD probe was used to collect data during Doppler scanning. The Fourier spectra for each $f_0 = 3$, 4, 5, or 6 MHz showed increased inharmonic band values compared to PMMALT and M. Both twinkling and inharmonic band magnitudes increased with PNP on the PMMA sample. The PMMALT sample showed some inharmonic elevation and twinkling, but M showed none. Correlations between twinkling and inharmonic signal were found for the PMMA sample at all Doppler transmit frequencies ($\rho \geq 0.691$, $p \leq 0.023$).

9:30

3aBAb7. Comparing power Doppler and ultrasound contrast agent imaging methods on *ex vivo* kidney stones. Jay M. Cliftmann (Graduate Program in Acoust., Penn State Univ., Pennsylvania State, University Park, PA 16802, jmc9636@psu.edu), Veer Gowda, and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The color Doppler twinkling artifact can help detect kidney stones but only appears on ~60% of stones. As the artifact is theorized to be caused by crevice bubbles, we compared power Doppler to contrast agent methods developed for imaging bubbles on 20 kidney stones in *ex vivo* porcine kidneys. Using an L7-4 on a Vantage NXT Verasonics ultrasound system with $P_+/P_- \approx 4.1/1.8$ MPa for power Doppler and $P_+/P_- \approx 2.8/1.0$ MPa for the

other methods, beamformed IQ data were acquired from plane-wave imaging methods: B-mode, amplitude modulation (AM) (5 versions), pulse inversion (PI), amplitude-modulated pulse inversion (AMPI), power Doppler, and pulse-inverted power Doppler. We placed region-of-interest boxes around stones and around echogenic tissue to find signal-to-tissue ratios (STRs) in dB. Compared to B-mode, the mean STRs across the stones were 9.3 ± 7.1 dB for AM, 4.9 ± 4.9 dB for PI, -0.8 ± 8.0 dB for power Doppler, and 3.5 ± 5.2 dB for pulse-inverted power Doppler. Compared to prior studies, observed higher B-mode STRs may be due to trapped gases or reduced attenuation in the benchtop setup. These results suggest that contrast imaging methods should be further explored to improve kidney stone detection with ultrasound [Work supported by NSF CAREER #1943937].

WEDNESDAY MORNING, 13 MAY 2026 FRANKLIN HALL 12, 8:55 A.M. TO 12:00 NOON

Session 3aEA

Engineering Acoustics and Physical Acoustics: Artificial Intelligence (AI) Meets Electroacoustic Transducers: Emerging Methods and Applications

Noori Kim, Cochair
Purdue University, 401 N. Grant St., Rm. 115, West Lafayette, IN 47907

Zane Rusk, Cochair
Bose,

Chair's Introduction—8:55

Invited Papers

9:00

3aEA1. A comparison of clinical outcomes and model-based predictions with a hearing aid embedded acoustic denoising network. Jason Galster (Sonova Group, 750 N Commons Dr., Aurora, IL 60504, jason.galster@sonova.com)

The treatment of hearing loss with hearing aids first aims to address the reduced audibility of sound. However, restoration of audibility alone will not meet the needs of all listeners. Even for experienced hearing aid wearers, the most common complaints relate to understanding speech in noisy environments. This presentation will describe the design of a hearing aid-embedded denoising network that is optimized to enhance speech in background noise while maintaining clinically acceptable sound quality. The results of multiple clinical trials will be summarized, each showing significantly improved speech recognition in noise when the denoising network is engaged. These behavioral data will be compared to the output of predictive models of speech recognition that include traditional fixed parameter approaches and comparatively modern AI-based systems that leverage techniques like automatic speech recognition.

9:20

3aEA2. Virtual listener based machine-learning optimization of hearing-aid processing. Ahsan J. Cheema (Eaton-Peabody Labs, Otolaryngology-HNS, Harvard Med. School, Harvard University, Boston, MA 02114, ahsancheema@g.harvard.edu) and Sunil Puria (Eaton-Peabody Labs, Otolaryngology-HNS, Harvard Med. School, Boston, MA)

Determining optimal hearing-aid settings is challenging because sound quality and speech understanding depend on individual's hearing loss, auditory-nerve health, cognitive/linguistic factors, and personal preference. Clinical fittings are largely empirical which limits generalization and making systematic exploration of the parameter-space impractical. We developed a computational, machine-learning framework that enables large-scale evaluation of hearing-aid parameter space using "virtual" listeners with controllable auditory and cognitive profiles. Clean and degraded speech stimuli were processed by a wide dynamic range compression hearing-aid while varying key parameters—including channel-wise gain, compression ratio, expansion, kneepoints, and attack/release time constants. Processed signals were passed through an auditory-periphery model, parameterized by individual audiometric loss and neural integrity, to produce time–frequency neurograms. These neurograms are inputs to transformer-based recognition models, augmented with a module to capture variability associated with cognitive/central factors. The trained model's word-recognition accuracy was used as a proxy for virtual listener performance and parameters were searched for that maximize predicted outcomes. This proposed optimization workflow is broadly applicable to other electroacoustic transducers where performance depends on high-dimensional parameter-space that is costly/impractical to explore experimentally. [Work supported by R01-DC07910 NIH-NIDCD.]

9:40

3aEA3. Multimodal artificial intelligence based sensing for hearables. Sile Yin (Res., Bose Corp., 100 The Mountain Rd., Framingham, MA 01701, sile_yin@bose.com), Marko Stamenovic (Res., Bose Corp., Framingham, MA), and Shuo Zhang (Res., Bose Corp., Framingham, MA 01701)

With the rapid advancement of deep neural networks that process information from multiple modalities—such as video, audio, and text—wearables can leverage multimodal AI models to solve challenging audio tasks, with potential applications in voice communications and selective hearing with multiple electroacoustic sensors and beyond. In this talk, we discuss recent work from Bose Research which incorporate video, bone-conduction vibration sensors, and ultrasonic sensing to provide improved capabilities on tasks such as audio source separation and keyword spotting. First, we present a system that leverages pretrained visual embeddings to perform audio-visual speech enhancement (AVSE) from the video and audio stream in real time. Second, we highlight work on target speaker extraction (TSE), comparing two strategies for targeting the wearer's voice: personalized speech enhancement (PSE), which uses enrollment utterances to represent the target, and auxiliary-sensor speech enhancement (AS-SE), which leverages in-ear microphones carrying target-specific cues. Example applications include the recently released Bose SpeechClarity technology for voice communications on earbuds. Finally, we present other work on Whisper Keyword Spotting (WKWS) with ultrasonic sensing, using existing electroacoustics actuator and sensors on a Bose headphone.

Contributed Paper

10:00

3aEA4. A geometry-agnostic deep learning framework for volumetric reconstruction of internal flow-noise from sparse boundary measurements. Trisha Campanaro (Penn State, 777 West harrisburg Pike, Middletown, PA 17057, txm5437@psu.edu) and Issam Abu-Mahfouz (Penn State, Middletown, PA)

The characterization of flow-induced noise and cavitation in complex fluid-structure systems often requires invasive instrumentation or computationally prohibitive high-fidelity simulations. This study proposes a nonintrusive "virtual sensing" framework capable of reconstructing internal aeroacoustic pressure fields using a sparse external sensor array. Training datasets are generated using STAR-CCM+, simulating turbulent flow through a series of benchmarked duct geometries designed to induce diverse flow separation and cavitation regimes. A convolutional neural network (CNN) architecture is trained to map these boundary pressure fluctuations to full volumetric flow-noise source terms. Unlike traditional transfer-function approaches, the proposed deep learning model captures the non-linear coupling and transient events characteristic of early-stage cavitation. Preliminary validation against CFD ground truth demonstrates that the framework predicts internal sound pressure levels (SPL) and identifies cavitation onset with greater than 90% accuracy using minimal downstream sensors. This methodology offers a robust tool for real-time monitoring of industrial piping and propulsion systems where internal sensing is infeasible, providing a scalable solution for acoustic structural health monitoring without requiring system-specific calibration.

10:15–10:35 Break

Invited Papers

10:35

3aEA5. Human temporal-bone measurements of an ultrasound-based MEMS audio transducer. Sunil Puria (Eaton-Peabody Labs, Otolaryngology-HNS, Harvard Med. School, 243 Charles St., Boston, MA 02114, Sunil_Puria@meei.harvard.edu), Kevin O'Connor, and Jeffrey Cheng (Eaton-Peabody Labs, Otolaryngology-HNS, Harvard Med. School, Boston, MA 02114)

Ultrasound-based MEMS transducers for in-ear applications can generate audible sound within the ear canal through ultrasonic modulation and demodulation of sound waves, enabling improved acoustic performance versus conventional earbuds. However, little has been reported on the acoustic coupling of such devices to the human ear, particularly at the ultrasonic operating-point frequency. This information is needed to identify any potential safety concerns regarding the prolonged personal use of these devices, so we conducted broadband measurements on one such device in human cadaveric temporal bones using an optical microphone to obtain pressure in the ear canal via a custom coupling fixture, and a 3-D laser Doppler vibrometer to measure velocity near the stapes head and cochlear promontory. We obtained stepped-tone frequency responses up to 220 kHz, as well as single-tone-driven FFT responses to investigate device-specific peaks at the ∼98.85-kHz operating point and its ∼197.7-kHz second harmonic. While the ear-canal-pressure peak at the second-harmonic frequency (∼90 dB SPL) generally exceeded the pressure peak at the operating-point frequency (∼80 dB SPL), only at the operating-point frequency were significant stapes-head and cochlear-promontory velocity peaks observed. These operating-point velocity peaks were approximately 0.01–0.04 mm/s at both cochlear locations. [Work supported by a research contract from xMEMS Labs to Mass Eye and Ear.]

10:55

3aEA6. Automated discovery of electrode layouts for piezoelectric microphone optimization. Will Leskowitz (Elec. and Comput. Eng., Univ. of Texas at Austin, 2501 Speedway, Austin, TX 78712, wpl@utexas.edu), Ehsan Vatankhah, Yuqi Meng, Zihuan Liu, Yen-Chen Wang, Zhichen Yan, and Neal A. Hall (Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX)

Electrode layout in piezoelectric MEMS microphones is a high-leverage design variable that directly shapes the sensitivity–capacitance–noise trade space and ultimately limits achievable system SNR. Despite this importance, electrode geometries are typically chosen through manual iteration over a large design space, restricting exploration and favoring familiar patterns. We present a physics-in-the-

loop AI framework for automated discovery of manufacturable electrode topologies, demonstrated on $LiNbO_3$ MEMS microphones. The approach couples a neuroevolutionary optimizer to a multiphysics finite-element model in COMSOL, enabling closed-loop search over function-defined electrode geometries guided by electroacoustic performance. Candidate layouts are generated by a structured generative model and decoded into fabrication-ready sensing and ground electrodes under explicit constraints on connectivity, spacing, and border contact. Each feasible design is evaluated by solving the coupled electrostatic–mechanical problem and extracting capacitance and open-circuit sensitivity, which are mapped to system-relevant objectives including harvested electrical energy and predicted front-end-limited signal-to-noise ratio using a compact noise model. The framework enables exploration of large electrode design spaces while remaining grounded in high-fidelity physics and illustrates how AI-assisted optimization can accelerate the design of electroacoustic transducers beyond manual or parametric approaches.

Contributed Papers

11:15

3aEA7. Machine learning assisted rapid multi-objective optimization of tFUS phased-array transducer design for deep-brain neuromodulation in mice. Sadman H. Labib (Mech. Eng., Texas Tech Univ., 805 Boston Ave., Dept. of Mech. Eng., Lubbock, TX 79409, salabib@ttu.edu) and Jingfei Liu (Mech. Eng., Texas Tech Univ., Lubbock, TX 79409)

Transcranial focused ultrasound (tFUS) is an emerging noninvasive approach for deep-brain neuromodulation, yet reliable targeting in mice is difficult because the skull distorts and attenuates acoustic waves. Phased-array transducers provide electronic focusing and steering, but outcomes depend strongly on design choices such as aperture geometry, frequency, element count, and element size. We propose a data-driven workflow to optimize phased-array designs for hypothalamic stimulation. Full-wave acoustic simulations are performed in a CT-derived 3-D mouse head model that captures heterogeneous soft tissue and skull structure. Candidate arrays are generated by varying frequency (200–1500 kHz), number of elements (16–128), and element diameter (300–550 μm), with elements randomly distributed on a bowl-shaped aperture. For each configuration, we quantify focal pressure gain, focal size and shape, targeting error relative to the hypothalamus, and side-lobe levels. The resulting dataset trains and tests surrogate models (random forest) that rapidly predict acoustic metrics from design variables. Then the surrogate is coupled to multi-objective optimization (NSGA-II) to identify pareto-optimal designs that maximize focal intensity while minimizing distortion and off-target exposure, substantially reducing overall computational cost compared with exhaustive simulation. The framework enables rapid exploration of array design space and provides practical guidance for mouse tFUS transducer development.

11:30

3aEA8. Machine learning-accelerated design of phononic crystals: From optimized experimental design through simulation to experimental validation. Samarjith Biswas (Dept. of Mater. Sci. and Eng., Univ. of Arizona, 750 N Cherry Ave., Tucson, AZ 85719, samarjithbiswas@arizona.edu), Kamaljeet Singh, Wataru Takeda, Pierre A. Deymier, Keith Runge, Hao (Helen) Zhang, Pierre Lucas, and Krishna Muralidharan (Dept. of Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ 85719)

A progressive machine learning framework for phononic crystal bandgap prediction is presented, advancing from geometric to material-geometric optimization with experimental validation. Two-dimensional phononic crystals with hexagonal lattices of circular holes were investigated through systematic design of experiments. In the initial two-parameter study, material properties remained fixed while Latin Hypercube sampling with maximin optimization generated parameter combinations for plate thickness and filling fraction, yielding 1000 COMSOL eigenfrequency simulations. Six machine learning algorithms—XGBoost, Random Forest, LightGBM, Gradient Boosting, multilayer perceptron (MLP), and support vector regression—were trained and compared. MLP achieved optimal accuracy ($R^2 = 0.874$), and optimal configurations were fabricated and experimentally validated. The second study expanded to five parameters, adding density, Young's modulus, and Poisson's ratio using a multi-phase experimental design combining Latin Hypercube sampling, Sobol sequences, and realistic material combinations. Physics-informed features capturing impedance mismatch and nonlinear geometric coupling were engineered from 784 simulations. Gradient boosting achieved the best performance ($R^2 = 0.827$), with bandgap predictions again validated through high-throughput fabrication and experimental characterization. This closed-loop approach provides a practical framework for accelerated phononic metamaterial development.

11:45

3aEA9. Speakers as sensors: Impedance-based acoustic load analysis via electroacoustic coupling and artificial intelligence. Noori Kim (Purdue Univ., 401 N. Grant St., Rm. 115, West Lafayette, IN 47907, kim4147@purdue.edu)

This study presents ongoing efforts to investigate speakers as sensing units. Specifically, the research examines the use of speakers (earphones) as sensors for detecting acoustic loads through variations in electrical impedance. By leveraging electroacoustic coupling and AI-based analysis, impedance-derived features are used to predict earphone insertion depth with high accuracy. A dense neural network achieved an overall classification accuracy of 87%, which improved to 91% when trained with speaker-specific data and up to 99% for individual units. Performance was further enhanced by identifying and utilizing optimal data regions. In addition, the proposed classification approach is discussed and compared with a regression-based formulation. This work integrates fundamental electroacoustic principles with modern AI techniques, establishing a foundation for smart, low-cost, and portable diagnostic systems for ear health and related biomedical applications.

Session 3aID

Interdisciplinary and Education in Acoustics: Student Paper Competition—Seminal Acoustics Papers

Bhisham Sharma, Cochair
Mechanical and Aerospace Engineering, Michigan Tech Univ., 1400 Townsend Drive, Houghton, MI 49931

Julianna C. Simon, Cochair
Graduate Program in Acoustics, Penn. State Univ., Penn State, 201E Appl. Sci. Build., University Park, PA 16802

Chair's Introduction—7:55

Contributed Papers

8:00

3aID1. Focused ultrasound mechanical effects in the advent of histotripsy. Grace M. Wood (Graduate Program in Acoust., Penn State, Old Main, University Park, PA 16802, gmw5253@psu.edu)

Focused ultrasound (FUS) is a therapeutic modality used to induce thermal and mechanical bioeffects in a millimeter-sized focal volume. Thermal ablation (rapid heating for tissue destruction) has been used clinically since 2004, but the potential for off-target damage is high if heat is not allowed to dissipate between pulses. This paper by Parsons *et al.* sought to harness the mechanical effects of FUS through cavitation to produce more localized treatment volumes. Porcine myocardium was treated with a 750-kHz therapeutic array. Pulses consisted of a primary 14-μs pulse at p- = 22-MPa followed by a 14-μs sustaining pulse. Seven amplitudes from p- = 0–22-MPa were investigated at a 0.33-kHz pulse repetition frequency (PRF) to determine the influence of sustained cavitation activity. Six PRFs from 0.07–3.1-kHz were also investigated with the preferred sustaining pulse amplitude. Following sonication, lesion cross-sections were observed grossly and histologically. Results showed sustaining pulses at amplitudes up to 6.9-MPa with PRFs up to 0.33-kHz produced a liquid homogenate that was easily irrigated from a lesion with clearly demarcated boundaries. These treatments emphasizing mechanical bioeffects were termed histotripsy—"histo" meaning tissue and "-tripsy" meaning to break apart. The advent of histotripsy led to a new area of biomedical acoustics research with a focus on controlled treatment outcomes and the potential for passive acoustic monitoring.

8:15

3aID2. Review of Stojanovic (1996) on high-speed underwater acoustic communications. Benjamin Viner (Whiting School of Eng., Johns Hopkins Univ., 11100 Johns Hopkins Rd., 8-224, Laurel, MD 20723, ben.viner@jhuapl.edu)

Before the 1990s, underwater acoustic communications were limited to extremely low data rates. Unlike radio communications, the ocean is quite unforgiving, and signals can become significantly distorted due to strong multipath arrivals, doppler shifting, and constantly changing environmental conditions. In 1996, Milica Stojanovic published her paper, "Recent Advances in High-Speed Underwater Acoustic Communications" [*IEEE J. Ocean. Eng.*, **21**, 1996], showing that the challenges of an underwater environment could be understood and corrected to achieve higher data rates. This review summarizes the main methods and contributions of the paper and explains why it has remained influential over the last 30 years. Stojanovic demonstrated that acoustic communications can support much higher data rates over longer distances by using both phase-coherent signals and adaptive receivers. Importantly, the author supported her claims with experimental results from ocean testing, proving the practical applications of her theory. By reframing challenges in acoustic communications as an engineering problem instead of a physical limitation, the paper has inspired much research in its field. To this day, its ideas continue to influence acoustic modem development and ongoing research in high-rate communications.

8:30

3aID3. Westervelt and the conception of the parametric array. Philip G. Kaufinger (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Appl. Res. Labs. and Walker Dept. of Mech. Eng., The University of Texas at Austin, Austin, TX 78758, pkaufinger@utexas.edu)

The nonlinear phenomenon called scattering of sound by sound occurs when two collimated sound beams with different frequencies intersect, forming a finite region of interaction that generates sound at the sum and difference of the two source frequencies. One of the most important figures in the early study of sound beam interactions was Peter J. Westervelt. Beginning with his presentation at the 59th Meeting of the Acoustical Society of America [Westervelt, *JASA* **32**, 934–935 (1960)] and culminating with his short seminal nonlinear acoustics paper [Westervelt, *JASA* **35**, 535–537 (1963)], Westervelt laid the theoretical foundation of the parametric array (also known as end-fire array) by having the collimated sound beams be collinear. In a parametric array, nonlinear interactions between two primary beams in the region in front of the source generate a difference-frequency sound beam that is substantially more directional than if the source radiated the same frequency directly, with sidelobes significantly suppressed compared to those of the primary beams. This presentation will discuss the history surrounding the parametric array and Westervelt's contribution to its development, as well as highlight subsequent literature further exploring the parametric array. [Work was supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

8:45

3aID4. Ocean ambient noise: Earthquakes, volcanos, ships, and so many bubbles. Andrew Penrod (Phys. and Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, andrewpenrod2023@gmail.com) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Understanding oceanic ambient noise is pivotal to progress in areas encompassing military operations to marine biology. With such diverse depths, salinities, climate, animal life, and even time of day across the globe, establishing a uniform method to accurately predict and comprehend

underwater noise was of the utmost importance. Gordon M. Wenz published "Acoustic ambient noise in the ocean: Spectra and sources" in December 1962, revealing that the many sources of underwater sound could be analyzed as individual parts of a greater whole. Combining "prevailing noises" found worldwide with regional conditions, his work was revolutionary in the field of acoustics and is still used today as a standard—known as the Wenz Curves. His breakthrough transformed research across many disciplines and expanded our understanding of environmental science while enhancing the functionality of marine engineering. In recent years, updates to Wenz Curves have been published, providing insights on ambient noise with greater wind speeds and higher frequencies than the Wenz Curves demonstrated. This poster compares the original with updated noise models. Wenz laid the foundation for our understanding of underwater acoustic ambient noise over 60 years ago, and we continue to build off of his legacy today.

9:00

3aID5. Biot's theory of elastic waves in porous media: A foundation for modern acoustic material design. Jennifer L. Celeste (Michigan Technolog. Univ., 1400 Townsend Dr., Houghton, MI 49931, jenniferlceleste@gmail.com)

In 1956, Maurice Biot published a foundational theory describing how elastic waves propagate through fluid-saturated porous solids. His work established a mathematical framework for understanding wave motion in materials where solid and fluid phases interact, predicting two compressional waves and one shear wave with frequency-dependent attenuation. A key insight of Biot's theory is that sound energy is dissipated through viscous motion of the fluid within the pore structure, making porous media effective sound absorbers. Nearly 70 years later, this theory remains central to acoustic material design. Modern fabrication methods, including additive manufacturing and architected porous materials, allow engineers to precisely control pore geometry and material structure. Despite these advances, Biot's equations continue to provide the predictive foundation needed to model acoustic behavior and optimize noise-reducing performance. This presentation highlights Biot's original contributions and explains why his theory remains essential for developing next-generation sound-absorbing materials.

9:15

3aID6. Overview of Johnson, Koplik, and Dashen's 1987 paper on: "Theory of dynamic permeability and tortuosity in fluid-saturated porous media". Anthony Ciletti (Mech. and Aerosp. Eng., Michigan Technolog. Univ., 1757 Upland Ave., Houghton, MI 49931, avcilett@mtu.edu)

The intricate internal geometry of porous materials poses a challenge to obtaining key properties needed to predict their behavior. In their 1987 paper, David Linden Johnson, Joel Koplik, and Roger Dashen of Schlumberger-Doll Research derive a model of dynamic permeability and dynamic tortuosity, bridging known behavior at the low- and high-frequency limits. Finding their model comparable to simpler models to describe parallel-running or networked tubes, the authors expound on the applicability of their model to porous materials generally. The authors conclude by showing how helium superfluid can be employed to probe important parameters at the frequency limits, and discuss the implication of a fractal internal surface on their model. Their paper is foundational to the field of poroacoustics, as demonstrated by its 3168 and counting Google Scholar citations. Among the citations are Champoux and Allard (1991), Pride *et al.* (1993), and Lafarge *et al.* (1995), which extended and improved upon Johnson, Koplik, and Dashen's model. The resulting JCA, JCAL, and JCAPL models became indispensable for poroacosutic material analysis. Equipped with these models, researchers have characterized, predicted, and optimized the acoustic performance of a wide range of porous media.

9:30

3aID7. Richard L. Weaver's contribution to attenuation and grain noise modeling in polycrystalline media. Olivia J. Cook (Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci., University Park, PA 16801, ojc3@psu.edu), Christopher M. Kube, and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Richard L. Weaver's *Diffusivity of ultrasound in polycrystals* (1990) laid the foundation for many theoretical and experimental studies of microstructure-induced scattering effects on ultrasonic wave propagation. The work introduced a Green's function-based framework for describing wave propagation in elastic media with random property fluctuations, addressing the behavior of both the mean field and the scattered field. The ensemble-averaged Green's function was expressed using the Dyson equation, yielding closed-form expressions for the complex propagation constant and attenuation within the Born approximation. These expressions provided physical insight into frequency- and grain size-dependent attenuation while also offering a practical alternative to purely numerical or regime-limited models. As a result, Weaver's attenuation formulation was readily adopted and extended by later authors to investigate microstructural features beyond equiaxed grains characterized by a single length scale. In addition, Weaver formulated the mean-square Green's function through the Bethe–Salpeter equation, providing a statistical description of the zero-mean scattered field. Although initially used to derive expressions for wave diffusivity, this component later served as the basis for theoretically complete descriptions of diffuse ultrasonic scattering, including grain noise measurements, with commonly used single-scattering models emerging as a limiting case.

Session 3aNS

Noise: Sounds of Freedom

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Jeanine Botta, Cochair
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Chair's Introduction—9:00

Invited Papers

9:05

3aNS1. The revitalising impact of "sacred acoustic environment" on the quest for transcendence, freedom, and a just society. Menino A. Tavares (Acoust. & Soundscaping of Revitalising Sacred Environment, Society of Pilar, 510 Wernett St., Del Rio, TX 78840, minino.tavares@archsa.org), Wade R. Bray (HEAD Acoust., Inc., Brighton, MI 48116), and Rebecca Simmons (Las Missiones, Archdiocese of San Antonio, San Antonio, TX 78210)

A "sacred space" meets the human need for transcendence, freedom, and righteous living. Therefore, Acoustical experiences in a "sacred space" are not mere observations of "impulse-responses" (based on ISO 3382 standards) nor mere "soundscapes" (based on ISO 12913-1/2/3)—They are impulse-responses designed to evoke soundscapes that are transcendent and revitalizing. Sacred spaces are pleasant, eventful, calm, vibrant, and more than merely affective. This study, comparing a few enclosed and outdoor sacred spaces, shows results of characterizing the transcendent metaphysical impact of "natural sounds" and "live performances" on listeners (congregation) in a sacred space. Metaphysical experiences such as "Heightened Awareness" and "Stillness" are found to be significantly regressing with acoustical perceptions of "loudness", "clarity", "tonality", and "intimacy". These metaphysical (transcendent) experiences are also found to significantly triangulate with binaural interaural XY plots, of binaural parameters: Loudness (Sottek Hearing Model, ECMA-418-2 4th Edition) (N5), Sharpness (Aures) (S), Fluctuation Strength (Sottek Hearing Model, ECMA-418-2 4th Edition) (F), and Tonality (Sottek Hearing Model, ECMA-418-2 3rd Edition) (T) (using ArtemiS SUITE 17). This empirical connection between metaphysical perceptions and acoustic environment provides a scope to orient sacred spaces to satisfy the quest for a free and just human society.

9:25

3aNS2. A quieter sound of freedom: Multiple-jet interactions in the delta IV heavy launch vehicle. Noah L. Pulsipher (Phys. and Astronomy, Brigham Young Univ., Brigham Young University, N 283 Carl F. Eyring Sci. Ctr., Provo, UT 84604, npuls@byu.edu), Kent L. Gee, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The now-retired Delta IV Heavy (DIVH) rocket produced a comparatively subdued acoustic signature for its size and power. With a triple-core configuration powered by three RS-68A engines, DIVH generated 9.4 MN (2.1 million lbf) of liftoff thrust, yet acoustic measurements indicate it was a few decibels quieter than Falcon 9, which consists of nine clustered engines on a single core and produces 7.6 MN (1.7 million lbf) of thrust. This contrast raises questions about how propulsion configuration shapes the sounds of launches and how the "sounds of freedom" associated with access to space might be influenced by multiple-jet interactions. This work presents a synthesis of acoustic data from three DIVH launches, interpreted in the context of prior studies on multiple-jet and multiple-scatterer acoustics. Measured sound power levels and directivity trends are compared with published twin-jet results to assess whether observed reductions are consistent with known interference and scattering mechanisms. Although the literature has not yet addressed sound power reductions, deviations of a few decibels in directivity are consistent with multi-source predictions. Findings suggest pathways for reducing acoustic impact through vehicle design and motivations for further analytical research on multiple-jet interactions. [Work supported by the U.S. Space Force through USACE.]

Contributed Paper

9:45

3aNS3. Modeling the performance of an idealized non-wearable active noise control system for military jet noise. Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, peterkj@byu.edu) and Micah Shepherd (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602)

Noise-induced hearing loss is prevalent among jet aircraft personnel due to the extended exposure to jet noise while operating near the aircraft, despite the use of hearing protection. Active noise control (ANC) has been proposed to supplement traditional hearing protection, especially at low frequencies. To investigate the limitations of this type of ANC system, an accurate model of military jet noise is desirable. An acoustic model has been developed for the T-7A-installed GE F404 engine (Olaveson and Gee 2024, *AIAA J*, **62**). The model defines equivalent sources which produce a pressure field matched to measurements of the aircraft, allowing for the calculation of the cross-spectral density matrix between arbitrary positions. Using this model, this paper will investigate how the pressure field is affected by a simple ANC setup. By enforcing perfect cancelation at the location of an error microphone, we evaluate the best-case scenario for an idealized ANC system. It will be shown how the physical volume of the region of silence surrounding the error microphone depends on both frequency and the location of the error microphone, reference microphone, and cancellation source. The negative impact of the ANC system on the surrounding acoustic field will also be discussed. [Work was supported by the Office of Naval Research.]

Invited Papers

10:00

3aNS4. Increasing access to space within a livable acoustical environment. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu)

Access to space enables advances that define modern society, including global communication, environmental monitoring, scientific discovery, and national security. As launch activity increases worldwide, rockets are becoming essential civic infrastructure rather than rare events. However, high-intensity rocket noise and sonic booms have increased potential to affect communities far beyond spaceports. These sounds could challenge personal freedoms by disrupting residential and natural soundscapes. This paper explores the potential tension between societal benefits of space access and noise impacts experienced by surrounding populations. Addressing this challenge requires advances in acoustic modeling, measurement, and human response, as well as engagement with policy and community stakeholders. As described in a 2025 *Acoustics Today* article by the author (https://doi.org/10.1121/AT.2025.21.1.73), the Acoustical Society of America is uniquely positioned to synthesize these perspectives and help guide responsible solutions that preserve both the freedoms enabled by spaceflight and the right to a livable acoustical environment.

10:20–10:35 Break

Contributed Papers

10:35

3aNS5. Estimating the acoustic efficiency of a model rocket using sound power measurements in a reverberant field. Jacob Sampson (Brigham Young Univ., N284, Provo, UT 84602, jacobbsampson@gmail.com) and Micah Shepherd (Brigham Young Univ., Provo, UT 84602)

Rocket noise has been studied since the Space Race and is critical for successful launches and space missions. Sound power generated by rocket engines is related to the total mechanical power of a rocket plume by a constant known as the acoustic efficiency, which historically ranged from 0.1% to 1% for different rockets. While acoustic studies from full-scale rockets are somewhat prevalent, very little research has been conducted on their small-scale counterparts. It is unknown how small-scale and full-scale rocket noise compare with how much mechanical power is converted to acoustic power. To address this discrepancy, a series of acoustic measurements were performed within a small-scale static rocket test facility on two different small-scale rocket engines. Since the facility is enclosed, a reverberant sound power technique was employed using a calibrated sound source. The sound power of the two engines will be presented in one-third-octave bands and compared with published full-scale data. Additionally, acoustic efficiency was computed for the small-scale rocket engines and found to be inline with recent measurements of full-scale rockets.

10:50

3aNS6. Acoustical analysis of the first demonstration test of the space launch system BOLE solid rocket motor. Elias L. Gettemy (Dept. of Phys., Univ. of Houston, 22502 Westenfield Ln., Katy, TX 77450, eliasgettemy@gmail.com), Kent L. Gee, Makayle S. Kellison (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602), and Jack A. Beyer (NASA Space Flight, South Daytona, FL)

The first horizontal static firing of the Space Launch System's Booster Obsolescence and Life Extension (BOLE) solid rocket motor occurred at Northrop Grumman's Utah facility on June 26, 2025. Near the end of the 2-min, 20-s firing of the approximately 4-million-pound-thrust motor, a nozzle anomaly occurred. This paper characterizes the near-field acoustics of the BOLE motor—the most powerful segmented solid rocket motor ever fired—and correlates acoustic features with events observed in high-resolution video. Near-field noise measurements were collected using a small linear array of microphones located approximately 80 m downstream and lateral to the nozzle and were time-synchronized with video footage of the test. Analysis of running sound pressure levels reveals fluctuations associated with changes in motor thrust and the nozzle gimbaling schedule. During the nozzle anomaly, two large explosive events generated high-intensity shock waves and a sudden shift in peak frequency to less than 4 Hz. Following the anomaly, a sustained decrease in running sound pressure level is observed, consistent with a reduction in motor thrust.

11:05

3aNS7. One-dimensional dynamic mapping of underwater radiated noise from inland waterway ships based on trajectory clustering and acoustic propagation modelling. Zonglin Lei (Key Lab. of Ministry of Education for Hydraulic and Water Transport Eng., Chongqing Jiaotong Univ., No. 66, Xuefu Ave., Nanan District, Chongqing, 400074, China, raymond.ling@foxmail.com), Wenjie Li (Key Lab. of Ministry of Education for Hydraulic and Water Transport Eng., Chongqing Jiaotong Univ., Chongqing, China), Hongbo Du (National Inland Waterway Regulation Eng. Technol. Res. Ctr., Chongqing Jiaotong Univ., Chongqing, China), and Yu Wan (Chongqing Field Observation and Res. Station on the Upper Yangtze River Waterway Ecology, Chongqing Jiaotong Univ., Chongqing, China)

To address the need for assessing underwater radiated noise (URN) from ships navigating inland waterways, this paper proposes a 1-D dynamic noise mapping method based on Automatic Identification System (AIS) data. First, by performing cluster analysis on AIS tracks, typical shipping routes are extracted, and noise calculation points are discretely distributed along these routes. Based on source spectrum model of ships, combined with parameters such as speed and length provided by the AIS, the source spectrum of the ship is calculated. Furthermore, integrating the characteristics of the riverbed substrate and the water column environment, an underwater acoustic propagation model is employed to estimate transmission losses (TL) at different distances. Furthermore, AIS data are processed in 1-min intervals, with the ship noise sound pressure level (SPL) calculated frame-by-frame at discrete points along the route. Employing the principle of energy superposition, the noise contributions from multiple ships are integrated to generate a 24-h dynamic noise distribution sequence. Ultimately, through temporal accumulation, a 1-D noise map representing the equivalent continuous SPL over a full day is generated. This enables the quantification and visualisation of the spatio-temporal characteristics of ship noise, thereby providing a basis for waterway noise management and ecological assessment.

11:20

3aNS8. Understanding the vibrational behavior of Olympic barbells under realistic loads. Joshua Langlois (Graduate Program in Acoust., Penn State Univ., 556 White Course Dr., State College, PA 16801, jgl5306@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA 16801)

The sport of Olympic weightlifting provides a spectacle of strength, power, and precision. The Olympic barbell is the implement used by these athletes in competition. The barbell consists of a cylindrical solid metal shaft, coupled on each end to a sleeve. Olympic barbells have a standard weight, diameter, and length but every brand of barbell will use different steel alloys, coatings, and sleeve coupling methods. Coupling methods vary between bare metal, bushings, or bearings. In this paper, we use experimental modal analysis to study the vibrational mode shapes and frequencies for one barbell under many different loads, and for several barbells under the same load. The mode shapes and frequencies help understand how Olympic weightlifters use the "whip" of the bar to accelerate the bar upwards.

11:35

3aNS9. The influence of diameter profile on the mode shapes and frequencies of baseball bats. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

On opening weekend of the 2025 MLB season, the appearance of "torpedo bats" stirred up controversy and a flurry of media interest for these new "bowling pin" shaped wood bats which deviated from the traditional bat profile shape. While these specific torpedo bats generated interest for a few months, they are certainly not the only variations on the shape profile (diameter as a function of length along the bat) for wood bats that have been used in professional baseball. This talk will explore the influence of shape profile on the resulting flexural bending mode shapes by comparing experimental modal analysis measurements for a variety of wood bats with significantly varying shape profiles. Attention will be primarily focused on the locations of the vibration nodes in the barrel region as these are closely related to the identification of the "sweet spot" region. For bats having a significantly different shape profile, especially in the barrel region, the vibration node locations in the barrel can differ by several centimeters. The primary focus will be on wood baseball bats, but if time permits additional data from multi-piece and softball bats will be included.

Session 3aPA

Physical Acoustics and Computational Acoustics: Software for Infrasound Research and Applications

Roger M. Waxler, Cochair
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Michelle E. Swearingen, Cochair
Construction Engineering Research Laboratory, US Army ERDC, P.O. Box 9005, Champaign, IL 61826

Chair's Introduction—8:30

Invited Papers

8:35

3aPA1. NCPAprop: An open-source infrasound modeling package. Claus Hetzer (National Ctr. for Physical Acoust., Univ. of Mississipppi, 145 Hill Dr., University, MS 38677, claus@olemiss.edu) and Roger M. Waxler (National Ctr. for Physical Acoust., Univ. of Mississipppi, University, MS)

Simulation of infrasound propagation in a collaborative research environment requires tested and validated numerical models that are freely available, readable enough to parse and understand, and efficient enough to run on a standard workstation. The NCPAprop package was designed to facilitate this kind of research by directly addressing all three of these criteria. Now in its second release version, and with the third release under active development, NCPAprop is an open-source command-line package written in C^{++} and is widely used in the international infrasound community. It provides two normal-mode models and a parabolic equation model that supports range-dependent atmospheric specifications and the inclusion of orography. The capabilities of the package will be discussed, as well as the status and direction of current development.

8:55

3aPA2. Application of open-source infrasound software to surface explosions—Propagation analyses. Philip S. Blom (Los Alamos National Lab., P.O. Box 1663, M/S F665, Los Alamos, NM 87545, pblom@lanl.gov), Jordan W. Bishop (Los Alamos, NM), and Jeremy Webster (Los Alamos National Lab., Los Alamos, NM)

Seismoacoustic R&D at Los Alamos National Laboratory (LANL) has produced a suite of open-source tools including infraGA/GeoAc and stochprop which enable propagation simulation via ray tracing and propagation uncertainty quantification, respectively. Usage of these tools will be presented to elucidate the regional infrasonic propagation of signals produced by a surface explosion that occurred at the Camp Minden Army Ammunition Plant in northern Louisiana in 2012 as well as several other events. The Camp Minden explosion occurred while the USArray network was deployed across the southeastern United States and provides a useful data set for studying regional propagation as it includes a relatively dense network of observing stations. Ensemble analysis using empirical orthogonal function will be demonstrated using the Camp Minden and other events to quantify the impact of atmospheric uncertainty on infrasonic propagation.

Contributed Papers

9:15

3aPA3. Development of a 3-D PE for long range infrasound propagation. Roger M. Waxler (National Ctr. for Physical Acoust., Univ. of Mississipppi, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu) and Claus Hetzer (National Ctr. for Physical Acoust., Univ. of Mississipppi, University, MS 38677)

For infrasound propagation most full wave models are restricted to a range-altitude plane and do not model out of plane propagation. This planar approximation is sufficient for many applications. Exceptions include the azimuth deviation produced by cross winds and out of plane ground reflections produced by nontrivial orography. The latter is particularly important because the shadowing effect of mountain ranges is significantly overestimated by the planar approximation. We have been working on extending the capabilities of the NCPAprop's ePape planar parabolic equation (PE) model to 3-D propagation. ePape solves the one way radial wave equation in cylindrical coordinates using a Pade approximant to represent the exponential of the range step operator. The extension to out-of-plane propagation retains the ePape Padé approximation for the range steps, but adds a stable Runge–Kutta algorithm for the azimuthal components. The basic algorithms and approximations as well as the inherent difficulties will be discussed and we will report on progress to date.

9:30

3aPA4. Revisiting normal-mode methods for acoustic–gravity wave propagation. Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrechtseweg 297, Utrecht 3731GA, Netherlands, jelle.assink@knmi.nl) and Roger M. Waxler (Univ. of MS, University, Mississipppi 38677)

Although theories and numerical methods for simulating acoustic–gravity waves using normal modes date back to the 1960s, practical and accessible implementations remain scarce. Recent large-scale geophysical events, such as the 2022 Hunga eruption, that generate these waves underscore the need for accurate and robust modeling tools. This presentation presents a recent extension of the normal-mode framework in the NCPAprop package, expanding its capabilities for acoustic–gravity wave propagation.

9:45–10:05 Break

Invited Papers

10:05

3aPA5. Infrapy, a Python-based infrasound analysis suite. Jeremy Webster (Earth & Environ. Sci., Los Alamos National Lab., LANL, MS F665, Bikini Attoll Rd., Los Alamos, NM 87544, jwebster@lanl.gov) and Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., Los Alamos, NM 87544)

Infrapy is a Python-based suite of tools designed for analysis of infrasound signals. It provides a command line interface, a fully functional GUI (Infraview) for fast and convenient event analysis, and a library that can be used to develop applications for custom studies. It includes an array beamformer with location and association capabilities, along with a single-sensor detector that utilizes spectral based detections. Currently, it can process local waveform files (SAC and miniseed format) and can import data from FDSN servers such as IRIS, or from SQL databases that utilize a CSS or KBCore schema. In 2020, LANL open sourced Infrapy to provide it to the broader community. In this presentation, we will review Infrapy's development and discuss its primary capabilities, as well as its recently added and upcoming features.

10:25

3aPA6. Developing tools for generating realistic synthetic infrasound array recordings. David N. Green (AWE Blacknest, AWE Blacknest, Brimpton RG7 4RS, United Kingdom, dgreen@blacknest.gov.uk)

Generating realistic synthetic infrasound array recordings is useful for tasks including detector performance testing, training inexperienced analysts, and modelling the impact of faulty datastreams. Although significant advances have been made in physics-based numerical modelling of realistic waveforms over regional distances (a few hundred kilometres), this task remains challenging for longer distance propagation paths of interest to nuclear test monitoring. Here, we describe the development of a tool for synthetic array recording generation, producing waveforms that exhibit characteristics consistent with measured signal features. This initial work is restricted to long ($>$1000 km) distance propagation scenarios, pertinent to simulation of recordings at International Monitoring System arrays that are being developed as part of the Comprehensive Nuclear-Test-Ban Treaty verification measures. We focus on the need for (1) a modular design to facilitate updating of empirical constraints, (2) a framework that promotes repeatable result generation, and (3) flexibility to allow users to test scenarios of interest.

10:45

3aPA7. Finding sparse representations of infrasound transients embedded in strong wind noise and clutter. Garth Frazier (NCPA, Univ. of Mississipppi, P.O. Box 1848, Oxford, MS 38677, frazier@olemiss.edu)

Detection of infrasound transients embedded in strong wind noise and clutter is a common difficulty when processing data collected with an array of infrasound sensors. High levels of wind noise make detection and estimation difficult because the option of improving the signal-to-noise ratio with temporal averaging is not possible for a transient signal. Strong clutter can require that a processing technique specifically designed to find multiple signals be used. Not knowing the spectral content and duration of the transient signals in advance creates another challenge. Several approaches can be used to address these issues. Firstly, estimating the wind noise spectrum and the clutter signals (if stationary) in advance of the arrival of a transient is valuable especially for initial detection and improves the statistical efficiency of the waveform estimator. Secondly, estimating the properties of the clutter can help separate the clutter signal from the transients. And, thirdly, using an appropriate library of signal representations can be used to address the uncertainties concerning the signal. This presentation addresses how to achieve all these capabilities within a single processor and focuses especially on an overcomplete library of signal representations that is suitable for the finding sparse representations of stationary clutter and transients of unknown spectral content and duration.

Contributed Paper

11:05

3aPA8. Nonlinear propagation of thermospheric infrasound through a stratified atmosphere. Sami Azzaz (Institut Jean le Rond d'Alembert, Sorbonne Univ., 152 Ave., Émile Cossonneau, Noisy-le-Grand 93160, France, sami.azzaz@sorbonne-universite.fr), François Coulouvrat (Institut Jean le Rond d'Alembert, CNRS, Paris, France), Régis Marchiano (Institut Jean le Rond d'Alembert, Sorbonne Univ., Paris, France), and Olaf Gainville (CEA, DAM, DIF, Bruyères-le-Châtel, France)

Infrasound is an excellent tool for characterizing either natural (meteors, volcanoes, and earthquakes) or anthropogenic (explosions, rockets, and sonic booms) sources, as well as for probing the atmosphere. However, infrasound is sensitive to the strong variability of atmospheric temperature, wind, and density. To use infrasound as a diagnostic tool, a thorough understanding of the propagation mechanisms and the ability to predict propagation in realistic atmospheres are necessary. Conventional methods, such as ray tracing or parabolic equation methods have limitations and direct numerical simulations of the full compressible Navier–Stokes equations are computationally expensive for long-distance propagation. We present an alternative approach based on solving a nonlinear scalar wave equation that incorporates atmospheric effects: diffraction, refraction from celerity gradient induced by temperature, decreasing overpressure amplitude, and nonlinearities from strong density variations with altitude and absorption. The algorithm, named FLHOWARD3D and developed for acoustic shock waves, has been adapted to model infrasound propagation up to the thermosphere by renormalizing the acoustic pressure with the square root of density. Results of propagation through stratified atmosphere will be shown and compared to conventional methods, pointing out the importance of nonlinear effects. [Work done in the framework of the LETMA joint laboratory CEA/CNRS/ECL/Sorbonne Université.]

Session 3aPP

Psychological and Physiological Acoustics: Blossoming Science and Cultivating Researchers in Psychological and Physiological Acoustics

Gregory M. Ellis, Cochair
Audiology and Speech Pathology, Walter Reed National Medical Military Center, 4494 Palmer Rd. N, Bethesda, MD 20814

Varsha H. Rallapalli, Cochair
Communication Sciences & Disorders, University of South Florida, 4202 Fowler Ave., PCD 1017, Tampa, FL 33620

Chair's Introduction—8:00

Invited Papers

8:05

3aPP1. Swept-modulation-depth stimulus for efficient neurometric physiological analyses of modulation detection. Afagh Farhadi (Purdue Univ., West Lafayette, IN 47906, afarhadi@purdue.edu), Arezou Rokhforouz, Andrew Olliff (Purdue Univ., West Lafayette, IN), Hari Bharadwaj (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA), and Michael G. Heinz (Purdue Univ., West Lafayette, IN)

Amplitude modulation (AM) conveys critical temporal cues for speech perception; prior work shows a close link between modulation detection and speech intelligibility. Slow modulation rates (e.g., as in speech envelopes) are critical but are difficult to measure because longer stimulus durations are needed to include sufficient modulation periods. Thus, current neurometric approaches that estimate modulation detection by repeating measurements across several modulation depths are very time intensive. This challenge is particularly pertinent for understanding neural mechanisms underlying modulation coding in hearing-loss conditions, which are directly investigated through single auditory-nerve (AN) fiber recordings in preclinical animal models of sensorineural hearing loss. The limited time available to record from single units is constraining. Similar challenges occur in electrophysiological measures such as envelope following responses (EFRs), where low signal-to-noise ratios require more repetitions and longer acquisition times. To address this challenge, we developed a swept-modulation-depth stimulus, in which modulation depth changes continuously over time. Using AN spike-train data and EFRs in the chinchilla model, we evaluated accuracy and efficiency of this method by comparing thresholds from swept

and discrete paradigms. Preliminary results demonstrate accurate and efficient estimation of AM detection thresholds using swept stimuli, improving the practicality of physiological studies of modulation coding.

8:25

3aPP2. Functional auditory deficits following occupational noise exposure or blast-induced traumatic brain injury (bTBI). Fernando Aguilera de Alba (Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47906, aguilerl@purdue.edu) and Michael G. Heinz (Purdue Univ., West Lafayette, IN 47906)

Service members are often exposed to harmful acoustic environments. Understanding how different acoustic exposures affect auditory processing is critical. We investigated peripheral and subcortical auditory deficits following continuous noise exposure or a single blast exposure to induce mild-to-moderate traumatic brain injury (TBI). Chinchillas were exposed to noise simulating aircraft-carrier for 1, 2, or 4 consecutive weeks (40 h/week at 87.5 dBA) or one single (1) blast at 150 kPa with bilateral hearing protection (earplugs). We employed a multi-metric auditory framework including tympanometry, wide-band middle-ear muscle reflex (WB-MEMR), otoacoustic emissions (DPOAE, SFOAE, and TEOAE), auditory brainstem response (ABR), and envelope following response (EFR) to amplitude-modulated stimuli. *Noise*: Reduced DPOAEs by 10–15 dB across all frequencies, SFOAEs decreased by 5–20 dB at mid-to-high frequencies, and WB-MEMR thresholds increased as early as 1 week. ABRs revealed mild-to-moderate threshold shifts across all frequencies (10–30 dB) while EFRs showed degraded temporal encoding. *Blast:* Subtle effects were observed for DPOAEs, SFOAEs, and WB-MEMR. Although hearing thresholds were mostly unaffected, temporal encoding was degraded as early as 3 days post-exposure followed by hypersensitivity. Both exposures showed mixed TEOAE effects across frequencies. Our study highlights the importance of implementing multi-metric auditory diagnostics.

8:45

3aPP3. Phoneme patterns and artificial intelligence reliability in infant-directed speech transcription. Sanoe Rapozo (Trinity College of Arts & Sci., Duke Univ., 4813 Beadle St., Las Vegas, NV 89122, sanoe.rapozo@duke.edu) and Tian C. Zhao (Dept. of Speech and Hearing Sciences/Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA 98195)

Adults modify their speech when addressing infants, a speech register known as infant-directed speech (IDS). Past research has focused on the acoustic differences between IDS and ADS (adult-directed speech), but differences in phonetic distribution have been minimally investigated, especially in naturalistic home environments. Examining these differences can clarify how the sound patterns infants hear shape phoneme perception and category formation, informing theories of speech processing and language acquisition. This study aimed to compare phoneme distributions of ADS and IDS in naturalistic home environments. We analyzed 10-s snippets of LENA recordings previously labeled as containing IDS or ADS through a crowdsourced approach. The audio snippets were transcribed using both an AI-based tool (WhisperX) and human transcribers to obtain an AI–human error rate. Both transcriptions were processed by the CMU Pronouncing Dictionary to extract ARPAbet and IPA phoneme distributions. Results indicated that IDS contained more AA (ɑ), G (g), and UW (u), while ADS contained more AY (aɪ), AH (ʌ), and S (s), among others. Phonetic distributions were also largely consistent across human and AI transcriptions. These findings highlight key phonemes potentially central to infant language acquisition, while also reducing time-intensive manual transcription by supporting the use of AI in phonetic research.

9:05

3aPP4. Associations between semantic context in speech recognition, spatial release from masking, and executive functioning in children using bilateral cochlear implants and typical hearing children. Nimesha D. Dantanarayana (Waisman Ctr., Univ. of Wisconsin-Madison, Waisman Ctr., 1500 Highland Ave., Madison, WI 53705, ndantanaraya@wisc.edu), Sara M. Misurelli (Otolaryngol.—Head and Neck Surgery, Univ. of Wisconsin-Madison, Madison, WI 53705), Shelly P. Godar (Madison, WI), and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI 53705)

Speech materials that assess speech understanding in quiet and noise in children with and without hearing loss (HL) are typically highly predictable in their semantic context. Because natural environments contain unpredictable speech information, ecological validity is stronger with materials that manipulate semantic context. This study investigated the effects of semantic context on speech recognition and spatial release from masking in children (ages 9–15 years) with bilateral cochlear implants (BiCIs) and age-matched typical hearing (TH) children, and examined associations with executive functioning. Sentences were drawn from a corpus of semantically coherent or anomalous target sentences. Three conditions were tested: quiet, target and masker co-located, and target and masker spatially separated; speech interferers were presented at varying signal-to-noise ratios. TH children were also tested with vocoded stimuli simulating aspects of CI listening. Results revealed that children with BiCIs and TH may rely on both semantic and spatial cues differently to recognize degraded speech. Whereas speech recognition with clear speech was associated with cognitive flexibility and attention inhibition, auditory attention and inhibition were associated with understanding vocoded speech. Executive functioning of children using BiCIs may also be associated with their speech recognition in noise. [Work funded by NIH-NIDCD Grant R01DC020355 to R.Y. Litovsky.]

9:25

3aPP5. Sex-dependent variability in binaural processing across the lifespan. Anhelina Bilokon (Dept. of Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, abilokon@umd.edu)

Sex differences in auditory function remain poorly understood, even though the NIH/NIDCD requires inclusion of sex as a biological variable. Most age-related hearing studies collapse across sex or focus on mean performance rather than variability, even though variance may reveal hormonal or biological influences. Men typically experience earlier high-frequency hearing loss, while women retain better thresholds until menopause. Electrophysiologic studies also show that women's auditory function fluctuates with hormonal changes across the menstrual cycle. Understanding these sources of variability is critical for explaining inconsistencies in aging data and identifying biomarkers of auditory decline. The goal of this talk is twofold: to present evidence for sex-dependent auditory variability and to outline biologically informed guidelines for designing sex-balanced studies that improve hearing research. As part of a retrospective investigation, I reanalyzed age-matched, normal-hearing male ($n = 22$) and female ($n = 24$) listeners across auditory measures:

lateralization, discrimination, and localization. Among older adults (>45 years; 10F, 8M), females exhibited greater variance in interaural time-difference lateralization ($p = 0.029$), suggesting increased temporal-processing variability with age. In contrast, males showed greater variability across most tasks overall. These findings highlight sex-specific contributions to auditory aging and emphasize the need for sex-specific models of hearing function.

9:45

3aPP6. Spatial perception with over-the-counter hearing aids. Pinar Ertürk (Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, perturk@bu.edu) and Virginia Best (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

While hearing aids are critical for providing audibility and supporting speech intelligibility, they can have negative effects on spatial perception. Currently, the range of available styles of hearing aids is rapidly expanding with the introduction of over-the-counter devices (OTCs), but little is known about their effects on spatial perception. Importantly, these devices are aimed at listeners with relatively mild hearing losses, who may be especially sensitive to spatial distortions. Our broad goal is to comprehensively characterize spatial perception with different OTCs, considering azimuth identification, front-back discrimination, and sound externalization. In an initial study, twelve adult listeners with normal hearing were bilaterally fitted with OTCs representing different form factors: behind-the-ear (BTE; Lexie B2 Plus with open and closed domes), completely-in-canal (CIC; Eargo SE), and earbuds (Apple Pro 2). Each listener completed the different spatial tasks while wearing each set of devices, with unaided performance as a reference. While the mean results revealed disruptive effects of each OTC for all tasks, the size of the effect depended strongly on the device and on the individual listener. A follow-up study is expanding the set of OTCs, including listeners with mild-to-moderate hearing loss, and supplementing the behavioral results with acoustical analyses of ear-canal signals.

WEDNESDAY MORNING, 13 MAY 2026 FRANKLIN HALL 11, 8:00 A.M. TO 10:25 A.M.

3a WED. AM

Session 3aSP

Signal Processing in Acoustics and Acoustical Oceanography: Bayesian Signal Processing II

Ning Xiang, Cochair
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Zoi-Heleni Michalopoulou, Cochair
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Ali Djafari, Cochair
Renminxi Road 1818, Shao Xing, China

Invited Papers

8:00

3aSP1. Beyond assumptions: Discriminating locally and extended reacting acoustic materials via Bayesian inference. Eric Brandão (Acoust. Eng. Program, Federal Univ. of Santa Maria (UFSM), Santa Naria, RS, Brazil), Martin Eser (Chair of Vibroacoustics of Vehicles and Machines, Tech. Univ. of Munich, Munich, Germany), and Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu)

The accurate *in situ* or free-field characterization of acoustic materials is highly dependent on prior assumptions regarding both the sound field and the material itself. This study presents a Bayesian framework for estimating the frequency- and angle-dependent absorption properties of materials using microphone array measurements. The proposed approach explicitly incorporates a prediction model while enabling robust parameter estimation from measured sound fields, based on a formulation rooted in the Sommerfeld integral. Within this framework, material properties are inferred within the Bayesian framework using nested sampling, which simultaneously performs parameter estimation and model selection. This facilitates a direct comparison between competing models of material behavior, moving beyond reliance on *a priori* assumptions. The method yields estimates of frequency- and angle-dependent parameters, including the characteristic propagation coefficient, surface impedance, and absorption coefficient. Furthermore, the Bayesian

framework quantifies the uncertainty of inferred parameters and naturally penalizes model over-complexity. Focusing on numerical simulations, this paper demonstrates model comparison between local and extended reacting materials. The results show that the proposed framework can effectively discriminate between these prediction models based on Bayesian evidence, while providing reliable parameter estimates under realistic measurement conditions.

8:20

3aSP2. Phase unwrapping in experimental characterization of porous media using Bayesian inference. Ziqi Chen (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Porous media are commonly characterized by their characteristic impedance and propagation coefficients. This work experimentally characterizes porous media in an impedance tube using a three-microphone method [Salissou and Panneton, *J. Acoust. Soc. Am.* **128**, 2868–2876 (2010)]. Extending the valid upper frequency limit, fundamentally constrained by the tube diameter, averages of multiple microphones at the same plane across the tube cross-section effectively extend the usable frequency range of impedance tube measurements. However, over a broad, extended frequency range, discontinuities arise in the estimated characteristic impedance and propagation coefficients due to phase wrapping. In this work, Bayesian inference is applied sequentially to resolve the wrapped phase. The prior knowledge is updated by the posterior at each data point (frequency bin). The proposed approach enables measurements over nearly twice the frequency range of the conventional method while effectively solving the phase-wrapping problem.

Contributed Papers

8:40

3aSP3. Causal adaptive time-frequency analysis via Bayesian universal switching and time-domain aliasing. Manan Mittal (Elec. and Comput. Eng., Stony Brook Univ., 100 Nicolls Rd., Stony Brook, NY 11794, manan.mittal@stonybrook.edu), Kanad Sarkar (Univ. of Illinois at Urbana Champaign, Urbana, IL), Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, Chicago, IL), and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

The short-time Fourier transform (STFT) is fundamentally limited by the Gabor uncertainty principle, requiring a global compromise between time and frequency resolution. Fixed-window analyses often obscure transient events with long windows or blur spectral features with short windows. This work presents the "Universal Switching Spectrogram," a probabilistic framework that dynamically adapts the effective analysis window length. The algorithm operates strictly causally, maintaining a distribution of "expert" states where each state represents a different window start time relative to the current sample. To enable the coherent blending of varying window lengths into a unified spectral representation, we utilize a time-domain aliasing (folding) technique that maps arbitrary window lengths onto a fixed-size FFT basis. A composite cost function—minimizing both spectral and temporal entropy while maximizing Gaussian likelihood—updates the state probabilities via a recursive Bayesian mechanism. With a computational complexity of $O(KN \log N)$ per sample (where K is the number of active experts and N is the FFT size), the method is computationally efficient and suitable for real-time operation. We demonstrate the algorithm's efficacy on synthetic test suites and real-world audio, showing that the method yields a higher resolution time–frequency spectrum.

8:55

3aSP4. Aeroacoustic source separation with statistical inversion in the time domain. Mitchell J. Swann (Penn State Univ., P.O. Box 30, State College, PA 16804, mjs8688@psu.edu), Samuel J. Grauer, Adam S. Nickels, Michael H. Krane, and Jeff Harris (Penn State Univ., State College, PA)

This work demonstrates the use of statistical inversion for source separation of time domain microphone array signals observing an aeroacoustic source. Here, the aeroacoustic source under study is an individual vortex ring convecting past the edge of a semi-infinite half-plane in an anechoic chamber. The vortex/edge (V/E) interaction is observed with a circular 12-microphone array centered on the edge of the half-plane where the V/E source is most significant. Motion of the vortex ring is captured with synchronous high-speed Schlieren. Inherent to the production of the individual vortex rings a spherically spreading pressure wave is produced which pollutes the measurement of the V/E interaction. Both of the sources present in the experiment are impulsive and non-stationary, for which a time domain approach to source separation is advantageous. A simplistic model approximating the experiment and the sources present is developed. Statistical inversion of the approximating model enables the estimation of the underlying source waveform time series of the two sources, in addition to their uncertainties. Estimated V/E source characteristics are used for comparison between different statistical inversion methods as well as prior experimental results.

9:10–9:25 Break

9:25

3aSP5. Real-world visual–inertial and acoustic simultaneous localization and mapping for underwater remotely operated vehicle navigation. Hala Abualsaud (ECE, Univ. of California San Diego, 10645 Calle Mar de Mariposa, Apt 6205, San Diego, CA 92130, habualsa@ucsd.edu), Ying-Tsong Lin (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), and Peter Gerstoft (Scripps Inst. of Oceanogr., UCSD, San Diego, CA 92130)

This work presents real-world experimental results from a Bayesian underwater simultaneous localization and mapping (SLAM) system deployed on a remotely operated vehicle (ROV) and evaluated in both controlled pool experiments and ocean settings. The SLAM system uses a feature-based visual SLAM approach that extracts repeatable keypoints and descriptors from a forward-facing camera and matches them across frames. These tracked visual features are fused with inertial measurements in a Maximum a Posteriori optimization to jointly estimate the ROV trajectory (pose over time) and a sparse visual landmark map. Camera calibration is performed to estimate the intrinsic parameters and lens distortion required for accurate feature projection. Acoustic sensing is integrated through an altimeter and sonar measurements, providing range-to-seafloor and to landmarks, enhancing vertical motion and overall 3-D trajectory estimation in visually degraded underwater conditions. System performance in the ocean setting is evaluated using fixed, moored reference points with known GPS locations to provide ground-truth trajectory validation.

9:40

3aSP6. Range estimation and uncertainty quantification of single-hydrophone underwater acoustic data using invertible neural networks. Mark Goldwater (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS #11, Woods Hole, MA 02543, mgoldwater@whoi.edu), Daniel P. Zitterbart (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Julien Bonnel (Scripps Inst. of Oceanogr., San DIego, CA)

Low-frequency signals in shallow-water environments can be decomposed into dispersive modal components with time-frequency positions that are strongly range-dependent. As a result, these signals are suitable for range-based source localization. However, robust inversion methods typically rely on Markov chain Monte Carlo algorithms which require expensive forward simulations of the propagation physics to explore high-dimensional

source and environment parameter spaces. Fortunately, trained machine-learning-based inversion approaches have yielded accurate acoustic source and environment inversion results at unprecedented speeds. Despite this, typical machine learning approaches are not naturally able to quantify uncertainty without model calibration or some assumed parameterization of the posterior. At the core of our work is the application of an invertible neural network (INN). The network learns a one-to-one mapping between the joint posterior distribution of relevant source parameters, such as range, and a known tractable distribution. Thus, drawing samples from the known distribution and passing them in reverse through the INN allows for the efficient sampling of the source parameter posterior which can be used in downstream localization algorithms. We demonstrate our method on both simulated and experimental acoustic data where modal dispersion is present. [Work supported by ONR and NDSEG.]

9:55

3aSP7. Wind-driven effects in shallow water acoustics: A data assimilation approach. Grant C. Eastland (US Navy, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell Rd., Keyport, WA 98345, grant.c.eastland.civ@us.navy.mil) and Justin Diamond (Mech. Eng., Univ. of Washington, Seattle, WA)

Wind-driven effects significantly impact acoustic propagation and reverberation in shallow water bays, creating a complex and dynamic environment that challenges the accuracy of traditional predictive models. The focus of this presentation aims to demonstrate potential improvement to the fidelity of the Navy's standard Parabolic Equation (PE) model by incorporating data assimilation techniques. This approach will integrate real-time, *in-situ* acoustic measurements with open-source meteorological data to provide a more accurate and comprehensive representation of the acoustic environment. Wind introduces variability through several mechanisms, including changes to surface roughness, bubble creation, and variations to the sound speed profile, all of which affect how sound travels and scatters in water. Traditional models often struggle to account for this dynamic variability and utilizing data assimilation of both acoustic and meteorological data would allow for the PE model to self-correct.

10:10

3aSP8. Physics-guided machine learning reconstruction of modal structure in the seabed characterization experiments. Sebastian Rojas-Ortega (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Evans Hall, Newark, DE 19716, ortega@udel.edu), Mohsen Badiey, and Christian D. Escobar-Amado (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

Accurate extraction of normal modes from underwater acoustic data is relevant for seabed characterization and geoacoustic analysis. In this context, efficiently predicting these depth-dependent mode structures is critical for understanding the acoustic field. Standard signal processing techniques for extracting these features can be computationally intensive or sensitive to noise, limiting their robustness in practical applications. This work introduces a physics-guided machine learning framework designed to predict depth-dependent mode shapes in a range-independent waveguide. The model is developed using a synthetic dataset generated with a normal mode model, parameterized with experimentally measured sound speed profiles from the Seabed Characterization Experiments (SBCEX) in 2017 and 2022. By learning the physical relationships defined by the waveguide, the framework provides a data-driven approach to characterize modal structures. The motivation is to enable modal structure estimation from sparse vertical sampling typical of SBCEX (8 hydrophones), where conventional extraction is unstable. We evaluate the model's performance in the low-frequency regime, demonstrating its potential as a robust, automated methodology for modal analysis in realistic ocean environments, where the model achieved correlation above 0.9 (averaged across modes 1–5) for frequencies below 200 Hz. [Work supported by ONR code 322.]

Session 3aUW

Underwater Acoustics: Acoustic Scattering

Andone C. Lavery, Cochair
Oregon State Univ., Corvallis, OR

Lucy Brock, Cochair
School of Oceanography, University of Washington, 1431 NE Boat St., Seattle, WA 98195

Contributed Papers

8:30

3aUW1. Analysis of close-range reflection and scattering from a seamount flank. Johnathan J. Todd (Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, jjt6090@psu.edu), William Hodgkiss (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Daniel C. Brown (State College, PA), and Chad M. Smith (The Penn State Univ., State College, PA)

During the 2024 New England Seamounts Acoustics (NESMA) experiment, bistatic measurements were made of close-range acoustic reflection and scattering from a seamount flank. Measurements were made with an omnidirectional low- to mid-frequency source operated by the Scripps Institution of Oceanography, and Penn State's towed receiver array, the Three Octave Research Array (THORA), at a depth near 150 m. The source–receiver geometry was designed such that the first acoustic arrival interacted with the same seamount flank region at differing grazing angles from 20° to 72°, with the source broadside to the THORA at their closest point of approach. Acoustic data were beamformed to improve signal-to-noise ratio, reduce own-ship noise, and isolate in-plane arrivals. A ray model was used to interpret time of acoustic arrivals, calculate attenuation due to water column absorption and spreading, and estimate average levels for reflection coefficient and incoherent scattering. This talk will discuss the details of these unique measurements and investigate a frequency and grazing angle dependence.

8:45

3aUW2. What Is the target strength of an infinite surface? Matthew S. Byrne (Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, matthew.s.byrne4.civ@us.navy.mil)

Target strength, or the acoustic scattering cross section, is an important parameter in active sonar performance models. It describes the ratio of sound scattered from a source to a receiver as a function of the frequency of the active signal and the orientation of the target with respect to the source–receiver pair. When assessing the target strength of an unknown target, it can be informative to consider theoretical solutions of scattering from canonical objects, such as a rigid sphere or cylinder. Scattering solutions to these problems can be useful for setting expectations for the target strength values of more complex, multi-featured targets. One common canonical object, as notably described by [1], is that of an infinite planar surface. However, the expression provided by [1] for the target strength of an infinite planar surface is not derived, explained, or provided with reference in [1], or any other publication known to this author. Therefore, the derivation has been developed and explained using a solution of the Helmholtz–Kirchoff integral. Results of this expression are compared to alternative numerical and analytical solutions. [1] Urick, R. J. *Principles of Underwater Sound*, 3rd ed. (McGraw Hill, 1983).

9:00

3aUW3. Seabed echo backscattering strength from multibeam echosounder data. Henry Manik (Dept. of Marine Sci. and Technol., Faculty of Fisheries and Marine Sci. IPB Univ., Kampus IPB Dramaga, Bogor Indonesia, Bogor 16680, Indonesia, henrymanik@apps.ipb.ac.id)

Seabed acoustic backscatter strength (BS) represents the seafloor response to incident acoustic waves and is widely used for mapping geological resources and benthic habitats. This response depends on seafloor properties such as roughness and acoustic impedance and can be derived from multibeam echosounder (MBES) measurements through the sonar equation. Although MBES is a key tool for marine mapping, accurate BS estimation requires compensation of several sonar-related parameters. This study proposes a machine learning-based method to estimate seabed backscatter using MBES echo time-series data. The results highlight the importance of sediment mapping and demonstrate the potential of combining acoustic methods with machine learning for seafloor classification.

9:15

3aUW4. The use of a Born-like series in multiple scattering. Ahmad T. Abawi (HLS Res., 12625 High Bluff Dr., Ste. 211, San Diego, CA 92130, abawi@hlsresearch.com)

Much of our understanding of scattering from targets originates from radar studies, which primarily focus on scattering in free space. However, in oceanic environments, acoustic scattering is significantly more complex due to the presence of boundaries, interfaces, and volumetric variations in the medium. Although the ocean can often be approximated by two half-spaces, this simplification alone is insufficient, and further approximations are frequently required to address practical scenarios. In this presentation, we provide a numerically exact solution for scattering from an elastic target near a pressure-release boundary, serving as a benchmark solution. We then present the widely used single-scatter approximation to reduce the complexity of the problem, examining its impact on the solution, its physical implications, and potential methods to regain lost accuracy. Subsequently, we employ successive corrections to the single-scatter approximation using a Born series expansion and demonstrate that convergence is achieved after only a few terms. In doing so, we derive a closed-form expression for the series, applicable for any number of terms. The results are validated through comparisons with experimental data and exact numerical solutions.

9:30–9:45 Break

9:45

3aUW5. Comparisons of approximate scattering models with numerical integral equation methods. Derek Olson (Oceanogr., Naval Postgrad. School, Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943, dolson@nps.edu)

Approximate scattering models for rough surface interaction with acoustic waves serve as the foundation for remote sensing and performance prediction for target detection systems. The three main approximations used for real-world application are the small-slope approximation (SSA), the small-roughness perturbation method (SRPM), and the Kirchhoff, or tangent-plane approximation (KA). These models have undergone study for their region of validity for simple Dirichlet boundary conditions, and a single-scale roughness spectrum, but little work to date has performed validity studies for the fluid–fluid boundary condition, or realistic roughness power spectrum models, such as a power-law form. We present such a comparison for fluid-fluid boundary conditions approximating sand (which has a critical angle) and mud (which does not have a critical angle) under a power law spectrum with varying spectral strength, spectral exponent, and outer scale. The power-law inner scale was kept fixed. The benchmark solution was provided by a coupled boundary integral equation on each side of the fluid interface. The region of validity for each of these boundary conditions is presented in terms of the dimensionless root mean square height and slope.

10:00

3aUW6. Exploiting the diffraction gap: An auto-guidance solution for supercavitating torpedoes. Henry Shen (Dept. of Civil, Environ., and Ocean Eng., Stevens Inst. of Technol., 1 Castle Point Terrace, Hoboken, NJ 07030, hshen14@stevens.edu)

Supercavitating torpedoes (SCTs) achieve high speeds exceeding 200 knots, but the turbulent ventilated cavity creates a self-noise environment that typically masks conventional active sonar returns. This research quantifies the operational envelope of a diffraction gap—a spectral acceptance window created by the interaction of nose-cone baffling and the extreme Doppler shifts inherent to high-velocity intercepts. To evaluate this tracking architecture, a high-fidelity stochastic simulation was developed integrating Thorp's volumetric absorption (4.0 dB/km) and empirical supercavity noise profiles. The study explicitly characterizes the SCT's burnthrough range against a reactive submarine target executing a maximal-aspect evasion maneuver. Results indicate that the high closing velocity generates a two-way Doppler shift of over 2.5 kHz, effectively shifting the target return out of the primary self-noise spectral band. By exploiting this shift, the system achieves a processing gain sufficient to secure a valid track lock at a range of 1.2 km, despite an initial signal-to-noise Ratio (SNR) well below −30 dB, validating this technique as a viable mechanism for terminal homing in self-noise-limited regimes.

10:15

3aUW7. Modeling and experimental evidence for the frequency dependence of sub-bottom reflections by an elastic layered seafloor. Matthew A. Milone (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8622 Kennel Way, San Diego, CA 92093, mamilone@ucsd.edu), William Hodgkiss, and Ying-Tsong Lin (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Seafloor composition and the resulting parameter values governing compressional and shear wave propagation significantly impact acoustic transmission, both within the seafloor itself and in the water column above. Numerical simulations frequently ignore shear waves for computational ease, instead modeling ocean bottom layers as higher-density fluid regions. A significant limitation of this approach is the loss of coupling effects between compressional and shear waves, through which the exchange of energy may increase or decrease acoustic pressure. Data collected in 2024 for the New England Seamount Acoustics (NESMA) experiment shows a striking example, where sub-bottom returns of an emitted pulse decrease in strength and even disappear completely within some frequency bands. A broadband wavenumber integration model which more realistically treats the seafloor as an elastic medium produces a comparable phenomenon in which specific shear wave speeds reduce and effectively eliminate expected sub-bottom returns. This work seeks to establish a connection between acoustic frequency and seafloor shear wave properties to quantify conditions under which sub-bottom returns may vanish. Comparison with a fluid seafloor wavenumber integration model shows the importance of including shear wave characteristics for an accurate representation of experimentally measured data.

10:30

3aUW8. Simscape model and validation of acoustic transducer array. Colin Flavin (Acoust., Penn State Univ., 300 Hwy 361, Crane, IN 47522, clf5671@psu.edu)

This work seeks to evaluate Simscape as an alternative way to explore the response of an acoustic transducer array. To begin evaluating its capabilities, ceramic stack assemblies with properties similar to those utilized in the F27 general purpose directional transducer are created and tested in real life, then compared to their models in Simscape. The stack assemblies consist of a lead metaniobate ceramic component and a tungsten backing plate and are tested both in-air and in-water as well as both individually and in array configuration. Simsape is a powerful and flexible, yet easy to understand environment that can model a variety of acoustic devices, from simple spring-mass assemblies to moving-coil loudspeaker. Its use in transducer array modeling is less explored, and its accuracy therewith could be a great benefit to those seeking an alternative to more expensive and time-consuming analysis methods.

10:45

3aUW9. Shallow-water underwater acoustic estimation of the number and locations of reflective surfaces via deep learning. Toros Arikan (Elec. Eng., Univ. of Notre Dame, 171 Fitzpatrick Hall, Notre Dame, IN 46556, tarikan@nd.edu)

The single-snapshot acoustic estimation of the locations of reflective boundaries in environments is an ill-posed open problem, with the state-of-the-art methods in the literature often implicitly assuming high signal-to-noise ratios (SNRs) for robust performance. Such methods generally assume the number of boundaries in the environment to also be known, and further require the combinatorial and difficult echo labeling problem to be solved as well. We have previously proposed a deep learning method (Neuro-COTANS) that outperformed the state-of-the-art, and also avoided having to solve the echo labeling problem and was therefore more robust. However, this method still carried the limitation of requiring the number of boundaries to be known, and also needed prior knowledge of roughly where the reflective boundaries were in the environment. We propose a more advanced method here that does not require either prior knowledge of the number of boundaries or their locations. By utilizing more advanced algorithms and neural network architectures, we thereby offer a single-snapshot mapping method with unprecedented multi-parameter estimation capabilities. This is an important step toward plug-and-play two-dimensional environment estimation methods that are applicable to any acoustic environment.

Session 3pAA

Architectural Acoustics and Noise: Interesting Measurement Techniques and Results Part II

Joseph Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, STE 209, Woodbridge, NJ 07095

Michael Raley, Cochair
PAC International, 2000 4th Ave., Canby, OR 97013

Invited Papers

1:00

3pAA1. Volume limits on ASTM field sound test standards. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Current ASTM airborne and impact field sound transmission testing standards impose upper and lower limits on the room volume when testing floor-ceiling assemblies. This is particularly important when the testing is used to determine code compliance in multi-family residential buildings. Of primary concern is the current 150-m^3 volume limit on the receive room, which if exceeded, makes compliance with the sound transmission code impossible. This presentation will discuss the reason for these limits and what modifications to the standards are recommended to resolve this situation.

1:20

3pAA2. Comparing methods for evaluating sound insulation of doors. David W. Dong (Westside Acoust., 801 S Grand Ave. Ste. 1020, Los Angeles, CA 90017, wdong@westsideacoustics.com), John LoVerde, Samantha Rawlings, and Sunit Girdhar (Westside Acoust., Los Angeles, CA)

Traditionally, doors are specified by their STC rating and evaluated in the field using NIC rating per ASTM E336. However, NIC measurements of doors are usually a measurement of a composite assembly and often involve poorly defined spaces such as a corridor or exterior. As a result, the NIC is a measurement of the room conditions as much as the door performance. In contrast, the door insertion loss method (DTC rating per ASTM E2964) is a measurement of the performance of the door itself. It should therefore be expected that the DTC and NIC for a given door would differ, and that the difference would depend on the size and characteristics of the spaces on either side of the door. The authors analyze field measurements of DTC, NIC, and sound intensity of several door/wall composite assemblies to elucidate the effects of the many elements involved and when it is appropriate to use each type of measurement. The results also shed light on when it is appropriate to use a composite STC rating to describe the door/wall system.

Contributed Paper

1:40

3pAA3. Field-measured airborne sound isolation of partitions incorporating acoustical specialty products. Matthew A. Christian (Resonance Acoust., 364 Bush St. Fl. 2, San Francisco, CA 94104, mchristian@resonanceac.com) and Randy Waldeck (Resonance Acoust., San Francisco, CA 94104)

Acoustical specialty products, such as Scafco, Steel-Con, or Marino\-WARE SoundGuard metal studs, acoustical gypsum board, mass-loaded vinyl, and acoustical "glue" products (i.e., Green Glue), are commonly proposed to improve partition sound isolation where cost or space constraints limit the use of upgraded acoustical partitions using standard building materials. On several recent projects, resonance acoustics has had the opportunity to conduct field tests quantifying the sound isolation performance of wall assemblies incorporating these products. This presentation compares field-measured airborne sound transmission data to laboratory performance data, where available, for partitions incorporating specialty acoustical products. Observed deviations from laboratory data are examined, with particular emphasis on construction detailing, installation practices, and other field conditions that may influence field performance.

1:55

3pAA4. Prediction of field measured impact insulation ratings without the use of completed assemblies or floor mock-ups. Wilson Byrick (Pliteq Inc., 131 Royal Group Cresent, Woodbridge, ON L4H 1X9, Canada, wbyrick@pliteq.com)

Determining the field-measured impact insulation ratings of a floor-ceiling assembly can be a critical code compliance item to obtain certificates of occupancy. Generally, at least a small-scale mock-up of the floor assembly is required to obtain field measurements in accordance with ASTM E1007. In design phase, engineers rely on laboratory testing according to ASTM E492 as there is no building in which to test. The international building code stipulates that code compliance may be achieved by basing the design on a laboratory test with an IIC rating 5 points greater than the field rating desired or required. This tends to be an oversimplification and depending on the specific structural response of the building, field testing can vary widely from this assumption. In this paper, we utilize a database of over 2000 laboratory tested floor-ceiling assemblies and a data comparison and filtration script to develop a novel tool for the prediction of field-measured impact insulation class. This paper includes numerous case studies where predictions are compared to the actual field-measured results. The procedure includes a simple measurement of the structure *in-situ* and may be particularly useful for new or unusual floor types where exact laboratory data does not exist.

2:15–2:30 Break

2:30

3pAA5. Use of vibration transmissibility to interpret sound isolation measurements when approaching laboratory flanking limits. Drake A. Hintz (PAC Int., 11130 Z St., Omaha, NE 68137, Drake.A.Hintz@gmail.com), Michael Raley (PAC Int., Canby, OR), Evelyn Way, and Aimee Ramfjord (Maxxon Corp, Hamel, MN 55340)

High-performance acoustical isolation products are often specified in buildings, but obtaining meaningful laboratory data for these systems can be challenging. When testing airborne sound isolation in accordance with ASTM E90 and structure-borne impact insulation in accordance with ASTM E492, results for high-performing assemblies are commonly limited by laboratory flanking. Conventional metrics and frequency-based analysis can be insufficient in distinguishing performance between assemblies using these testing and analysis methodologies. Here, laboratory measurements were conducted on a common six-inch concrete slab in combination with several ceiling isolators that provide high levels of sound/vibration isolation. Resultant laboratory data are analyzed and used to show how laboratory flanking restricts the conclusions that can be made. Vibration measurements taken concurrently are analyzed using transmissibility to determine whether additional conclusions can be drawn for the relative performance of the various ceiling isolators.

Contributed Papers

2:50

3pAA6. Between slab and silence: A repeatability-first study of impact isolation in ceramic tile systems on an 8-in concrete slab. Felix Baronet (AcoustiTECH, 1050 Bd Leon Vachon, Saint-Lambert-de-Lauzon, QC G0S2W0, Canada, fbaronet@acousti-tech.com)

After roughly 15 years of going against the current, by performing rigorous acoustic tests either in real life conditions (field) or in large-scale mock-ups, we decided that it was time to invest into a first of its kind comparative study. Between slab and silence lies a narrow set of bands where a hard-finish footfall becomes either tolerable—or endlessly problematic. We present a laboratory case study of ceramic tile systems on a full-scale **8-in concrete slab**, designed for **repeatability-first** comparison. The campaign returns to **bare-slab checks** between flooring changes, producing an internal reference that quantifies stability and strengthens cross-system inference. A **rarely documented tile-only** condition provides a practical hard-surface midpoint. Interpretation centers on **third-octave normalized airborne and impact spectra**, distinguishing genuine isolation gains from spectral redistribution and highlighting the frequency regions that govern perceived improvement. Findings are reported **non-proprietarily** to focus on mechanisms and methods.

3:10

3pAA7. Case study: Comparing pink noise and bass music for noise isolation class testing. Martin J. Mann (Babich Acoust., 4 Jenny Lynn Ct, Apt C, Pittsburgh, PA 15239, martin@babichacoustics.com), AnnMarie R. Hirsh, and Jeffrey E. Babich (Babich Acoust., Pittsburgh, PA)

A particular hotel was interested in adding two live music venues to their first floor vacant shell space. These venues would be located directly beneath a majority of the hotel's existing guest rooms. Sound isolation testing was completed to evaluate the existing floor-ceiling assembly, including all penetrations such as guest room plumbing and grease duct shafts. The sound isolation testing was done in accordance with ASTM E336. With the knowledge that the new spaces were to be music venues, it was decided that testing would be done with pink noise per the ASTM standard plus music that included a lot of transient bass components. In this case study, the results between using pink noise and music as sound sources are compared. Additionally, increasing the number of 1/3 octave bands for a typical single number sound isolation rating was evaluated, along with potentially adding a low-frequency sound transmission class/noise isolation class number to improve sound isolation recommendations for designers.

Session 3pAB

Animal Bioacoustics, Computational Acoustics and Acoustical Oceanography: Machine Learning Approaches for Analyzing Large Acoustic Datasets II

Irina Tolkova, Cochair
K. Lisa Yang Center for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850

John Allen, Cochair

Invited Paper

1:00

3pAB1. Over the air domain adaptation of a deep learning detector–classifier. Mark Thomas (Halifax, JASCO Appl. Sci., 20 Mount Hope Ave., Dartmouth, NS B2Y 4S3, Canada, mark.thomas@jasco.com), S. B. Martin, and Eloise Callaghan (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada)

Effective acoustic monitoring of marine mammals requires systems capable of real-time adaptation in dynamic ocean environments. This talk presents a framework for deploying deep learning models on autonomous platforms, highlighting the transition from static architectures to adaptive edge computing. We first detail the development of a deep learning-based algorithm for multi-species detection, classification, and localization. We then present results from the deployment of this system on an uncrewed surface vehicle in Cape Cod Bay. Notably, by utilizing over-the-air updates to fine-tune the model with *in situ* data, we achieved an 82% reduction in the median false positive rate, underscoring the critical role of domain adaptation in operational settings.

Contributed Paper

1:20

3pAB2. Using supervised machine learning to classify six distinct bowhead whale social calls. Alexander J. Hare (Earth and Ocean Sci., Univ. of Victoria, 2923 Graham St., Victoria, BC V8T3Z5, Canada, alexhare1991@gmail.com), Stephen Insley (Wildlife Conservation Society Canada, Whitehorse, YT, Canada), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and William D. Halliday (Wildlife Conservation Society Canada, Whitehorse, YT, Canada)

Climate change is driving shifts in the Arctic marine environment towards being completely ice-free during the summer, facilitating increased vessel traffic. Underwater noise produced by vessels has the potential to significantly disrupt acoustic communication between marine mammals such as bowhead whales (*Balaena mysticetus*) through acoustic masking. Despite the impact that increased vessel noise may have on bowhead whales, we still have a poor understanding of how this species will respond to this environmental change. In order to address this knowledge gap, we extracted bowhead whale calls from continuous recordings taken between 2018–2021 at six sites around the eastern Beaufort Sea. We then filtered these calls for quality, retaining only high-quality calls for the extraction of 16 acoustic features for use in classification. Using a random forest model, we classified six distinct call categories with high reliability. This classification model lays important groundwork for examining bowhead acoustic behavior and how their calls change in response to vessel traffic. Making use of a 10-year acoustic dataset collected across the eastern Beaufort, we can assess the ecological impacts of acoustic masking on bowhead whales before vessel traffic increases dramatically in the Beaufort Sea.

1:35–1:50 Break

Invited Paper

1:50

3pAB3. Challenges and opportunities in combining calibrated sound pressure level data with emerging bioacoustics tools to gain novel insights about wildlife in national parks. Cathleen Balantic (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr. Ste. 200, Fort Collins, CO 80525, cathleen_balantic@nps.gov), Davyd H. Betchkal (Natural Sounds and Night Skies Div., National Park Service, Denali Park, AK), Joshua P. Cohen, and Emma L. Brown (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

The National Park Service has amassed a vast terrestrial record of acoustic data at over 1000 monitoring sites across more than 140 park units in the United States. This acoustic record comprises simultaneous digital audio and calibrated sound pressure level data streams collected to characterize the acoustic environment, noise sources, and spectral ambient sound level in national parks. Now, new tools for automated acoustic detection of wildlife are allowing managers to introspect the latent knowledge in acoustic records related to how wildlife respond to the acoustic environment. We illustrate emerging questions with a case study investigating spatiotemporal dynamics of the acoustic environment on automated detection volume of avian species using a subset of National Park Service sites in the Intermountain Region. Concurrently, the National Park Service is collaborating to make this large terrestrial audio dataset publicly available and accessible. This emerging work investigates questions about wildlife vocalization dynamics at scale and demonstrates opportunities offered by large audio datasets collected on public lands.

Contributed Papers

2:10

3pAB4. Generalizing computer vision for bioacoustics and detecting bearded seal vocalizations to new acoustic scenarios. 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), Erin M. Fischell (Acbotics Res., LLC, East Falmouth, MA), and Martin Siderius (Portland State Univ., Portland, OR)

Generalizing deep learning models across distinct acoustic environments remains a significant challenge in bioacoustics. This study addresses this by applying the YOLOv5 object detection algorithm, previously trained on Arctic recordings [10.1109/JOE.2023.3307175], to bearded seal spectrograms, treating bioacoustic monitoring as a computer vision task. While the model achieved high accuracy in its original domain, its robustness was rigorously tested in a controlled experiment at Ashumet Pond. This evaluation introduced significant environmental differences and included an out-of-distribution downsweep signal to further assess detection limits. Results demonstrate successful zero-shot transfer for known vocalizations despite the domain shift. Crucially, the network identified the new signal by its dominant downsweep feature, mapping it to the morphologically similar long downsweep once signal overlap decreased. This new signal is then incorporated into the training classes to refine classification accuracy, ensuring the system can accurately distinguish and classify emerging call types across diverse underwater habitats. [Work supported by NSF award ID 2322365.]

2:25

3pAB5. Acoustic classification of false killer whale populations across the Central Pacific using machine learning. Brijonnay Madrigal (Scripps Inst. of Oceanogr., Univ. of California San Diego, 8635 Kennel Way, La Jolla, CA 92037, brmadrigal@ucsd.edu), Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Jennifer L. McCullough (NOAA Fisheries, Honolulu, HI), Yvonne Barkley (CIMAR, Honolulu, HI), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

False killer whales (*Pseudorca crassidens*) occur throughout the Central Pacific Ocean including the Marianas and Hawaiian Islands. Fishery interactions have led to increased threats and the subsequent decline of Hawai‘i populations necessitating the advancement of passive acoustic monitoring techniques to reliably validate population designation and improve stock assessment and management. Our goal is to develop a machine learning approach to acoustically discriminate populations on a regional scale (Marianas/Hawai‘i), providing a critical foundational framework for future Hawai‘i population-level discrimination. Acoustic data were collected from NOAA ship-based line-transect surveys from 2010–2024, encompassing 13 cruises. Whistles and clicks were detected using PAMGuard. Genetic, visual, acoustic, and spatial information was used to verify population which provided labeled data for supervised machine learning. We are implementing multiple methods including convolutional neural networks and clustering algorithms to assess if these approaches can capture regional differences from recorded signals. Preliminary results from over 115 encounters indicate a subset of whistle contours can be clustered based on contour similarity; however based on an evaluation of model performance, it remains unclear whether these data are regionally distinct. Intraspecific acoustic classification is challenging so we will compare methodologies, highlight current progress, discuss inherent challenges, and address conservation implications.

2:40–3:00 Panel Discussion

Session 3pAO

Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, Signal Processing in Acoustics, and Physical Acoustics: New Results and Future Directions for Ocean Sensing Using Seismic Reflection Platforms

Likun Zhang, Cochair
National Center for Physical Acoustics and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38655

Vanessa M. ZoBell, Cochair
Biological Oceanography, Scripps Institution of Oceanography, 8622 Kennel Way, La Jolla, CA 92037

Madusanka Madiligama, Cochair
Phys. and Astronomy, Univ. of Mississippi, Dept. of Phys. and Astronomy, 108 Lewis Hall, University, MS 38677

Invited Paper

1:00

3pAO1. Using a single airgun as a low-frequency impulsive source for ocean acoustics experiments in coastal waters. Julien Bonnel (Scripps Inst. of Oceanogr., 9500 Gilman Dr., # 0238, San DIego, CA 92093, jubonnel@ucsd.edu), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712), and Andrew R. McNeese (ARL:UT, Austin, TX)

Ocean acoustics experiments rely on a variety of sources and receivers to characterize acoustic propagation and the ocean environment (water column, sea surface and sediments) at various time and spatial scales. This talk focuses on low-frequency ($f < 500$ Hz) propagation in coastal water (depth < 500 m) over mid-to-long ranges ($r < 25$ km). In this context, we explore the use of a single GI-210 airgun to generate low-frequency impulsive sounds, and its capacity to actively minimize unwanted residual bubble pulses. This generates a source signal that is nearly a perfect impulse, and removes the need for source deconvolution. Additionally, a thorough comparison is made with two other tomographic impulsive sources used in the same context: Navy explosives (SUS charges) and inert implosive rupture-disk systems (RIUSS). The study relies on various experimental datasets, all collected on the New England Mud Patch (NEMP). Both near-field and propagated signals are compared using the following metrics: peak level, peak-to-bubble ratio, SEL, spectral content, and time–frequency dispersion of normal modes. Advantages and drawbacks of each system for environmental inversion are discussed. [Work supported by the Office of Naval Research.]

Contributed Papers

1:15

3pAO2. Debubbling strategies in marine seismic acoustics for recovering weak water-column reflections. Madusanka Madiligama (Inst. for Data Sci., Univ. of Mississippi, Dept. of Phys. and Astronomy, 108 Lewis Hall, University, MS 38677, mbabeyko@go.olemiss.edu) and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS)

Marine seismic surveys employ airgun sources that generate a strong primary acoustic pulse followed by persistent bubble oscillations, commonly referred to as bubble waves. While these bubble waves have a limited impact on conventional sub-seafloor imaging, they significantly degrade data quality in applications that rely on weak water-column reflections, such as seismic oceanography and acoustic environmental analyses [Zou *et al.*, *Ocean Sci.* **17**(4), 1053–1066 (2021); Zou and Zhang, *JASA* **150**(5), 3852–3860 (2021)]. The spectral and temporal overlap between bubble waves and desired signals makes their suppression particularly challenging, especially in shallow-water environments. This talk reviews recent approaches developed to mitigate bubble-wave contamination in marine seismic data, with an emphasis on post-acquisition debubbling techniques [Madiligama *et al.*, *JASA* **158**(3), 2420–2447, 2025]. Methods including signature deconvolution, matched filtering, and wavelet-based extraction are examined, highlighting their underlying assumptions, performance limitations, and sensitivity to source characterization and propagation effects. Emerging data-driven and physics-informed approaches are also discussed, focusing on their potential to improve robustness while preserving physically meaningful signal content. Key challenges associated with generalization, artifact generation, and validation are identified. [Work supported by NOAA.]

1:30

3pAO3. Far-field signal comparison between a Tuned Pulse Source and conventional seismic airguns. Ludovic Tenorio-Hallé (Marine Conservation Res., 94 High St., Kelvedon C05 9AA, United Kingdom, ltenorio@mcr-team.org), Lynne Hodge (NOAA SEFSC, Miami, FL), Sean Wiggins (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Vanessa M. ZoBell (Biological Oceanogr., Scripps Inst. of Oceanogr., La Jolla, CA), John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Lance Garrison (NOAA SEFSC, Miami, FL), Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Melissa Soldevilla (NOAA SEFSC, Miami, FL)

Seismic airguns are a major contributor to underwater noise in the Gulf of Mexico (GoM). This study compares opportunistic passive acoustic measurements from a recentlydeveloped Tuned Pulse Source™ (TPS) and a conventional airgun array. Recordings from six sites across the GoM were analyzed to detect pulses from each source type and quantify their received levels (RLs). Measurements from the nearest site (50–90 km range) showed RL reductions of 10–20 dB for the full-band TPS signals compared to those of conventional airguns. Long-range measurements (50–600 km) showed reductions of 10–15 dB at 40 Hz. As a result of a lower source level, individual TPS pulses were detectable over shorter distances than those of conventional airguns, which were detected at ranges of up to 1000 km. Bathymetry strongly influenced propagation, with efficient propagation in deep waters and rapid attenuation over continental slope and shallower regions. Sound propagation modeling at 40 Hz from a location sampled by both surveys predicted a ∼43% reduction at 1000 depth in the area where TPS signals were detectable relative to conventional airguns. The combined *in situ* measurements and modeling demonstrate substantial potential for alternative seismic sources to reduce anthropogenic noise levels and lessen impacts on marine fauna.

1:45–2:00 Break

2:00

3pAO4. Seismic airgun array directivity across the Gulf of Mexico. Vanessa M. ZoBell (Scripps Inst. of Oceanogr., Univ. of California San Diego, 8622 Kennel Way, La Jolla, CA 92037, vmzobell@ucsd.edu), Sean Wiggins (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Ludo Tenorio-Halle (San Diego, CA), Lynne Hodge (NOAA NMFS Southeast Fisheries Sci. Cente, Boulder, CO), John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Arturo Serrano (Universidad Veracruzana, Veracruz, Mexico), Adolfo Gracia (Instituto de Ciencias del Mar y Limnología, Mexico City, Mexico), Melissa Soldevilla (NOAA NMFS Southeast Fisheries Sci. Cente, Miami, FL), and Kaitlin Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Ocean noise in the Gulf of Mexico/America arises primarily from seismic airgun surveys (SAGS). Airgun pulses are the most pervasive low-frequency signals in the Gulf, sometimes propagating hundreds of kilometers from their source. The LISTEN project has maintained a decade of time-synchronized passive acoustic monitoring (PAM) stations spanning the continental shelf, slope, and abyssal depths providing a large aperture PAM array in the region. These long-term recordings reveal that airgun activity is nearly continuous throughout much of the Gulf and has produced some of the highest low-frequency sound levels observed in U.S. waters. The array recordings also show that airgun surveys exhibit distinct directivity patterns of radiated sound resulting in higher sound levels at frequencies above 100 Hz depending on transit line heading relative to monitoring stations. Using an airgun array source model and parabolic equation propagation loss model, we modeled the directivity patterns across the Gulf and validated these predictions with *in situ* sensors. Model accuracy varied by frequency, site location, source location, and a range of environmental variables tested. These results contribute to the development of predictive modeling for ambient noise in the Gulf and reinforce previous research that highlights the influence of airgun surveys on high ambient noise levels in the Gulf.

Invited Papers

2:15

3pAO5. A fully automate framework for seismic mapping of ocean turbulence. Jingxuan Wei (School of Ocean and Earth Sci., Univ. of Southampton, National Oceanogr. Ctr., European Way, Southampton SO14 3ZH, United Kingdom, jingxuan.wei@soton.ac.uk), Kathryn Gunn (School of Ocean and Earth Sci., Univ. of Southampton, Southampton, United Kingdom), Shijun Cheng (Div. of Physical Sci. and Eng., King Abdullah Univ. of Sci. and Technol., Thuwal, Saudi Arabia), Bieito F. Castro, Alessandro Silvano, Alberto N. Garabato, Espe Broullón, and Timothy Henstock (School of Ocean and Earth Sci., Univ. of Southampton, Southampton, United Kingdom)

Turbulent mixing is a fundamental process maintaining the global overturning circulation. However, its spatiotemporal variability makes it challenging for traditional observations to picture a comprehensive global pattern. Seismic oceanography (SO) is an acoustic technique that utilizes marine seismic reflection surveys to detect small temperature and salinity variations across the water column at $O(10)$ m resolution. Despite two decades of development and a unique ability to map turbulent mixing with unprecedented spatial coverage, SO remains underexploited by the broader oceanographic community. This issue stems from a steep learning curve rooted in geophysical jargon and complex data processing, further obscured by a lack of standardized workflows and uncertainty quantification. To bridge this interdisciplinary gap, we developed a Python-based software that automates the data processing pipeline for mixing rate derivation. The framework integrates seismic data processing, two-step sound speed inversion, internal wave spectra analysis, and uncertainty quantification into a single, streamlined execution, reducing processing time from months to hours. This tool will provide a standardized, user-friendly approach for researchers from all backgrounds, facilitating wider adoption of SO in physical oceanography. To demonstrate its capability, we present preliminary results investigating the spatial distribution and variability of turbulent mixing processes around the East Antarctic shelf seas.

2:35

3pAO6. Environmental characterization using ambient seismic sources. John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu) and Bruce Cornuelle (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093)

We present an approach for characterizing the ocean environment using ambient seismic sources generated by commercial airgun surveys, with a focus on the Gulf of Mexico (GOM). Seismic surveys conducted for offshore exploration provide a dense and persistent acoustic source field. Automatic identification system (AIS) vessel tracking reveals extensive airgun activity across the GOM, creating an opportunity to exploit these signals for long-range acoustic propagation studies. Acoustic data were collected during an active seismic survey period in January 2015, with airgun signals from five survey vessels detected at five receivers over propagation ranges of

approximately 50 to 500 km. We developed a software tool, *RainbowClick*, to separate airgun pulses from individual source ships, enabling detailed analysis of pulse timing and structure. Signal propagation was modeled using a parabolic-equation approach within a sound-speed field derived from the MITgcm-GOM oceanographic model, incorporating bathymetry, water-column structure, and sediment properties. These long-range propagation paths require accurate treatment of both surface and bottom interactions. Source localization based on first-arrival time-difference measurements showed strong agreement with AIS ship positions, while detailed waveform modeling remains an ongoing challenge.

WEDNESDAY AFTERNOON, 13 MAY 2026 FRANKLIN HALL A, 1:00 P.M. TO 3:00 P.M.

Session 3pBA

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Award Poster Session

Virginie Papadopoulou, Cochair
Dept. of Radiology, Univ. of North Carolina, Chapel Hill, Chapel Hill, NC 27599

Timothy Hall, Cochair
Dept. of Biomedical Engineering, Univ. of Michigan, Ann Arbor, MI 48109

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.

1aBAa7. Improving photoacoustic imaging through the skull using deep learning: Considering 3D effects
Matthew Olmstead

1aBAa12. Sub-millimeter 3D-printed mouse brain vascular phantom for ultrasound imaging validation
Student presenter: Luca Giaccone

1aBAb1. MICROBEAD: A novel Doppler detectable device with suspended microparticles for use as a biopsy marker
Student presenter: Benjamin Wood

1aBAb4. Evaluating the likelihood of treatment success using the passive cavitation signal from a test pulse prior to focused ultrasound treatments of tendinopathies
Student presenter: Grace M Wood

1aBAb6. Optimization methods for trajectory estimation in low-cost 3D ultrasonic imaging
Student presenter: Brian Boitnott

1pBA2. Non-invasive ICP Monitoring and ischemia detection via transcranial and ocular ultrasound
Student presenter: Sophie Haddad

1pBA6. Acoustic micro-rheology via droplet cavitation dynamics
Student presenter: Sander Spiekhout

1pBA8. Non-invasive mapping of arterial compliance and pulse pressure using physics-informed neural networks and ultrasound imaging: In silico, phantoms validation, and *in vivo* human feasibility
Student presenter: Tuhin Roy

1pBA10. Ultrasonic guided wave elastography for early detection of atherosclerosis: a novel three-layer k-Wave simulation framework
Student presenter: Azin Nadi

1pBA12. Quantitative ultrasound characterization of vessel density: Effects of microbubble concentration
Student presenter: Parniyan Norouz Zadeh

2aBAa2. High-speed optical dynamics of repeated acoustic vaporization of a single droplet
Student presenter: Sander Spiekhout

2aBAa3. Design and construction of a low-cost amplifier for focused ultrasound applications
Student presenter: Joshua Elbertson

2aBAa4. Ultrasound-induced dynamics of nanodroplets and microbubbles in synthetic cystic fibrosis mucus: Toward pulmonary drug delivery and mucus rheology assessment
Student presenter: Hieu Doan

2aBAa8. Quantitative cavitation imaging: Computational considerations and potential pitfalls
Student presenter: Abigail Collins

2pBAa2. Characterizing cavitation behavior in tissue mimicking phantoms under single-cycle histotripsy
Student presenter: Ruth Mutua

2pBAa3. Evaluation of pressure-modulated single cycle histotripsy methods for efficient and tissue-selective ablation
Student presenter: Sara Elna has

2pBAa4. Focused ultrasound targeting of the cribriform plate/skull base enhances intranasal brain delivery and glymphatic transport
Student presenter: Siaka Fadera

2pBAa5. Ultrasound-enhanced bacteriophage delivery results in transient bacterial reduction within complex biofilms
Student presenter: Sara Keller

2pBAa6. Modelling and the search for mechanisms of sonodynamic therapy
Student presenter: Stephanie Carmen Walton

2pBAb4. Near-real-time passive cavitation imaging via GPU-accelerated sparse frequency domain beamforming
Student presenter: Rahul Sheoran

2pBAb6. Power-Passive Acoustic Detection and Dynamic Differentiation: Mechanisms of Subspace Selection and Power Estimation
Student presenter: Nathan Caso

2pBAb7. Beamforming with large elements for an increased field of view in 3D ultrasound B-mode imaging
Student presenter: Mick Gardner

2pBAb8. Diffraction factor of a one-dimensional rectangular linear array for quantitative ultrasound imaging
Student presenter: Khalid Abdall

2pBAb9. Applications of beamforming using quasi-holographic back propagation for extracting information about scattering by penetrable objects
Student presenter: Heather A Moon

2pBAb10. Shear wave holography construction of a simplified source condition for focused shear wave beams in soft elastic media
Student presenter: Branch T Archer

3aBAa4. Multi-feature acoustic vortex characterization and automated quality classification via physics-informed ensemble model
Student presenter: Chadi Ellouzi

3aBAa5. Preliminary results of kidney stone comminution using acoustic vortices
Student presenter: Sergio Maldonado-Ortega

3aBAa7. Acoustic vortices stimulation on HPSC-derived cardiac organoids
Student presenter: Chengzhi Wang

3aBAa8. Iso-propagation vortex waves for robust long-range underwater acoustics
Student presenter: Jianyu Hua

3aBAb1. Quantitative lung ultrasound spectroscopy vs computed tomography: A comparative study to differentiate pneumonia and cardiogenic pulmonary edema
Student presenter: Mattia Perpent

3aBAb4. Multicenter clinical study on pleural sliding quantitative characterization by means of multifrequency lung ultrasound
Student presenter: Chiara Zan Grandi

3aBAb6. Passive cavitation detection analysis of color Doppler twinkling materials with varying levels of peak negative pressure
Student presenter: Benjamin Wood

3aBAb7. Comparing power Doppler and ultrasound contrast agent imaging methods on ex vivo kidney stones
Student presenter: Jay M Cliftmann

4aBA10. Characterization of flow and dose-dependent histotripsy bubble dynamics for DVT models
Student presenter: Eric Joe Galindo

4aBA11. Histotripsy-induced activation of platelets in whole blood
Student presenter: Erik Saucedo

4pBAa1. Characterization of pre-focal cavitation during histotripsy and Its influence on focal cavitation dynamics
Student presenter: Leah R Holmes

4pBAa2. Ultrasound-guided delivery of fatty acid nitroalkene (CP8)-loaded microbubbles for treating triple-negative breast cancer
Muhammad Wahab Amjad

4pBAa3. Ultrasound-targeted microbubble cavitation for the treatment of myocardial microvascular obstruction: A porcine study
Student presenter: Soheb Anwar Mohammed

4pBAa4. Non-viral targeted inducible nitric oxide synthase (ins) delivery to the brain for modulating sterile inflammation induced by ultrasound-mediated BBB opening
Student presenter: Aref Ghaderi

4pBAa5. Acoustic emissions-guided focused ultrasound for liquid biopsy in infiltrative gliomas: a multicenter, open-label, phase 1/2 trial
Student presenter: Pavlos Anastasiadis

4pBAb3. Wearable ultrasound technology for deep tissue monitoring and beyond
Student presenter: Muyang Lin

4pBAb6. Simultaneous real-time multichannel M-mode and Doppler imaging in a miniature wearable ultrasound device
Student presenter: Abhishek Aher

4pBAb7. Printable and flexible ultrasound transducers using laser-induced porous graphene and piezopolymer
Student presenter: Ehsan Ansari

4pBAc1. Ultrasound control of cell fate determination of human induced pluripotent stem cell-derived neural progenitors *in vivo* via heat shock promoters
Student presenter: Kama Bell

4pBAc2. Pressure responses of different lipid-shelled microbubbles: A perspective on pressure dependent stability and subharmonic response
Student presenter: Mehmet Yapar

4pBAc4. Comparing focused ultrasound to dry needling in two *ex vivo* tendinopathy models
Student presenter: Surabhi J Menon

Session 3pCA

Computational Acoustics: Showcases of High Performance Computing in Acoustics (Poster Session)

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

Timothy Walsh, Cochair
Simulation Modeling Sci., Sandia National Laboratories, Sandia National Laboratories, MS 0897, Albuquerque, NM 47906

Ralph T. Muehleisen, Cochair
Energy Systems, Argonne National Laboratory, 9700 S. Cass Ave., Bldg 362, Lemont, IL 60439-4801

All posters will be on display and all authors will be at their posters from 1:30 p.m. to 3:00 p.m.

Invited Paper

3pCA1. Computational methods for acoustic radiation problems with rotational symmetries. Kenneth Czuprynski (ARL / Penn State, 5130 ARL Main Lab, University Park, PA 16802, kdc168@psu.edu) and John B. Fahnline (ARL / Penn State, State College, PA)

Structural acoustic problems that couple finite element and boundary element methods typically require solving large, dense linear systems arising from the boundary element portion of the analysis. For large structural acoustics problems, this can become computationally prohibitive. However, when the boundary surface in question is rotationally symmetric, the linear system becomes block circulant, allowing efficient solvers to be applied. If the boundary surface contains only portions that are rotationally symmetric or contains nested rotational symmetries, the matrix is no longer block circulant but still exhibits some circulant structure within sub-matrices. In this work, we present methods that take advantage of these structures, discussing computational improvements and amenability to parallelism in HPC environments.

Contributed Papers

3pCA2. Exploration of gradient and concept-based neural network interpretation through rotorcraft classification. Gordon M. Ochi (U.S. Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil) and Matthew G. Blevins (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Rotorcraft have unique acoustic emissions tied primarily to their blade pass frequencies. This study builds upon previous results of a Welch power spectral density based multi-layer perceptron (MLP) artificial neural network (ANN) that achieves 93% accuracy and F1 score on a large acoustic rotorcraft emission dataset. Seeking to understand the model's analysis, we explore techniques typically used on images in gradient-based and concept-based interpretability methods. We show that gradient-based methods like Saliency, Smooth Grad, and Gradcam generalize down to the 1-D feature space of MLPs. Results of these methods are shown to highlight the apparent spectral energy bins of importance for the ANN, although the insights do not match expectations of harmonic peak identification. Two higher level concepts are also explored through the Testing with Concept Activation Vectors (TCAV) method: The removal of all harmonic peaks versus retaining only the harmonic peaks. TCAV results indicate that the suitability of either high level concept is highly class dependent.

3pCA3. Optimal sizing of perfectly matched layers for acoustic analyses. Alyssa Bennett (Naval Surface Warfare Ctr. Carderock, 9500 MacArthur Blvd., Bethesda, MD 20817, alyssa.m.bennett7.civ@us.navy.mil), Abram Huang, Patrick Kozlowski, and Kuangcheng Wu (Naval Surface Warfare Ctr.—Carderock Div., West Bethesda, MD)

Numerical analyses have been widely used to predict acoustic radiation and scattering of submerged structures subjected to internal or external excitations. One of challenges in the numerical analyses is to properly model the structure's surrounding unbound domain and satisfy Sommerfeld boundary condition. Instead of explicitly modeling the unbounded domain, various methodologies have been developed to reduce computational burden and meet the Sommerfeld boundary condition. Those methodologies include infinite elements, perfectly matching layer (PML), and others. This study will focus on PML which is an artificial absorbing layer to absorb energy entering the layer. A decay function is used to represent the rate of absorption within the PML. This study investigates the optimal sizing of the PML domain in a large FE model to minimize computational resources while ensuring the proper implementation of the Sommerfeld boundary condition across all frequencies of interest.

3pCA4. Scalable 3-D simulations of attenuating nonlinear acoustic waves with MPI and the discontinuous Galerkin method. Drew A. Murray (Comput. Sci. and Eng., Michigan State Univ., East Lansing, MI), Jacob S. Honer (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, East Lansing, MI 48824, mcgough@egr.msu.edu)

A parallel discontinuous Galerkin (DG) code that models 3-D nonlinear, lossy wave propagation is described and evaluated. The DG approach is applied to the conservation laws of mass and momentum to yield the necessary update equations for simulating nonlinear full-wave motion, and additional terms inject loss and absorbing boundary conditions into the solution. This algorithm is efficiently parallelized with OpenMPI operating on small "halo" layers of elements. The wave propagation results are presented graphically using slices of the 3-D data in which the characteristic shape of nonlinear waves is readily visible. Parallel efficiency is measured using tests of both strong and weak scaling, comparing a wide range of core counts over several different problem sizes. The strong scaling and weak scaling runtime performance are evaluated up to a maximum of 1040 cores distributed across thirteen compute nodes. The results of these scaling tests show that the parallel performance of the code approaches linear scaling, with the largest problem size and core count achieving an efficiency of 83% (compared to serial execution) for strong scaling on 960 cores. Similarly, the weak scaling showed 77% efficiency (compared to serial execution) on 1040 cores.

3pCA5. Validation of a 3-D high-order discontinuous Galerkin solver with transducer boundary conditions through comparisons with the fast nearfield method. Jacob S. Honer (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), Drew A. Murray (Comput. Sci. and Eng., Michigan State Univ., East Lansing, MI), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw Ln., Rm. 2120, Michigan State University, East Lansing, MI 48824, mcgough@egr.msu.edu)

A 3-D discontinuous Galerkin (DG) solver is presented for ultrasound propagation in heterogeneous media. The formulation advances particle velocity and density perturbation fields using explicit time integration, while numerical fluxes couple elements and enforce material interfaces, enabling layered configurations and spatially varying material properties. DG is ideal for these calculations due to high-order accuracy, element-local operators, and a flux-based boundary-condition model for finite-aperture sources. Implementation choices governing accuracy and computational cost are considered, comparing consistent and diagonal (lumped) formulations. Transducer boundary conditions are implemented for single-element transducers and linear arrays. Two enforcement strategies are investigated: a pointwise imposition of the boundary drive at face evaluation points, and an L^2 projection of the drive onto the face polynomial space. The projected approach reduces aliasing for coarse meshes and high-order discretizations by representing boundary data consistently within the face polynomial basis. Linear validation in homogeneous media is performed via comparisons to the fast nearfield method for single-element and multi-element configurations, including focused sub-apertures with transmit delays. Transient pressure fields and centerline waveforms are compared in order to demonstrate accurate boundary-driven excitation across various aperture and focusing conditions.

3pCA6. Comparing clustering algorithms and input features for preliminary labeling of acoustic data samples. Emily C. Bacon (Phys. and Astronomy, Brigham Young Univ., N286 ESC, Provo, UT 84602, e2cook98@byu.edu), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Paul R. Leary, Kay L. Gemba, and Kevin B. Smith (Naval Postgrad. School, Monterey, CA)

Labeling acoustic data for use in supervised learning algorithms can be a long process, especially for large datasets. This work explores how different unsupervised clustering algorithms and input features affect the resulting clusters. The goal is to determine which combination of algorithm and input features shows most promise as a tool to obtain preliminary data labels for large acoustic datasets. This approach is tested on pressure and vector sensor data from the Monterey Accelerated Research System (MARS). This work compares the results of a k-means clustering algorithm against other clustering algorithms. In addition, the impact of the parameter sets included in the input feature vector, such as bearing, peak level, OASPL, is investigated. Combinations of algorithms and input features will be compared, and an assessment presented for the potential in using feature inputs and clustering as a preliminary labeling tool for large acoustic sets.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Andrew C. Morrison, Chair
Natural Sci., Joliet Junior College, 1215 Houbolt Dr, Joliet, IL 60431

Chair's Introduction—2:15

Invited Paper

2:20

3pED1. On the joys of teaching acoustics to graduate students. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

Tom Rossing (who established the Acoustics Education Prize) was my master's degree advisor. I have written elsewhere about the benefits of being his student. While much of his influence on my academic career involved experimental projects, it was Rossing who introduced me to ASA and acoustics education. My academic career (so far) has been almost equally split between 16 years teaching physics at a small private undergraduate institution and 15 years teaching graduate level acoustics at a large public R1 university. Some academics thrive on research; I thrive on the opportunity to study and teach acoustics. In this lecture, I will attempt to explain why teaching acoustics to graduate students brings me so much joy. I get to teach seven different acoustics graduate level courses covering fundamental and advanced theoretical topics and applications from noise control to musical instruments. I have access to a well-equipped laboratory space for developing demonstrations and experimental projects. I get to interact with smart students who ask challenging questions (many of my acoustics animations were inspired by questions from students). And every semester I get to continue learning more about the subject I fell in love with during my own graduate education.

Session 3pID

Interdisciplinary and Student Council: Hot Topics in Acoustics

Caner Baydur, Cochair
College of Design and Innovation, Tongji Univ., Siping Rd. No 1239, Shanghai, 200092, China

Sarah R. Bellavance, Cochair
Communicative Sci. and Disorders, New York Univ., 665 Broadway, 6th Floor, New York, NY 10012

Cassidy A. Christie, Cochair
Univ. of Texas at Austin, Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78758

Makayle S. Kellison, Cochair
Phys., Rollins College, N283 Eyring Science Center, Provo, UT 84602

Invited Papers

1:00

3pID1. Acoustic metamaterials for noise control: Sculpting silence. Yun Jing (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, jing.yun@psu.edu)

In recent years, acoustic metamaterials have attracted significant attention as engineered structures capable of manipulating sound waves in ways not possible with conventional materials. When designed properly, they can offer excellent noise reduction—through either reflection or absorption—while maintaining a deeply subwavelength thickness. In this talk, I will provide an overview of recent developments in acoustic metamaterials specifically for noise control. I will begin by discussing mechanisms for narrow-band, near-perfect sound absorption, followed by strategies for broadband absorption enabled by optimized non-local coupling. I will also examine how the causality principle governs these behaviors and defines the physical limits of absorption performance. Additionally, I will discuss ventilated acoustic metamaterials, which provide the dual capability of blocking sound while allowing for the passage of airflow. Combining results from the broader research community with findings from our own group, I will conclude by offering a perspective on the future challenges and opportunities in this evolving research area.

1:20

3pID2. Recent advances in histotripsy. Tatiana Khokhlova (Dept. of Medicine, Univ. of Washington, 1013 NE 40th St., CIMU, Portage Bay Bldg., Seattle, WA 98125, tdk7@uw.edu), George R. Schade (Dept. of Urology, Univ. of Washington, Seattle, WA), Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Gilles P. Thomas (APL-CIMU, Univ. of Washington, Seattle, WA), Gerald L. Lee (Bioengineering, Univ. of Washington, Seattle, WA 98125), Minho Song (Radiology, Stanford Univ., Palo Alto, CA), Pavel B. Rosnitskiy (Dept. of Medicine, Univ. of Washington, Seattle, WA), Ekaterina M. Ponomarchuk (Acoust., Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Oleg A. Sapozhnikov (Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Univ. of Washington/Moscow State Univ., Moscow, Russian Federation)

Histotripsy is a pulsed focused ultrasound-based method to non-invasively liquefy tissue or other material through localized bubble activity. Since its inception over two decades ago histotripsy has seen a remarkable growth in the number of its techniques and biomedical applications, and in clinical adoption, culminating in the recent FDA approval of a histotripsy system for mechanical ablation of liver tumors. This talk will cover the physical mechanisms behind different histotripsy techniques and the associated instrumentation, principles of ultrasound imaging guidance, and bioeffects of histotripsy in different medical contexts. Current challenges and emerging ways to mitigate them will also be discussed.

1:40

3pID3. From acoustic features to clinical meaning: Emerging technologies for interpretable AI in pediatric speech. Marisha Speights (Commun. Sci. and Disord., Northwestern Univ., Frances Searle Bldg., 2240 Campus Dr., Evanston, IL 60208, marisha.speights@northwestern.edu), Vishal Shrivastava, and Chethan Saligram (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Interpretable artificial intelligence (AI) has become an increasingly important topic in speech communication as data-driven methods are used to support clinical decision-making in speech health. This talk highlights acoustic technologies that advance interpretable AI for pediatric preschool-age speech. Recent work on computer-assisted syllable analysis of continuous speech, published in the 2024

JASA special issue on *Acoustic Cue-Based Perception and Production of Speech* (Speights *et al*., 2024), demonstrates how linguistically structured, landmark-based acoustic representations improve robustness and interpretability relative to conventional spectral features and clinical metrics. Advances in automatic speech recognition further enable intelligibility to be characterized using Speech Intelligibility Probabilities (SIPs), which rely on model-intrinsic uncertainty rather than transcript-based accuracy metrics. Phoneme- and language-level recognition analyses additionally reveal age-stratified confusion structure associated with speech development. Together, these developments point toward future directions in which interpretable, developmentally grounded representations play a central role in advancing speech acoustics and clinical AI for pediatric speech.

Session 3pNS

Noise: Pickleball Noise

Braxton Boren, Chair
Performing Arts, American Univ., 4400 Massachusetts Ave. NW, Washington, D.C. 20016

Chair's Introduction—1:00

Invited Papers

1:05

3pNS1. The distance myth for pickleball noise. Dale Van Scoyk (PSM Consulting LLC, 26494 Clarkston, Bonita Springs, FL 34135, dale@pickleballsound.com) and Barry R. Wyerman (ADC LLC, Huron, OH)

The rapid growth of pickleball has introduced new community noise challenges, particularly in residential environments. In response, some municipalities have proposed or adopted fixed minimum setback distance requirements between pickleball courts and residences as a primary noise mitigation strategy. The effectiveness of setback distance alone in reducing community noise impact is highly variable and depends on numerous interacting factors, including source characteristics, court orientation, propagation paths, ground effects, topography, and noise mitigation strategies. Fixed distance thresholds fail to account for these variables and do not reliably predict either sound levels or perceived annoyance. Field measurements, sound modeling software, and case study observations demonstrate the limitations of distance-based criteria. Alternative performance-based approaches are discussed, emphasizing site-specific acoustic assessment, with sound power characterization, operational controls, noise mitigation strategies, and outcome-based noise limits. Recommendations for evidence-based regulatory frameworks are made that better align with established acoustical science while balancing recreational access and community well-being. This information shows that setback distance limits are acoustically unsound, insufficiently protective, and potentially misleading when used as a stand-alone regulatory tool.

1:25

3pNS2. A recommended metric and noise standard for pickleball noise. Barry R. Wyerman (PSM Consulting LLC, 1125 Sheltered Brook Dr., Huron, OH 44839, wyerbr@gmail.com) and Braxton Boren (Performing Arts, American Univ., Washington, DC)

The impulsive noise from a pickleball impact can be measured with several different settings on a sound level meter or with—peak, impulse, fast, slow, Leq, and Ldn. These settings will result in different decibel levels for the same pickleball impact. The proper metric will be the one that is both easily measured and best quantifies the loudness or annoyance from pickleball. An adjusted Ldn is used in ANSI S12.9 Part 4 for environmental noise assessment and in ANSI S12.9 Part for compatible land usage. There are several reasons why this metric has limitations when used for pickleball site analysis. The limitations of ANSI S12.9 Part 4 and Part 5 for pickleball noise will be outlined. A preferred metric for pickleball noise will be described based on the maximum sound pressure level with a fast meter response and an A-weighting. This metric accounts for the variation in sound level of different pickleball impacts, for the frequency response of human hearing to these impacts, and for the transient response of hearing to impulsive sounds from pickleball. Finally, a recommended noise standard for pickleball noise is proposed that could be used by any municipality as an addendum to a noise ordinance to address the unique characteristics of pickleball noise.

1:45

3pNS3. Psychoacoustic noise metrics for pickleball sound. Braxton Boren (Audio Technol., American Univ., 4400 Massachusetts Ave NW, Washington, DC 20016, boren@american.edu)

Pickleball is the fastest-growing sport in the US, with over 26 000 outdoor pickleball courts built or converted since 2018. The hard paddle and ball used in the sport produce a high-frequency impulse noise burst which leads to many more noise complaints than from tennis or most other racquet sports. The annoyance from pickleball sound has also led to discussions of the most appropriate way to measure pickleball noise acoustically, with different proponents using long-term LAeq or SEL with an impulsive penalty, LApk, LAFmax, or per-unit SEL (sometimes with another impulsive penalty added). This study applies existing perceptual metrics for annoyance from impulsive noise to pickleball strikes based on the level difference (D_L) and onset rate (R_{on}) of the strikes. Correlations between different acoustic metrics and the perceptual metrics are examined, with some comments given regarding those with the best predictive power for pickleball noise annoyance in municipal noise codes.

2:05

3pNS4. Architected cores for modified pickleball paddle vibration. Robert M. Unetich (Pickleball Sound Labs, 1811 Woodlands Circle, Pittsburgh, PA 15241, itsrmu@gmail.com), Michael Rapp, Emma Kim, Eisa Al Mazrouei, Kevin Zhang, Mohammad Islam, and Vincent Sokalski (Carnegie Mellon Univ., Pittsburgh, PA)

The primary cause of the "pop" sound of pickleball play is air-pressure vibration associated with paddle vibration. This project examines approaches to reduce the magnitude and duration of this vibration through modification of the mechanical response of the paddle, including the use of architected internal structures. While softer paddle surfaces, can reduce vibration amplitude and duration, they typically do so at the expense of desirable mechanical play characteristics. This work explores how architected core materials, including 3D-printed auxetic and other anisotropic lattices, can alter the coupled vibration modes of the paddle face while retaining the play characteristics of high-performance paddles. In the design described, conventional are replaced by a mechanical architecture with tailored vibration modes that interact with those of the paddle face. Internal structures with prescribed values of Young's modulus and Poisson's ratio are selected to counteract deformation of the exterior paddle surfaces. Finite element analysis (FEA), implemented using ANSYS, is used to model resonant modes, stress distributions, and effective compressibility and to guide design optimization. Additively manufactured prototype paddles are fabricated, and measurements from these paddles are used to validate the models and assess the acoustic and mechanical benefits of the core design. Time-domain and frequency-domain acoustic signatures are presented, and the technical session will include demonstrations of the resulting paddle response.

Contributed Paper

2:25

3pNS5. Noise mitigation of pickleball impact sound using micro-perforated panel absorbers. Subhrodeep Ray (Temple Univ., 1947 N 12th St., Phildelphia, PA 19104, subhro.ray@temple.edu) and Haijun Liu (Temple Univ., Philadelphia, PA)

Pickleball impact noise is a prominent community-noise source with a narrow-band spectral peak near 1 kHz and strong annoyance potential. This work investigates micro-perforated panel absorbers designed and optimized to target this band while maintaining performance under oblique incidence, which is representative of outdoor court sound fields. An analytical impedance model based on Maa's theory is used to guide parameter selection (perforation diameter, panel thickness, and backing depth) for a 300–1600 Hz design range. Simulations are performed using a pressure-acoustics framework with narrow-region acoustics for perforations to capture viscothermal losses and angle-dependent behavior. To broaden effective absorption and improve angular robustness, a parallel absorber concept combining multiple perforation ratios is evaluated. Experimental validation is conducted using normal-incidence impedance-tube measurements and a larger-scale acrylic enclosure test to examine performance beyond the tube configuration. Results show that the optimized design achieves strong absorption near 1 kHz and that the parallel configuration yields a wider effective bandwidth with reduced sensitivity to incidence angle compared to a single-parameter panel. These findings support practical absorber designs for mitigating pickleball noise in community settings.

Session 3pPA

Physical Acoustics, Acoustical Oceanography, Animal Bioacoustics, and Signal Processing in Acoustics: Sensing Vector Acoustic Fields

W. C. K. Alberts, Cochair
CCDC-Army Res. Lab., 2800 Powder Mill Road, Adelphi, MD 20783

Gregory W. Lyons, Cochair
DEVCOM Army Res. Lab., 2800 Powder Mill Rd, Adelphi, MD 20783-1473

David Dall'Osto, Cochair
Appl. Phys. Lab. at the Univ. of Washington, 1959 NE Pacific Street, Seattle, WA 98195

Invited Papers

1:00

3pPA1. Collective sensing on remote, synchronized acoustic vector sensors. David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, 1959 NE Pacific St., Seattle, WA 98195, dallosto@uw.edu)

The acoustic vector field provides directional information unique to the sound emitters in the environment. Spatially diverse, time-synchronous measurements of the acoustic vector field provide unambiguous information to localize sound emitters in a complicated acoustic scene, and this distributed sensing approached is exploited in operational urban acoustic surveillance systems. In this talk, we examine two enabling technologies to sense both the vector (acoustic velocity) and scalar field (acoustic pressure) at the same location, which when combined form a measure of the acoustic intensity vector. The small footprint of these sensing devices is enabled by micro electrical mechanical systems (MEMS) sensors, one based on measurements from a tri-axial MEMS accelerometer embedded in a light-weight polystyrene sphere and the other using a combination of omnidirectional and dipole MEMS microphones. The directional information from multiple sensing nodes provides a robust means to track continuous sound emitters, and internode spatial separation and synchronous timing enables accurate triangulation/trilateration of impulsive noises.

1:20

3pPA2. Demonstration of a directional acoustic sensing float powered by ocean thermal energy conversion. Emily Barosin (Integral Consulting Inc., 200 Washington St., Ste. 201, Santa Cruz, CA 95060, ebarosin@integral-corp.com), Kaustubha Raghukumar (Integral Consulting Inc., Santa Cruz, CA), Paul Chamberlain (Scripps Inst. of Oceanogr., La Jolla, CA), Grace Chang (Integral Consulting Inc., Santa Cruz, CA), Yi Chao (Seatrec, Vista, CA), Sean Griffin (Proteus Technol., Slidell, LA), John Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA), Matthew Mazloff (Scripps Inst. of Oceanogr., La Jolla, CA), and John P. Ryan (Res., MBARI, Moss Landing, CA)

There is a paucity of acoustic data at full ocean depths, primarily due to the logistical challenges in deploying and recovering traditional acoustic sensing platforms in difficult-to-reach regions of the world's oceans. To address this challenge, a near real-time, ocean thermal energy conversion-powered, directional acoustic profiling float has been developed that can be navigated using modeled ocean currents. The float battery is recharged by an ocean thermal energy conversion (OTEC) engine and directional acoustic measurements are made using an off-the-shelf vector sensor suspended below the float that measures acoustic pressure and particle velocity in the 10 Hz–3 kHz frequency range. Other data collected include conductivity, temperature and depth, float performance metrics, and telemetered GPS positions during float surfacing. We report on the results of a field trial that was conducted offshore of San Diego, California (USA) where the float was deployed for a period of 48 h in ∼600 m deep water. We also report on preliminary results of a field trial conducted offshore of Hawaii. The float completed several dives down to 400 m at which it parked for 30 m, and acoustic pressure and velocity measurements were taken. Upon resurfacing, median spectra and directionality over 5-m data segments were telemetered to shore. Ambient noise data consisted of wind noise, passing vessels, and other anthropogenic sound sources, all of which were directionally resolved.

1:40

3pPA3. When do we need to measure acoustic particle motion for marine soundscape studies? Lessons learned using compact hydrophone arrays throughout the Western Atlantic. Ian T. Jones (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, University of New Hampshire, 8 College Rd., Morse Hall, Durham, NH 03824, ian.t.jones@unh.edu), S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, NS, Canada), Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH), T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Julien Bonnel (Scripps Inst. of Oceanogr., San DIego, CA), and Julien Flamant (Ctr. National de la Recherche Scientifique, Université de Lorraine, Nancy, France)

Marine bioacousticians are increasingly interested in measurement of acoustic vector fields (particle motion) to investigate sound sensitivity, noise impacts, and use of soundscapes by fishes and invertebrates. These animals detect sounds by transducing particle motion, and only some sense acoustic pressure. Bioacousticians studying these taxa often ask: when is a single pressure sensor sufficient to estimate particle motion versus measurement of the 3-D vector field? While measuring directionality or polarization properties requires a vector sensor, amplitude (particle motion level) can sometimes be obtained from a single hydrophone. Past and current empirical studies with compact hydrophone arrays in a variety of underwater habitats will be contrasted to answer this question. Focus regions include Caribbean coral reefs, the U.S. Mid and South Atlantic continental slope, and offshore Gulf of Maine waters. Together with recent related literature, these studies highlight that the ratio of measured particle motion amplitude to that estimated from single hydrophones depends on frequency, depth, near versus far field, time and frequency averaging, and ambient noise versus discrete sound sources. Such factors can lead to high variability of these ratios within and across datasets. We will provide recommendations for marine bioacousticians as they seek to quantify particle motion amplitudes.

2:00

3pPA4. Estimation of seabed properties at the New England Mud Patch using vector acoustic measurements. Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, gpotty@uri.edu) and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Broadband signals were measured on multiple ocean bottom recorders (OBX) deployed on the seabed during the Seabed Characterization Experiment (SBCEX 2022) in the New England Mud Patch (NEMP). The SBCEX experimental area is characterized by a seabed consisting of soft surficial layer of mud on top of sand. The objectives of the study include understanding the influence of the soft sediment layer on the acoustic propagation and to infer the properties of the mud layer using inversion techniques. OBXs measure velocities in three mutually perpendicular directions concurrently with acoustic pressure. Compressional and shear wave speeds and attenuation were estimated using acoustic pressure and particle velocity data in this study. The sediment thickness data from seismic surveys and core data were used to constrain the inversions. The sediment properties estimated by previous inversions using data from the same location were used for comparison and validation. The estimates are also compared with predictions from Viscous Grain Shearing (VGS) and the modified, corrected, Revil, extended Biot (mCREB) models. [Work supported by the Office of Naval Research, code 322 OA.]

2:20

3pPA5. Customizable acoustic signal processing framework using system-on-chip device. Aidan V. Stanford (National Ctr. for Physical Acoust., Univ. of Mississippi, 145 Hill Dr., University, MS 38677, avstanf1@go.olemiss.edu), William G. Frazier (National Ctr. for Physical Acoust., Univ. of Mississippi, Oxford, MS), and Wayne Prather (National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

Recently, a number of system-on-chip (SoC) devices have been released that combine conventional microprocessors with a field-programmable gate array (FPGA) in the same package. While these devices have had success in other scientific fields, there is still much to explore regarding their applications to acoustics. To test their efficacy in this respect, we use a Xilinx SoC device to implement a custom on-the-edge acoustic array processor. Digital signal processing routines are implemented as conventional programs running on the microprocessor, and accelerator hardware is being developed and programmed into the FPGA fabric to boost overall performance. This research presents the architecture of the array processor, details how the SoC performs in acoustic signal processing applications, and demonstrates how hybrid SoCs can be utilized by acousticians to create specialized, performance-driven frameworks for their processing needs.

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

G. Christopher Stecker, Chair
Center for Hearing Res., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131

Chair's Introduction—1:00

Invited Paper

3pPP1. Dynamic spatial sensitivity in auditory cortex. John C. Middlebrooks (Otolaryngol., UC Irvine, 18 Vega Ct, Irvine, CA 92617, j.midd@uci.edu)

Location is a conspicuous attribute of a sound source. The well-known cortical maps of space in the visual and somatosensory systems might lead one to search for a map of sound-source locations in the auditory cortex. That turns out to be a fool's errand. The best efforts of our group and of others have failed to discover a point-to-point map of space in the auditory cortex, although such maps are found in the midbrain. Instead, we find that single neurons in the auditory cortex are broadly tuned for source location, with location-specific firing patterns that represent sound-source locations panoramically. Moreover, the spatial tuning of neurons is dynamic, sharpening when an animal is engaged in a localization task or, even more so, when competing sound streams are present. Despite the 180-to-360-degree spatial receptive fields of cortical neurons, a spatial stream segregation task demonstrated quantitatively similar ~5–10 deg spatial resolution among human and cat psychophysics and cat cortical neuron responses. Our early failure to find a cortical space map reflects a feature, not a defect. We now appreciate that the panoramic and dynamic properties of cortical neurons are well adapted to the demands of auditory spatial perception.

Session 3pSA

Structural Acoustics and Vibration: Structural Acoustics and Vibrations Tutorial

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

Peter Kerrian, Cochair
ATA Eng. Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128

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

Chair's Introduction—1:00

Invited Paper

3pSA1. Experimental modal analysis tutorial. Kevin Napolitano (ATA Eng. Inc., 13290 Evening Creek Dr. South, San Diego, CA 92128, kevin.napolitano@ata-e.com) and Peter Kerrian (ATA Eng. Inc., San Diego, CA)

This 2-h session will provide an overview of experimental modal analysis (EMA) emphasizing the use of frequency response functions applied to aerospace engineering applications. After a brief overview of EMA concepts using simple lumped parameter models, this presentation will provide an in-depth review of the steps required in a typical aerospace level modal survey test including test planning, test setup, data collection, signal processing, mode shape extraction, and validation. Finally, recent advances in modal survey testing using acoustic sources as exciters will be discussed.

Session 3pSP

Signal Processing in Acoustics and Signal Processing in Acoustics: Convex Optimization and Sparsity in Signal Processing

Yongsung Park, Cochair
Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543-1050

Youngmin Choo, Cochair
Seoul National Univ., 1 Gwanak-ro, Gwanak-gu, Seoul, 08826, Korea

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

Chair's Introduction—1:00

Invited Papers

1:05

3pSP1. Advances in sparse beamforming: From gridless to sparse Bayesian processing. Yongsung Park (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543-1050, yongsung.park@whoi.edu)

Sparse signal processing has revolutionized direction-of-arrival (DOA) estimation by overcoming the Rayleigh resolution limit. Unlike conventional beamforming techniques, sparsity-driven formulations explicitly exploit the sparse representation of array signals, enabling super-resolution and robustness under limited data conditions. This talk provides an overview of sparsity-driven beamforming methods developed over the past decade. We review the evolution from grid-based l1-regularized beamforming to gridless approaches based on atomic norms, alternating projection, and Wirtinger gradient methods, highlighting their theoretical foundations and practical implications. Extensions to multi-frequency, wideband, and difference-frequency processing are discussed. Furthermore, we highlight probabilistic approaches, such as sequential sparse Bayesian methods, for tracking time-varying DOAs. Representative applications in underwater acoustics demonstrate how sparse optimization enables enhanced localization performance under limited snapshots, array configurations, and complex propagation environments. This overview aims to connect advances in convex optimization and sparsity with emerging challenges in modern beamforming and array processing for ocean acoustic applications.

1:25

3pSP2. System-informed neural networks for sonar signal processing: Frequency analysis and DOA estimation. Youngmin Choo (Seoul National Univ., 1 Gwanak-ro, Gwanak-gu, Seoul 08826, Korea, sonacer@snu.ac.kr)

We propose a system-informed neural network (SINN) framework designed to solve linear inverse problems. The SINN architecture is based on an adaptively learned iterative soft-thresholding algorithm and explicitly incorporates the underlying system model into the loss function. Rather than learning a direct mapping from inputs to outputs, SINN estimates solutions by solving the associated linear systems. In this work, the SINN framework is applied separately to frequency analysis and direction-of-arrival (DOA) estimation, resulting in two task-specific SINN-based approaches. For frequency analysis, SINN is combined with a deep ensemble strategy to further refine detected frequency components, while for DOA estimation, SINN is used in conjunction with an auxiliary network that predicts the presence of arrivals across different directions, such as azimuthal or elevation angles. The proposed SINN-based approaches are evaluated using both simulated data and real measurements from the SWellEx-96 experiment, and the results demonstrate clearer frequency components and more accurate DOA estimates, while enabling real-time processing due to the offline training phase.

1:45

3pSP3. Passive localization and ambiguity resolution in sparse linear arrays via factor graph optimization. Paige Pfenninger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 32 Vassar St., Marine Robotics Group, Cambridge, MA 02139, ppfenni@mit.edu), Jungseok Hong, and John Leonard (MIT, Cambridge, MA)

Studies were conducted to investigate the passive localization of multiple moving sources using a sparse 8-element linear hydrophone array with a 75-foot inter-element spacing. While large-aperture arrays offer high angular resolution for low-frequency sources, they are fundamentally limited by spatial aliasing and numerous grating lobes. To deal with spatial ambiguities, frequency-difference beamforming is utilized to synthesize a low-frequency "autoproduct" field where the array is effectively Nyquist-sampled. This work investigates a framework for further resolving ambiguities by treating the localization problem as a global trajectory optimization rather than a series of independent snapshot estimates. By representing the directional measurements as multi-modal likelihoods within a factor

graph, kinematic constraints of moving sources are incorporated to distinguish true trajectories from aliased artifacts. Experimental data containing multiple moving and stationary sound sources were analyzed to evaluate the robustness of the tracking model in a shallow water environment. Multiple sources including micromodems, single high-frequency pingers, and motor noises are analyzed. Theoretical limits of array gain and localization accuracy are explored in the presence of high sidelobe levels. Preliminary numerical simulations suggest that while frequency-difference processing suppresses spatial ghosts, the performance of the localization model may be sensitive to the signal-to-noise ratio of the out-of-band frequencies.

2:05–2:20 Break

2:20

3pSP4. Geoacoustic inversion with the Wasserstein distance: Imposing convexity on the inverse problem. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu) and Christina Frederick (Dept. of Mathematical Sci., NJIT, Newark, NJ)

Geoacoustic inversion often relies on matching signals received at arrays of hydrophones with time series computed using sound propagation models. These synthetic signals, commonly referred to as replicas, are generated for varying values of environmental parameters. Identifying the optimal parameter values is achieved by finding the best agreement between the measured and synthetic time series. This is typically done by minimizing the mismatch between replicas and observed data. A common metric used to measure this mismatch is the Euclidean distance. A major challenge in inversion is the presence of local minima in the objective functions, which increases computational cost, complicates multidimensional optimization, and hinders the identification of the global minimum. Moreover, replica and true time-series are rarely aligned, with the beginning of the true signal being difficult to identify. The resulting shifts in time cause obstacles in inversion with the Euclidean distance. Optimal transport offers an alternative framework by employing different similarity measures that lead to objective functions with more favorable optimization properties, e.g., it has been shown that optimal transport-based metrics are convex with respect to shifts in time. In this study, we investigate the application of optimal transport to geoacoustic inversion and evaluate its performance across several sound propagation scenarios, especially in the case of misalignment of replica and true time-series. [Work supported by ONR.]

2:40

3pSP5. Braided multipath channel representations and locally constrained sparse optimization in shallow water acoustic communications. Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

The shallow water acoustic channel is well-known to exhibit rapidly fluctuating delay spread due to time-varying multipath arrivals. The channel support maybe sparse but the degree of sparsity can vary substantially as a function of changing unpredictable oceanic conditions, e.g., surface wave focusing events. This makes shallow water acoustic channel tracking an interesting sparse optimization challenge. Popular convex optimization methods to track this time-varying channel include but, are not limited to, classical least square error minimization techniques, as well as sparse optimization methods such as basis pursuit and mixed norm solver techniques. The talk, with recent interpretations on prior work with visions of how to move forward, will highlight trade-offs between some of these well-known techniques in terms of their channel estimation accuracy, as well as their agility to follow unpredictable changes in the channel delay spread. We will also provide an overview of different channel representations using braid manifolds, and highlight how different braid representations can capture different physical and statistical elements of the shallow water acoustic channel multipath. In particular, we will discuss, based on Bellhop simulations and SPACE08 experimental field data, how braided optimization can employ locally constrained gradient descent approaches for tracking and predicting multipath eigenray bundles.

3:00

3pSP6. Underwater acoustic recovery based on Riemannian gradient descent tensor completion. Poline Xian (Comput. Sci., Hong Kong Baptist Univ., 224 Waterloo Rd., Kowloon Tong, Hong Kong, Hong Kong 999077, Hong Kong, poline3939@gmail.com)

When the signal propagates in a multipath underwater acoustical environment, the Doppler effect affects signal detection and estimation. Current existing algorithms use the low-rank and sparse matrix completion approaches. These methods have high computational complexity and can cause a loss of phase, propagation, and channel information. They will fail to provide the most accurate estimation. In this work, we propose the Riemannian gradient descent method for tensor completion in underwater acoustic recovery. We construct a data tensor that incorporates propagated signals. The algorithm performs the higher-order singular value decomposition on the signal. It recovers the signals with subspace projection based on the data geometry. Experimental results show that compared with several traditional methods, the proposed tensor recovery method is faster, more robust, and accurate.

Session 3pUW

Underwater Acoustics: Air–Sea Interaction Acoustics

Karim G. Sabra, Cochair
Mech. Eng., Georgia Inst. of Technol., 771 Ferst Drive, NW, Atlanta, GA 30332-0405

Isabelle Brandicourt, Cochair
Oceanogr., Univ. of Washington, 1501 NE Boat St., #206, Seattle, WA 98195

Contributed Papers

1:00

3pUW1. Monitoring ocean rainfall with ambient acoustics: Improving robustness across recording locations. Charles J. Berg (Naval Undersea Warfare Ctr. Newport, 1176 Howell St., Newport, RI 02841, charles.j.berg12.civ@us.navy.mil) and Vijay Peddinti (Naval Undersea Warfare Ctr. Newport, Newport, RI)

Machine learning algorithms can estimate rain rates from acoustic power spectral densities by exploiting the broadband underwater sound produced by rainfall. While these methods show promise, their validation has typically been limited to the site used in training, leaving generalization to an unseen location unverified. To address this uncertainty, this study evaluates the robustness of a broadband support vector machine (SVM) rain detector using 12 months of acoustic and weather data from two Ocean Observatories Initiative (OOI) recording stations off the coast of Oregon, USA. When the detector was trained exclusively at one site, its performance degraded at the second, untrained site due to systematic shifts in mean spectral levels across all frequencies. This work proposes a robust spectral estimation procedure that mitigates spectral level variation caused by interfering sounds, propagation effects, and hydrophone calibration errors. Applying this spectral estimation procedure dramatically improves the SVM rain detector's performance at the untrained location. This finding suggests that to effectively use ambient sound for ocean rainfall monitoring across different sites, it is crucial to address inconsistencies in the spectral levels caused by environmental variation and measurement error.

1:15

3pUW2. Rainstorm acoustic scenes in the Gulf of Maine. Matthew C. Bisson (Univ. of New Hampshire, 105 Main St., Durham, NH 03824, matt.bisson@unh.edu) and Jennifer Miksis-Olds (Univ. of New Hampshire, Durham, NH)

Soundscapes are the combination of sound sources recorded at a single location characterized by amplitude, periodicity, impulsiveness, and uniformity. In the terrestrial environment, auditory scenes are acoustic environments or phenomena that can be identified by combinations of key acoustic metrics agnostic to time or geographical location. This concept can be applied to the underwater environment to identify underwater acoustic scenes. By using soundscape analysis, unique combinations of acoustic metrics can be linked to acoustic scenes in the underwater environment. A preliminary study analyzed rainstorms as an underwater acoustic scene. Acoustic data from several different rainstorm events were analyzed from the Gulf of Maine. Acoustic scene matrices for each rainstorm event were created and statistical techniques were used to determine the unique combination of metrics that represented the acoustic scene for rainstorms. The combination of metrics determined to represent rainstorms in the Gulf of Maine was then applied to rainstorm events from other geographic regions to assess the transferability of the rainstorm acoustic scene developed from Gulf of Maine data to rainstorms at large. Underwater acoustic scenes can be created using this framework for many different acoustic states in environments. Defining the unique combination of acoustic metrics for individual underwater acoustic scenes will enable machine learning algorithms to support real time, autonomous detection or identification of underwater acoustic scenes.

1:30

3pUW3. Underwater wind noise model based on bubble clouds and comparisons to observations from Monterey Bay. Tarun K Chandrayadula (Ocean Eng., Indian Inst. of Technol. Madras, Dept. of Ocean Eng., Indian Inst. of Technol., Madras, Chennai, Tamil Nadu 600036, India, tkchandr@iitm.ac.in) and Ranga P. Raju (Ocean Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu, India)

The underwater noise at frequencies 0.5–20 kHz is generally due to the bubbles caused by wind. The bubbles occur in huge congregations called plumes. The current literature contains models for plumes that give the number of bubbles in them, the sizes, and their respective distributions across depth. There are no models however that use these models to predict the noise-levels. This paper considers two competing theories for noises from plumes. One is the individual bubble oscillation theory, where each bubble is an entrapment of air surrounded by water. The bubbles radiate a spectrum of frequencies depending on their sizes. Multiple number of such individual bubbles are simulated using models for plumes, and the resultant is calculated. The collective bubble oscillation on the other hand models the individual bubbles, but also includes the effect of the air in the rest of the bubble-plume. The air in the other bubbles absorb sounds that are radiated, and also lowers the resonant frequencies in comparison to individual bubbles. This paper uses both these models, and compares them against deep-water hydrophone recordings off Monterey Bay in California. The two models seem to account for noises in different regions of the frequency spectrum.

1:45

3pUW4. Estimating flow speeds using the coherence of co-located hydrophones. Robert T. Taylor (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, rtaylor119@utexas.edu), Megan Ballard (Univ. of Texas at Austin, Austin, TX), Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Jim Thomson, Kristin Zeiden (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), J. T. Farrar (Woods Hole Oceanographic Inst., Woods Hole, MA), Leah Johnson, and Harper Simmons (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Studies exploring hydrophone flow noise have produced wide ranging relationships between flow speed and the recorded ambient acoustic spectra. This is a natural result of the complexity of the problem, where factors including transducer size, shape, and mounting configuration, impact the surrounding boundary layer as it transitions from laminar to fully turbulent. At slow speeds, where flow is primarily laminar over the hydrophone, the turbulent signal is often masked by ambient sound. The present

investigation pertains to the flow-noise/flow-speed relationship at speeds of less than 50 cm/s. A year-long acoustic collection was made with a moored 52-element vertical line array that spanned the majority of the 420 m water column on a ridge in the central Nordic Seas. Flow noise is present despite hydrophones being equipped with flow shields. The coherence of the acoustic signal measured on adjacent hydrophones is used to distinguish the ambient sound from the pseudo sound of the flow noise, enabling direct estimates at speeds less than 10 cm/s even under high ambient noise conditions. Independent measurements of local flow speed were provided by an environmental mooring equipped with multiple acoustic Doppler current profilers, located approximately 1 km from the acoustic array.

2:00

3pUW5. Passive acoustic estimation of surface wave spectra in the ice-free Arctic Ocean. Brian J. Granger (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Appl. Res. Lab. UT, 10000 Burnet Rd., Austin, TX 78758, brian.granger@utexas.edu), Robert T. Taylor, Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kristin Zeiden, and Jim Thomson (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105)

The ice-free Arctic Ocean is subject to strong, highly variable winds that alter the ambient acoustic environment and produce a rapidly evolving sea surface. Additionally, extremely cold air temperatures much of the year contribute to upward-refracting sound-speed profiles that increase acoustic signal interaction with the sea surface. Accurate sound propagation modeling therefore requires a detailed understanding of sound–surface interactions. Knowledge of surface wave spectra enables estimation of surface roughness, a key parameter for acoustic propagation modeling; however, instrumenting this environment, particularly with surface-based sensors, is exceptionally challenging. This study uses passive, underwater acoustic sensors to relate ambient sound, wind speed, and surface wave spectra. Acoustic data from a moored vertical line array deployed in the ice-free Arctic near Jan Mayen Island from 2022–2023 were used to estimate wind speed and wave spectra. These surface spectra estimates were compared to independent measurements and estimates to ascertain the viability of the passive approach. Comparisons include data from surface buoys, moored upward-looking acoustic Doppler current profilers, and surface spectra from the European Reanalysis Fifth Generation (ERA5) database. This work highlights the potential for improved characterization of ocean surface conditions and sound propagation modeling based solely on passive acoustic measurements. [Work sponsored by ONR.]

2:15

3pUW6. Acoustic binary classification of ice cover conditions using deep learning. Leora D. Robinson (Phys. and Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, leorarobinson13@gmail.com) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Due to climate change, the amount of ice cover in polar regions has become more variable. Because ice cover impacts sound propagation, proper sonar performance depends on an understanding of the presence or absence of ice. This project investigates the possibility of using deep learning to make this distinction. In preparation for ocean testing, initial measurements were collected in a laboratory tank. The experiment used a linear chirp from 10–50 kHz recorded at variable source and receiver positions in the tank. Approximately 150 L of pebble ice were added to the water tank, and sound was recorded before ice was added, while it was present, and after it melted. From each 1-s signal, the spectrogram was calculated and labeled based on the presence of ice. These labeled data were used to train a convolutional neural network (CNN) for binary classification for the presence or absence of ice. The trained CNN was applied to data collected on different days with different ambient noise conditions. The most successful CNN reached 81% accuracy on the unseen data. These results indicate the potential for deep learning to identify ice coverage in naval applications.

Plenary Session and Awards Ceremony

Michael Vorländer
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Geoffrey Edelmann – For contributions to underwater communications and compressive sensing

Kenneth W. Good, Jr. – For service to the Society and contributions to improving speech privacy

Tatiana D. Khokhlova – For pioneering shock-based boiling histotripsy

Kang Kim – For contributions to elasticity imaging, ultrasound thermal strain imaging, and photoacoustic imaging

Brian B. Monson – For contributions to extended high frequency hearing in humans

Matthew B. Winn – For contributions to the understanding of speech perception and listening effort

Likun Zhang – For contributions to the understanding of acoustic radiation forces in complex wave fields and vortex beams

Presentation of Science Communication Awards

A Cathedral of Sound by Madeleine Schwartz, Malika Khurana, Mika Gröndahl and Yuliya Parshina-Kottas (New York Times Magazine online, 3 March 2023)

The Science of Sound, a two-part episode in the *"Where What if Becomes What's Next" Podcast* Laurie Heller, Podcast Presenter (Carnegie Mellon University, 17 and 31 July, 2025)

Pistols in St Paul's: Science, Music, and Architecture in the Twentieth Century by Fiona Smyth (Manchester University Press, September 2024)

The Sonic Soul of Notre Dame by David Levin (Brandeis Magazine, Summer 2025)

Introduction of Prize Recipients

2025 Rossing Prize in Acoustics Education to Daniel A. Russell

2026 Hartmann Prize in Auditory Neuroscience to John C. Middlebrooks

2026 Medwin Prize in Acoustical Oceanography to Shima Abadi

The Prize was presented at the Monday Medwin Prize in Acoustical Oceanography Lecture

Presentation of Awards

R. Bruce Lindsay Award to Whitney L. Coyle

Trent-Crede Medal to Andrew N. Norris

Helmholtz-Rayleigh Interdisciplinary Silver Medal in Physical Acoustics and Biomedical Acoustics to Oleg Sapozhnikov

Honorary Fellowship to Judy R. Dubno

Gold Medal to Anthony A. Atchley

R. BRUCE LINDSAY AWARD

Whitney L. Coyle

2026

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is no more than 10 years post terminal degree on 1 July at the time of Award acceptance and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942	Beverly A. Wright	1995
Leo L. Beranek	1944	Victor W. Sparrow	1996
Vincent Salmon	1946	D. Keith Wilson	1997
Isadore Rudnick	1948	Robert L. Clark	1998
J. C. R. Licklider	1950	Paul E. Barbone	1999
Osman K. Mawardi	1952	Robin O. Cleveland	2000
Uno Ingard	1954	Andrew J. Oxenham	2001
Ernest Yeager	1956	James J. Finneran	2002
Ira J. Hirsh	1956	Thomas J. Royston	2002
Bruce P. Bogert	1958	Dani Byrd	2003
Ira Dyer	1960	Michael R. Bailey	2004
Alan Powell	1962	Lily M. Wang	2005
Tony F. W. Embleton	1964	Purnima Ratilal	2006
David M. Green	1966	Dorian S. Houser	2007
Emmanuel P. Papadakis	1968	Tyrone M. Porter	2008
Logan E. Hargrove	1970	Kelly J. Benoit-Bird	2009
Robert D. Finch	1972	Kent L. Gee	2010
Lawrence R. Rabiner	1974	Karim G. Sabra	2011
Robert E. Apfel	1976	Constantin-C. Coussios	2012
Henry E. Bass	1978	Eleanor P. J. Stride	2013
Peter H. Rogers	1980	Matthew J. Goupell	2014
Ralph N. Baer	1982	Matthew W. Urban	2015
Peter N. Mikhalevsky	1984	Megan S. Ballard	2016
William E. Cooper	1986	Bradley E. Treeby	2017
Ilene J. Busch-Vishniac	1987	Yun Jing	2018
Gilles A. Daigle	1988	Adam Maxwell	2019
Mark F. Hamilton	1989	Julien Bonnel	2020
Thomas J. Hofler	1990	Likun Zhang	2021
Yves H. Berthelot	1991	Meaghan A. O'Reilly	2022
Joseph M. Cuschieri	1991	Julianna C. Simon	2023
Anthony A. Atchley	1992	Christopher Kube	2024
Michael D. Collins	1993	Andrea P. Arguelles	2025
Robert P. Carlyon	1994		



ENCOMIUM FOR WHITNEY L. COYLE

… for contributions to understanding the acoustics of the clarinet and for service to the Society

13 MAY 2026 • PHILADELPHIA, PENNSYLVANIA

A native of Kentucky with an undergraduate degree from Murray State University, Whitney Coyle is currently chair of the physics department and an associate professor of physics at Rollins College in Winter Park, Florida. She was introduced to the field of acoustics during a National Research Foundation Research Experience for Undergraduates with James Cottingham at Coe College while she was undergraduate studying music and mathematics. These complementary interests led her naturally into the field of musical acoustics. After completing her undergraduate degree Whitney pursued a Ph.D. at Penn State, originally working with Victor Sparrow on outdoor sound propagation before beginning her research in musical acoustics.

Dr. Coyle's doctoral research was funded by the NSF Graduate Research Fellowship Program and concentrated on the physics of the clarinet. While in graduate school she was also named a Graduate Research Opportunities Worldwide Fellow by the NSF, as well as a Chateaubriand Fellow by the French embassy. These two fellowships enabled her to spend the majority of the last three years of her graduate studies with Jean Kergomard at the Laboratoire de Mécanique et d'Acoustique in Marseille, France. Her work in Marseille on predicting the playing frequencies of clarinets is frequently cited and was an important contribution to understanding the physics involved in the unique sound of the instrument.

After her graduate work Dr. Coyle joined the faculty at Rollins College, an undergraduate liberal arts college that emphasizes teaching above all other priorities. Although she was required to teach six undergraduate courses each academic year, with no teaching assistants or graders to ease the workload, Dr. Coyle still managed to establish a student-centered research effort involving undergraduate students in both experimental and computational acoustics. Her students have made more than 20 presentations at ASA meetings in the last decade and are coauthors on several peer-reviewed articles. Her research on the acoustics of musical instruments continues, with her undergraduate students playing key roles in published work on the analysis of clarinet tonguing techniques, mouthpiece analysis, and flow dynamics. Recently she and her students have also collaborated with Kent Gee at Brigham Young University to study the acoustics of rocket launch vehicles from the Kennedy Space Center.

In addition to her research and teaching Whitney is a long-time member and active supporter of the Acoustical Society of America (ASA). She has been a member of the ASA for almost two decades, beginning when she was still an undergraduate student. During her graduate studies she maintained her membership in ASA and was an active member of the Student Council, serving as the student representative from the Technical Committee on Musical Acoustics, and chairing the Student Council from 2012 to 2014. Since receiving her Ph.D. Dr. Coyle has continued to support the ASA, serving on numerous committees including Publication Policy, Women in Acoustics, Education in Acoustics, Public Relations, the Technical Committee on Musical Acoustics, and chairing the Web Advisory Committee. She has also regularly presented her work at ASA meetings where she was awarded two Best Student Paper Awards in 2009 and 2014 and Best Young Presenter in Noise in 2013.

Dr. Coyle continues to teach, mentor undergraduate students, maintain a vibrant research program and actively support the ASA while also raising two beautiful daughters and running the occasional marathon.

THOMAS MOORE

ACOUSTICAL SOCIETY OF AMERICA

TRENT-CREDE MEDAL



Andrew N. Norris

2025

The Trent-Crede Medal is presented to an individual, irrespective of nationality, age, or society affi liation, who has made an outstanding contribution to the science of mechanical vibration and shock, as evidenced by publication of research results in professional journals or by other accomplishments in the field.

PREVIOUS RECIPIENTS

Carl Irwin Vigness	1969	Gideon Maidanik	1991
Raymond D. Mindlin	1971	Preston W. Smith, Jr.	1996
Elias Klein	1973	David Feit	1999
J. P. Den Hartog	1975	Sabih I. Hayek	2003
Stephen H. Crandall	1978	Jerry H. Ginsberg	2005
John C. Snowdon	1980	Peter R. Stepanishen	2011
Eric E. Ungar	1983	Earl G. Williams	2016
Miguel C. Junger	1987		



ANDREW N. NORRIS

…for foundational contributions to the theory of acoustic and elastic metamaterials and to wave propagation in complex media and structures

13 MAY 2026 • PHILADELPHIA, PENNSYLVANIA

Andrew (Andy) N. Norris was born and raised in Dublin, Ireland. The youngest of seven children, he attended Christian Brothers schools from elementary through high school. Andy excelled at mathematics and physics in high school, making the Mathematical Physics program at University College Dublin (UCD) a natural fit for his undergraduate studies. He finished his bachelor's degree in 1977, receiving First Class Honors, and continued at UCD to complete a Master of Science (MS) degree one year later. It was during that time that he was first formerly introduced to wave mechanics by Michael Hayes. These formative studies clearly had a significant influence on Andy's career, which has spanned numerous topics germane to the field of acoustics, including ultrasonic non-destructive testing (NDT), waves in porous, granular, and composite materials, nonlinear acoustics, acoustic wave scattering from structures, and acoustic metamaterials.

After finishing his MS, he chose to pursue his Ph.D. in the United States at Northwestern University under the supervision of Jan D. Achenbach who had just recently published his classic textbook *Wave Propagation in Elastic Solids*. Those were the early days of ultrasonic NDT and Andy made key contributions to the field soon after his arrival, authoring his first paper on ultrasonic wave scattering from cracks in in the first volume of the journal *Wave Motion* in 1979. Thirty-two years later, Andy became the editor-in-chief of that journal. Andy's doctoral research focused primarily on the topic of scattering from cracks and scattered field inversion, interacting with many that laid the foundations of ultrasonic NDT, including Bruce and David Thompson and Lazlo Adler. Andy truly had a hand in putting the "Q" (for "Quantitative") in QNDT, as Achenbach was known to say.

After graduate school, Andy took a postdoctoral position at Exxon where his work focused on seismic waves in poroelastic media and Gaussian beam summation to calculate ray propagation and complex point sources. This included research with Nobel laureate John Robert Schrieffer on the topic of Gaussian wave packet propagation which likely informed Andy's numerous contributions to high frequency asymptotics for scattering problems. His research on acoustic scattering included collaborations with Gregory Kriegsmann and Edward Reiss to describe scattering by membranes and later extensive contributions on scattering and diffraction of waterborne acoustic waves and flexural waves in beams and plates. An example of Andy's unique ability to adapt mathematical tools developed for one physical domain to address more general problems was his use of Gaussian beam summation to find an exact representation of complex point sources for waves propagating in arbitrary inhomogeneous media [J. Opt. Soc. Am. 3(2), 2005–2010, (1986)].

Andy took a faculty position at Rutgers University in 1985 where he quickly expanded his research interests to include effective medium theory, borehole acoustics, nonlinear acoustics, and structural acoustics. Early in his career at Rutgers, he focused on topics relevant to seismic waves in collaboration with Robert Burridge, David Johnson, Bikash Sinha, and Sergio Kostek. A representative example was his derivation of integro-differential delay equations for waves in finely layered media that described the non-intuitive behavior of weak scattering by the numerous interfaces of finely layered media yielding a leading order scattering effect [J. Acoust. Soc. Am. 94(5), 2884–2894 (1993)]. He also re-cast the model equations of poroelasticity in a form that is easier to interpret and studied wave propagation in anisotropic and nonlinear media. A notable contribution to the field of nonlinear acoustics was his concise yet authoritative chapter on *Finite-Amplitude Waves in Solids* in Blackstock and Hamilton's textbook on *Nonlinear Acoustics*.

Andy also studied numerous structural acoustics problems throughout the 1990's, work that was influenced by David Feit and Preston Smith, Jr. in the US and Gerry Wickham and David Crighton in the UK. He explored the use of ray theory to study underwater scattering, diffraction, and radiation problems for fluid-loaded structures and developed numerous analytical models describing flexural wave propagation and scattering in beams and plates. Taken together, Andy's papers on these topics are must-reads for anyone wishing to understand the intricacies of sound scattering from elastic structures.

The extensive research summarized above provided the foundation and insights that made Andy uniquely suited to make foundational contributions to the field of acoustic metamaterials, for which he is now well known. Arguably his most prominent contribution to this field is *Acoustic Cloaking Theory* [Proc. R. Soc. A, 464, 2411-2434, (2008)]. That work provided an authoritative description of the spatially varying elastic properties required to achieve acoustic cloaking. Prior to this paper, researchers had concluded that the acoustic analogue to electromagnetic cloaking was impossible using conventional elastic materials without requiring infinite mass, which Andy aptly described as "a massive problem." His work paved the way for the design of exotic acoustic materials for acoustic cloaks and lenses using transformation acoustics and provided the impetus to create pentamode metamaterials with tailorable anisotropic sound speeds. He later fabricated these materials and experimentally validated their behavior in an underwater environment [J. Acoust. Soc. Am., 151(1), 168-179, (2022)]. Additional contributions include dynamic homogenization models of metamaterials with subwavelength asymmetry that lead to direction-dependent characteristic acoustic impedance and novel wave scattering behavior [Proc. R. Soc. A, 468(2142), 1629–1651 (2012)]. He has also made numerous recent contributions to the topic of engineered materials and dynamic structures which enable non-reciprocal acoustic and elastic wave propagation [J. Mech. Phys. Sol. 101, 10-29, (2017)].

Andy's substantial contributions to acoustics, spanning over four decades are a result of his deep and seemingly effortless understanding of mathematical physics. His unique ability to bring knowledge from multiple topics together using clear and incisive writing has led him to make foundational and long-lasting contributions across acoustics, most recently in acoustic metamaterials as recognized by this medal. However, simply focusing on his professional contributions misses his most notable attributes, which are his demeanor and character. Andy is a wonderful colleague who is a pleasure to work with and who makes everyone around him better. He routinely works with a wide range of collaborators and elevates junior colleagues, ensuring a legacy that will last for many decades to come.

MICHAEL HABERMAN
CHRISTOPHER KUBE

Helmholtz-Rayleigh Interdisciplinary Silver Medal
in
Physical Acoustics and Biomedical Acoustics



Oleg Sapozhnikov

2026

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997	Timothy J. Leighton	2013
David E. Weston	1998	Mark F. Hamilton	2014
Jens P. Blauert	1999	Henry Cox	2015
Lawrence A. Crum	2000	Armen Sarvazyan	2016
William M. Hartmann	2001	Blake S. Wilson	2017
Arthur B. Baggeroer	2002	Kenneth S. Suslick	2018
David Lubman	2004	Barbara G. Shinn-Cunningham	2019
Gilles A. Daigle	2005	Michael R. Moldover	2021
Mathias Fink	2006	George L. Augspurger	2022
Edwin L. Carstensen	2007	Vera A. Khokhlova	2023
James V. Candy	2008	D. Keith Wilson	2024
Ronald A. Roy	2010	N. Ross Chapman	2025
James E. Barger	2011		

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993



ENCOMIUM FOR OLEG A. SAPOZHNIKOV

…for contributions to nonlinear acoustics and its applications to medical ultrasound

13 MAY 2026 • PHILADELPHIA, PENNSYLVANIA

Oleg Anatolievich Sapozhnikov was born in Batyrevo, a village with several thousand people in the Chuvash Republic of the USSR (now Russia), approximately 500 miles east of Moscow. His father was a veterinarian and his mother taught history at a local high school.

After being awarded a Gold Medal when he graduated from the high school in Batyrevo in 1979, Oleg entered the Physics Faculty of Lomonosov Moscow State University (MSU). He studied mathematics and physics at a high level and soon became one of the best students in his class, and the go-to student to answer questions raised by his classmates.

While a student at MSU Oleg joined a research group that specialized in nonlinear acoustics, in 1982. Leading this group were Professors Oleg Rudenko, an accomplished theoretician, and Alexander Karabutov, a talented experimentalist. Nonlinear acoustics became the primary focus of Oleg's scientific activities ever since. The research group was in the department specializing in laser physics and nonlinear optics, and this exposed Oleg to advanced research and education in acoustics, optics, and photoacoustics. Vera Khokhlova started at MSU the same year as Oleg Sapozhnikov, and they joined Oleg Rudenko's group simultaneously. This was the beginning of their lifelong collaboration that included their formation of the Laboratory for Industrial and Medical Ultrasound (LIMU) at MSU in 2014.

Oleg received his MSc degree from MSU in 1985 and defended his PhD thesis in 1988. Since then he has served as a faculty member in the Department of Acoustics within the Physics Faculty of MSU. He defended his Doctor of Science thesis in 2008, which described his research on characterization of ultrasound sources, nonlinear self-action phenomena, and physical mechanisms associated with the use of high-power ultrasound and shock waves in lithotripsy. Shortly thereafter he became a full professor in the Physics Faculty of MSU.

In 1994 Oleg spent a year as a visiting researcher in Lyon, France at the National Institute of Health and Medical Research (INSERM) working with Dominique Cathignol and Jean-Yves Chapelon. Oleg's first trip to the United States was in spring 1996 to visit the University of Texas at Austin, a leading center for nonlinear acoustics. He returned to the U.S. in summer 1996 to begin collaborating with the Center for Industrial and Medical Ultrasound (CIMU) that Larry Crum created in the Applied Physics Laboratory (APL) at the University of Washington (UW) in Seattle, a partnership that continues to this day. Both Oleg and Vera became essential participants in the CIMU group, performing joint experiments, supervising graduate students, and arranging mutual research projects and visits between Moscow and Seattle.

Oleg is a leading and highly cited international expert on physical acoustics, and in particular, on nonlinear acoustics used in biomedical applications. He combined advanced fundamental research performed throughout his academic career as a Professor of Acoustics at MSU with his studies at APL-UW, renowned for its strong translational research in biomedical ultrasound. This combination, together with Oleg's engaging personality and mentorship abilities, led to the transition of his fundamental ideas into patented technologies and ultimately new medical devices.

Oleg is well known for bringing thoughtful analogies from different fields of physics to ultrasound, which often lead to simple explanations of sometimes very complicated phenomena. While he is an exceptional theoretician, he has also conceived and led many experimental projects in physical acoustics, often in nonlinear and biomedical acoustics. His most notable research contributions range from purely theoretical work on group analysis of nonlinear equations in acoustics, to modeling acoustic radiation force, performing experimental studies on acoustical holography, identifying the physical mechanisms of kidney stone breakage in shock-wave and burst-wave lithotripsy, tissue liquefaction in

boiling histotripsy, and interpreting the twinkling artifact in ultrasound Doppler imaging of kidney stones. The acoustical holography method that Oleg developed and introduced in early 2000s to the medical community is now considered to be a critical step in the characterization of ultrasound medical transducers and their fields. The idea of using pulsed shear waves propagating transverse to the push beam that he demonstrated experimentally in 1996 has evolved to become a powerful tool in diagnostic shear wave elasticity imaging, for nondestructive evaluation, and other applications.

Oleg has been an active member of the ASA for many years. He is an ASA Fellow, served as an Associate Editor for *the Journal of the Acoustical Society of America* (JASA), was a member of the ASA Committee on International Research and Education, and has organized special sessions at ASA meetings. Oleg is also a Member of the Governing Board of the International Congress on Ultrasonics (ICU), a Board Member of the Russian Acoustical Society, and a Member of the Scientific Council on Acoustics in the Russian Academy of Sciences, heading the Physical Ultrasound section. In 2004 he was one of the first two recipients of the Early Career Award from the International Commission for Acoustics (ICA), and in spring 2025 he gave a plenary lecture at the joint ICA/ASA meeting in New Orleans.

In conclusion, Oleg Sapozhnikov is an outstanding physicist and acoustician with very broad international recognition. He has contributed significantly to building a foundation for modern physical acoustics and its biomedical diagnostic and therapeutic applications. He is therefore a highly deserving recipient of the Helmholtz-Rayleigh Interdisciplinary Silver Medal in Physical Acoustics and Biomedical Acoustics of the Acoustical Society of America.

MARK F. HAMILTON
MATHIAS FINK
VERA A. KHOKHLOVA
LAWRENCE A. CRUM

HONORARY FELLOWSHIP
in the
ACOUSTICAL SOCIETY OF AMERICA



Judy R. Dubno

2026

An individual who has attained eminence in acoustics or who has rendered outstanding service to acoustics may be elected to Honorary Fellowship in the Acoustical Society of America.

HONORARY FELLOWS

Thomas A. Edison	1929	Henrik A. S. Nødtvedt	1988
Harvey Fletcher	1949	Leo L. Beranek	1994
Vern O. Knudsen	1954	Robert W. Young	1997
Paul E. Sabine	1954	Carleen M. Hutchins	1998
Frederick A. Saunders	1954	Leonid M. Brekhovskikh	1999
Floyd R. Watson	1954	Michael Longuet-Higgins	2002
Harvey C. Hayes	1960	Walter Munk	2004
Walter G. Cady	1971	Amar G. Bose	2011
Elfyn J. Richards	1980	Dah You Maa	2012
Maurice A. Biot	1983		



ENCOMIUM FOR JUDY R. DUBNO

. . . for outstanding contributions to understanding auditory perception, hearing loss, and speech recognition, and for exceptional leadership in the acoustics community

13 MAY 2026 • PHILADELPHIA, PENNSYLVANIA

Judy R. Dubno is an outstanding researcher, mentor, and leader who has made truly extraordinary contributions to acoustics within the Acoustical Society of America (ASA) and beyond, and who is richly deserving of recognition as ASA Honorary Fellow.

Judy Dubno was born and raised in New York City where she attended the city's public schools. She received her bachelor's degree from City College of New York and her PhD degree in Speech and Hearing Science from the City University of New York (CUNY) Graduate School and University Center. Her doctoral research involved fundamental work in predicting consonant confusions by individuals with normal and impaired hearing. She was heavily involved in the development, recording, and analysis of the CUNY Nonsense Syllable Test, among the first tests to quantitatively assess speech recognition in varied listening situations, and still in use today. Judy went on to complete a post-doctoral fellowship at the UCLA School of Medicine, where she remained as a faculty member for several years. It was in Los Angeles that Judy met her husband, John, in 1985. In 1991, she moved to the Medical University of South Carolina (MUSC) in Charleston, where she currently serves as Distinguished University Professor and Vice Chair of Research in the Department of Otolaryngology–Head and Neck Surgery.

Over her career, Judy has been one of the most outstanding scientists in the field of audiology. Her multi-faceted research on human auditory systems has made many seminal contributions to the field, ranging from understanding the encoding of auditory information in simple sounds and speech to how these abilities change in adverse listening conditions, with age, and with hearing loss. In each of these areas, Judy has had a significant impact on our fundamental knowledge of auditory perceptual abilities, and on clinical audiologic methods of assessment and rehabilitation. A particular focus of her research has been to improve our understanding of presbyacusis—the gradual and irreversible decline in hearing ability that affects most people as they age—and its consequences for the perception of speech and other sounds. Judy led a series of foundational studies examining the specific types of acoustic cues and listening situations that lead to problems for hearing-impaired and older people. An important contribution has been her leadership of a 35+ year longitudinal cohort study of age-related hearing loss. This work has been highly influential and greatly increased our understanding of the changes in auditory perception and processing that occur as a consequence of aging and hearing loss. Judy has also been active in research and public service to support improved accessibility and affordability of hearing health care, especially hearing aids, and to enact the necessary federal policies and legislation. She has served on a National Academies of Sciences, Engineering, and Medicine committee on the topic and as a member of the Lancet Commission to Address the Global Burden of Hearing Loss.

Judy's contributions to auditory science have been recognized by many prestigious awards. A partial list of Judy's honors includes the Life Achievement Award and the Jerger Career Award for Research in Audiology of the American Auditory Society, Honors of the Association and the Alfred K. Kawana Award for Lifetime Achievement in Publications of the American Speech-Language-Hearing Association (ASHA), the Award of Merit of the Association for Research in Otolaryngology (ARO), the South Carolina Governor's Award for Excellence in Science, and the ASA Gold Medal. In addition, Judy is a Fellow of the ASA, ASHA, American Academy of Otolaryngology–Head and Neck Surgery (AAO-HNS), International Collegium of Rehabilitative Audiology, and Collegium Oto-Rhino-Laryngologicum Amicitiae Sacrum.

Judy has also been an exceptional mentor and advisor to more than 100 graduate and medical students, postdoctoral fellows, otolaryngology residents, and junior faculty, many of whom have gone on to excel in careers in audiology research, practice, and teaching. In recognition of Judy's remarkable contributions to training the next

generation of audiologists and medical professionals, she was awarded MUSC's Peggy Schachte Research Mentor Award in its inaugural year of 2012.

Judy's service and leadership contributions to organizations related to acoustics have been extraordinary in their breadth and significance. Judy has served on or led an incredible 43 ASA councils, committees, task forces, and working groups. To mention just a few, she served as ASA President (2014-15), Vice President (2010-11), and Executive Council member (2004-07), and co-chaired two ASA strategic leadership initiatives (2011-12 and 2014-15). Of particular importance, as ASA's first elected Treasurer (2019-25), Judy was instrumental in stabilizing the Society's financial operations at a critical time, overseeing a systematic transition from large annual deficits to essentially the break-even point over the extent of her term.

As one of her personal favorite ASA roles, Judy, together with her friend and colleague Brigitte Schulte-Fortkamp, instigated and organized the ASA School in acoustics for graduate students and early-career professionals. Their plan was to hold ASA Schools biennially, over the weekend preceding the spring ASA meeting in even-numbered years, a schedule that has been followed since 2012. In addition to providing lectures, demonstrations, and mentorship, ASA School was intended to enhance the long-term participation and engagement of students and early-career professionals in the field of acoustics and in the ASA, helping to keep the Society strong.

Judy has also made tremendous contributions to national/international organizations outside of ASA, including about 50 boards, councils, foundations, and committees. Of particular note are President, Secretary-Treasurer, and Chair of the Long-range Planning Committee of ARO; Board of Directors of the American Institute of Physics and of the Hearing Health Foundation; Research Advisory Board for AAO-HNS; National Academies of Sciences Board on Behavioral, Cognitive, and Sensory Systems; and National Institutes of Health Advisory Councils. Judy has also served as Associate Editor of the *Journal of Speech and Hearing Disorders* and *Audiology and Neurology*, and on the Editorial Board of the *Journal of the American Academy of Audiology*.

The scope and impact of Judy Dubno's research, mentoring, and leadership in acoustics are truly extraordinary. However, just as impressive are the expertise, insight, diligence, and integrity that she continues to bring to every undertaking, scientific or organizational, motivated by the ASA's mission "to generate, disseminate, and promote the knowledge and practical applications of acoustics." We congratulate her most warmly on this prestigious recognition as Honorary Fellow of the Acoustical Society America.

STAN DOSSO
MICHAEL VORLÄNDER
BARBARA SHINN-CUNNINGHAM

Gold Medal



Anthony A. Atchley
2026

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954	K. Uno Ingard	1997
Floyd A. Firestone	1955	Floyd Dunn	1998
Harvey Fletcher	1957	Henning E. von Gierke	1999
Edward C. Wente	1959	Murray Strasberg	2000
Georg von Békésy	1961	Herman Medwin	2001
R. Bruce Lindsay	1963	Robert E. Apfel	2002
Hallowell Davis	1965	Tony F. W. Embleton	2002
Vern O. Knudsen	1967	Richard H. Lyon	2003
Frederick V. Hunt	1969	Chester M. McKinney	2004
Warren P. Mason	1971	Allan D. Pierce	2005
Philip M. Morse	1973	James E. West	2006
Leo L. Beranek	1975	Katherine S. Harris	2007
Raymond W. B. Stephens	1977	Patricia K. Kuhl	2008
Richard H. Bolt	1979	Thomas D. Rossing	2009
Harry F. Olson	1981	Jiri Tichy	2010
Isadore Rudnick	1982	Eric E. Ungar	2011
Martin Greenspan	1983	William A. Kuperman	2012
Robert T. Beyer	1984	Lawrence A. Crum	2013
Laurence Batchelder	1985	Brian C. J. Moore	2014
James L. Flanagan	1986	Gerhard M. Sessler	2015
Cyril M. Harris	1987	Whitlow W. L. Au	2016
Arthur H. Benade	1988	William M. Hartmann	2017
Richard K. Cook	1988	William A. Yost	2018
Lothar W. Cremer	1989	William J. Cavanaugh	2019
Eugen J. Skudrzyk	1990	Judy R. Dubno	2020
Manfred R. Schroeder	1991	James F. Lynch	2021
Ira J. Hirsh	1992	Michael J. Buckingham	2022
David T. Blackstock	1993	Mark F. Hamilton	2023
David M. Green	1994	Ingo R. Titze	2024
Kenneth N. Stevens	1995	Arthur N. Popper	2025
Ira Dyer	1996		



ENCOMIUM FOR ANTHONY A. ATCHLEY

…for his contributions to the science of physical acoustics and his dedicated service to the Acoustical Society of America

13 MAY 2026 • PHILADELPHIA, PENNSYLVANIA

Three months before Sputnik was launched, Anthony was born in Lebanon, Tennessee, where the tempo of life was marked by the long, soulful wail of the woolen mill's steam whistle three times each workday. Growing up during the space race sparked Anthony's life-long fascination with space exploration, astronomy, and physics.

Young Anthony spent many happy days roaming the fields, woods, and creeks on his maternal grandparents' ranch. Holidays usually meant a family gathering, incidentally producing enough people to undertake the seasonal move of the cattle herd from one pasture to another, along a route including a section of major highway and narrow country roads. Anthony learned that the cattle mostly knew where they wanted to go, and only needed people to remove barriers, and not let them make wrong turns or get injured along the way. He is silent on whether this early experience influenced his future leadership style.

After graduating from the University of the South (Sewanee, Tennessee), Anthony attended graduate school at New Mexico Tech, studying thunderstorms which formed in the summer months over a laboratory atop an 11,000 ft mountain. He had the great fortune to learn electromagnetic theory from Rossi Lomanitz, who had started his graduate studies with J. Robert Oppenheimer and finished with Richard Feynman.

As Anthony was finishing his master's degree, another student told him about physical acoustics research at the University of Mississippi. This was the first time Anthony ever heard of acoustics as a field of advanced study. Fortunately, the library at New Mexico Tech subscribed to the *Journal of the Acoustical Society of America* (JASA*),* and Anthony's journey in acoustics and the Acoustical Society of America (ASA) began.

Pursuing a PhD at the University of Mississippi, Anthony started with a group studying the underwater sound that lightning makes when it strikes the ocean's surface. Being the newest graduate student, Anthony was assigned to deploy a hydrophone from the research vessel that was chasing thunderstorms off the Mississippi coast. When conditions seemed right for lightning, his job was to jump into the ocean, swim away from the boat with the hydrophone and associated electronics in tow, and then swim back to the boat. The audio of one of the underwater recordings of a lightning strike is available on the Discovery of Sound in the Sea website.

Thankfully, Anthony's thunderstorm-chasing days ended when Larry Crum offered him a position studying acoustic cavitation. In 1984, diagnostic ultrasound was becoming widespread and Larry suggested to Anthony that he consider whether these scanners could induce acoustic cavitation*,* which might be dangerous to fetuses. Anthony collected a wide data set of the thresholds of cavitation in different aqueous media. Larry then suggested he do some theoretical calculations to try to create a valid model of cavitation inception. As this project became more complicated, Anthony consulted with Andrea Prosperetti, a globally renowned fluid-dynamics expert. Their classic paper "*The Crevice Model of Bubble Nucleation" (JASA* 1989*)* has received over 350 citations---31 citations in 2025 alone.

After graduating from the University of Mississippi, Anthony was offered a position at the Naval Postgraduate School (NPS) in Monterey, CA. But he had also received a Hunt Fellowship to work in Bob Apfel's group at Yale. Facing a major dilemma, he convinced NPS that he should go to Monterey for a quarter, and then to Yale for a year, promising that he would then return to Monterey. At Yale, he joined a coterie of rising stars, and Anthony and his colleagues published the first papers on passive and active cavitation detection, techniques used universally today.

Back in Monterey, Anthony met his future wife Mary, who also worked in the Physics Department, and shifted his research focus to thermoacoustics and related fundamental physical acoustics. He was awarded the ASA R. Bruce Lindsay Award in 1992 "For contributions to the understanding of acoustic cavitation and thermoacoustics." Two of his widely cited papers from that time show Anthony's attraction to unexpectedly tough challenges. Although promoted to Chair of the Physics Department at NPS, Anthony saw more

opportunities elsewhere, and in 1997 he became Chair, Graduate Program in Acoustics at Penn State University (PSU).

Anthony's style was to think big and be very welcoming, gaining tremendous respect throughout the university as well as the acoustics program. Anthony let the way in Distance Education, working with others, especially Dick Stern, who had started the distance program in 1987. At that time, most courses at PSU were either purely in-class or purely broadcast for distance students in the evenings. Under Anthony's leadership, exploiting new internet technology, all courses shifted in 2006 to live transmission *and* recording of daytime courses, so daytime distance students and evening, working-professional students could feel like they were part of the real classroom attended by resident students. PSU Acoustics follows the same model today, using Zoom. Anthony achieved other great things during his 12-year tenure as leader of the PSU Acoustics Program, including fundamental nonlinear acoustics and aircraft noise research, but it was clear that he was destined for bigger things.

Anthony became Associate Dean for Administration and Planning in the Penn State College of Engineering in 2008, transitioned to Associate Dean for Research and Administration in 2009, and then Senior Associate Dean in 2014. Anthony kept his number two position in the College under three externally hired Deans, who normally clean house when setting up a new administration. Anthony also co-chaired important Penn State committees that kept Penn State operating throughout the Covid-19 pandemic. In 2022, he delayed retirement to serve as Acting and then Interim Dean of the College of Engineering, doing a great job during a challenging budgetary period. Anthony retired from Penn State on June 30, 2023.

Anthony has a distinguished publication record, but perhaps his greatest contribution to the ASA has been his strong record of service. He served on the ASA Executive Council from 1998–2001, as Vice President from 2003-2004, and as President from 2006-2007. He also served as chair or member of the Technical Program Organizing Committee for seven national and international meetings. For 14 years, Anthony served on the Acoustical Society Foundation Board, and as Chair from 2012-2015. In 2012 he was asked to Chair the newly formed Finance Committee. He also served as a member of several additional ASA committees, chairing three, including Physical Acoustics, Education in Acoustics, Prizes and Special Fellowships, Medals and Awards, Investments, Publication Policy, Tutorials, and Long-Range Planning. As a former President, he served on the American Institute of Physics Governing Board for 9 years and was elected to their Executive Committee. He also served on the American Society of Engineering Education's Engineering Research Council Board of Directors.

Those of us for whom Anthony has quietly removed barriers and prevented wrong turns are grateful for his leadership.

LAWRENCE A. CRUM
GREGORY W. SWIFT
VICTOR W. SPARROW

Session 4aAA

Architectural Acoustics, Psychological and Physiological Acoustics, Speech Communication, and Signal Processing in Acoustics: The Many Dimensions of Jont B. Allen's Scientific Legacy

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180

Sunil Puria, Cochair
Otolaryngol.-HNS, Harvard Univ., 243 Charles St., Boston, MA 02114

Chair's Introduction—8:15

Invited Papers

8:20

4aAA1. How fortunate I was to be mentored by Jont B. Allen during my Ph.D. at Bell Labs. Sunil Puria (Eaton-Peabody Labs, Otolaryngol.-HNS, Harvard Med. School, 243 Charles St., Boston, MA 02114, Sunil_Puria@meei.harvard.edu)

Like the movie *It's a Wonderful Life*, my own career trajectory hinged on a chance encounter. Had I not met Jont B. Allen, my life in auditory science would have unfolded very differently. My doctoral research, conducted largely at the storied AT&T Bell Laboratories in Murray Hill, NJ, immersed me in a culture of high-impact innovation where fundamental science and real-world engineering were inseparable. Under Jont's mentorship, I developed theoretical models of middle-ear and cochlear mechanics that connected physical acoustics with applied research. Bell Labs was a crucible for modern technology, and working with Jont meant being challenged to think deeply, quantitatively, and creatively. Our collaborations resulted in two of my most favorite papers published in JASA. Jont's unique mentoring style, characterized by his unwavering dedication to clarity, rigor, and first-principles thinking, profoundly influenced my scientific and engineering approach, which has remained steadfast ever since. Beyond the equations and models, he taught me how to ask better questions, challenge assumptions, and pursue ideas that bridge theory, experiment, and application, all the while having fun and building lifelong friendships. I will always cherish the many enjoyable evenings spent at Jont and Pat's home, where science, laughter, and community came together.

8:35

4aAA2. Jont B. Allen's influence on hearing aid signal processing. Brian C. Moore (Dept. of Psych., Univ. of Cambridge, Downing St., Cambridge CB2 3EB, United Kingdom, bcjm@cam.ac.uk)

During his time at AT&T Bell Labs, Jont B. Allen played a key role in developing the first commercially available two-channel wide-dynamic-range compression (WDRC) hearing aid, the ReSound device, based on the design of Edgar Villchur. Jont Allen, Edgar Villchur, and I (and Mead Killion) were all on the Scientific Advisory Board of ReSound, and we had some very lively discussions, to put it mildly. Jont was also responsible for developing a test assessing the growth of loudness in different frequency bands [Allen, J. B., Hall, J. L., & Jeng, P. S. (1990). Loudness growth in 1/2-octave bands (LGOB); a procedure for the assessment of loudness. *J. Acoust. Soc. Am.* **88**, 745–753] that was used for fitting the ReSound device and likely contributed to its commercial success. With the advent of digital hearing aids, the overlap-add method developed by Jont [Allen, J. B. (1977). Short term spectral analysis, synthesis and modification by discrete fourier transform. IEEE Trans. Acoust. Speech Sig. Proc. **25**, 235–238] became widely used for implementing WDRC and other forms of signal processing. Variations of this method continue to be used today as part of the deep neural networks that are increasingly being implemented in hearing aids.

8:50

4aAA3. Jont Allen and SYSid. Gary W. Elko (MH Acoust., 25A Summit Ave., NJ 07901, gwe@mhacoustics.com)

I first met Jont in person in 1984 when I joined Bell Labs. I was familiar with Jont from the classic papers he wrote with David Berkley on the image model for room acoustics and the dereverberator algorithm. The first lab computer I used for real-time simulations of array beamforming was based on real-time software that he and his colleagues wrote running on a Data General minicomputer that had many buttons and potentiometers that allowed users to interact with the program in real-time. Jont's experimental approach was meticulous in planning, execution, and analyzing the data from his experiments. I learned a lot from observing how he worked. One specific project that we worked on together was a systems identification application that was called SYSid. In the late 80's, I helped develop this application on an IBM PC augmented with an Ariel DSP16 TI TMS320-based DSP board. In this commemorative talk, I will describe the basic design and capabilities of SYSid. I will show SAMHX which is a derivative of SYSid that was developed to analyze microphone array systems and devices at MH acoustics over the past two decades.

9:05

4aAA4. Acoustics and the auditory system. Stephen Neely (Boys Town Nat'l Res. Hosp., 555 North 30th St., Omaha, NE 68131, neelys@gmail.com)

This presentation highlights research collaborations with Jont Allen, beginning with work conducted at Bell Labs in 1975. The initial project investigated the dereverberation of speakerphone audio through the spectral inversion of simulated room impulse responses. The study concluded that exact inversion is theoretically restricted to minimum-phase systems (Neely & Allen, 1979). Subsequent collaborations addressed cochlear mechanics and psychoacoustics. Allen and Neely (1992) simulated experimental measurements with theoretical models to describe the micromechanics of the cochlea. This work contrasted two alternative points of view regarding cochlear function. Later, Allen and Neely (1997) utilized signal detection theory to derive a theoretical link between the physical domain of sound intensity and the perceptual domain of loudness. Most recently, Norgaard, Allen, and Neely (2021) analyzed the causality of aural impulse responses. This study revisited the physical acoustic principles established in the 1979 work, applying similar theoretical constraints to define the physical realizability of acoustic signals in the ear.

9:20

4aAA5. Jont Allen: Polymathic engineer, scientist, teacher, and mentor. Susan E. Voss (Eng., Smith College, 100 Green St., Ford Hall, Northampton, MA 01063, svoss@smith.edu)

One of the most transformative experiences of my professional life was being paired with Jont Allen as my mentor in 1992 through the AT&T Graduate Research Program for Women at Bell Laboratories in Murray Hill, New Jersey. While the grant provided financial support, its true value lay in the mentorship and internship I received during the summer before I began graduate school at MIT. Working with Jont was a profound learning experience. He taught me invaluable lessons about research and mentoring, and his guidance was instrumental in the development and execution of the experiments that led to my first peer-reviewed publication, *"Measurement of acoustic impedance and reflectance in the human ear canal,"* published in *JASA*. Jont's contributions in this area laid important groundwork for the establishment of these measures as clinical tools and continue to shape the field today. Beyond his pioneering work in middle-ear impedance and reflectance, Jont's polymathic contributions span auditory mechanics, psychophysics, speech perception, and fundamental mathematics and physics. His wide-reaching scholarship has profoundly shaped the field of speech and hearing and the careers of many, including myself, and I am deeply grateful to be among them.

9:35–9:50 Break

9:50

4aAA6. From electroacoustic theory to data-driven models: Insights rooted in learning from Jont Allen. Noori Kim (Purdue Univ., 401 N. Grant St., Rm. 115, West Lafayette, IN 47907, kim4147@purdue.edu)

Electroacoustic systems form a foundational interface between electrical and acoustic domains, enabling both the generation of sound perceptible to humans and the sensing of acoustic information for computational processing. Through this bidirectional transduction, electroacoustic devices are deeply intertwined with human hearing, auditory perception, and acoustic science. In this talk, we review experimentally observed electrical impedance characteristics arising from electroacoustic coupling in a range of transducers. By examining variations across devices and operating conditions, we identify common structural and physical features that govern their behavior. These experimental observations are then interpreted using classical matrix-based representations, including ABCD matrix models, and extended toward modern data-driven approaches such as machine learning and artificial intelligence. By bridging classic electroacoustic theory with contemporary computational techniques, this talk aims to provide a unified perspective on electroacoustic systems—highlighting how foundational principles continue to inform emerging methodologies and applications in auditory sensing and signal analysis.

10:05

4aAA7. Hall of mirrors. Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de)

The groundbreaking publication by Allen and Berkley (Image method for efficiently simulating small-room acoustics, 1979, https://doi.org/10.1121/1.382599) with 5790 citations is probably one of the most cited articles in JASA. It contains a rigorous application of the image source model for a rectangular room and its implementation in computer software. This work was of crucial importance for all simulation techniques of room impulse responses with high temporal resolution—and thus for digital signal processing and auralization. The work is revisited and discussed further, focusing on modal theory and hybrid models.

10:20

4aAA8. From the image model method to its enhanced extensions for room impulse response simulation. Jingdong Chen (Northwestern Polytechnical Univ., 49 Hudnut La, Belle Mead, NJ 08502, jingdongchen@ieee.org), Jacob Benesty (INRS-EMT, Univ. of Quebec, Montreal, QC, Canada), and Chao Pan (Northwestern Polytechnical Univ., Xi'an, Shannxi, China)

The image model method, developed by Prof. Jont Allen, is a well-recognized technique for simulating room impulse responses. According to Google Scholar, the original paper has been cited nearly 6000 times, while many subsequent studies have employed the method without explicit citation. There is little doubt that this approach is among the most influential methods in the fields of room acoustics and speech processing. It has been widely adopted, and efforts to adapt and extend the method for different applications have attracted significant interest over the past several decades. In this paper, we extend the classical image model method by proposing an anchor-point image model to simulate impulse responses that incorporate both source radiation and sensor directivity patterns. To determine the orientations of all virtual sources, anchor points are introduced for the real sources, enabling the corresponding orientations of

the virtual sources to be derived. An efficient algorithm is developed to generate room impulse responses while accounting for directional pattern functions, fractional time delays, and computational complexity. The proposed model and algorithms are applicable to a wide range of acoustic problems, providing an effective tool for simulating room acoustics as well as for improving and evaluating acoustic signal processing algorithms.

10:35

4aAA9. Ear to brain: Honoring Jont Allen's contributions to hearing and speech. Fan-Gang Zeng (UCI, 110 Med. Sci. E, Irvine, CA 92697, fzeng@hs.uci.edu)

I have had numerous interactions with Jont Allen since 1987, but the most memorable occurred in Yellowstone National Park, where our families happened to meet in 2001. That encounter reinforced my long-held view of Jont Allen as a brilliant engineer and also revealed him as a devoted family person, mentor, friend, and colleague. Many foundational ideas in modern hearing and speech research trace back to Harvey Fletcher and his colleagues at Bell Labs in the 1920s–1940s. Jont Allen has carried this Bell Labs legacy forward, making seminal contributions to both research and education across topics ranging from eardrum impedance and cochlear mechanics to auditory and speech perception. In this talk, I will build on two foundational ideas—the ear as a filter and the ear as a compressor—extending them to loudness and speech perception and relating them to the encoder–decoder framework in artificial intelligence.

THURSDAY MORNING, 14 MAY 2026 FRANKLIN HALL 9, 8:00 A.M. TO 11:35 A.M.

Session 4aBA

Biomedical Acoustics: Shocking Considerations in Therapeutic Ultrasound I

Kenneth B. Bader, Cochair
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Jonathan R. Sukovich, Cochair
Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109

4a THU. AM

Invited Papers

8:00

4aBA1. Shock wave exposures in medical applications: Highly nonlinear acoustic beams and physical mechanisms of corresponding therapeutic bioeffects. Vera Khokhlova (Phys. Faculty, Lomonosov Moscow State Univ., Leninskie Gory, MSU, 1/2, Moscow 119991, Russian Federation, vera@acs366.phys.msu.ru), Maria M. Karzova, Petr V. Yuldashev, and Oleg A. Sapozhnikov (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation)

A number of modern ultrasound therapeutic approaches rely on shock waves generated by high-power focusing transducers. Ultrasound sources of different dimensions, geometries, and frequencies have been employed. Single shock pulses, short-duration ultrasound bursts, and continuous waves have been used. Abrupt shocks gradually develop in the initial pressure waveform when propagating from the transducer to the focus due to combined effects of acoustic nonlinearity and diffraction. In addition, the waveform becomes strongly asymmetric with higher sharp positive pressures and lower in magnitude smooth negative pressures. The presence of shocks significantly broadens the spectrum of possible ultrasound-induced bioeffects in tissue. Here, specific features of shock wave fields, their spatial and temporal patterns, challenges of their accurate modeling and measuring, effects of nonlinear saturation of shock amplitude are overviewed. While methods have been developed for characterizing shock wave fields in water, tissue inhomogeneities often distort the focusing, resulting in diminished shock amplitudes. Experimental verification of shock-forming conditions *in situ* is thus needed. Mechanical (cavitation, radiation force, streaming) and thermal bioeffects in tissue in the presence of shocks are discussed. Corresponding clinical applications, such as histotripsy, lithotripsy, shock wave therapies, and ultrasound-guided thermal ablation, are also presented. [Work supported by RSF 25-12-00157.]

8:20

4aBA2. Medical shock waves: Outcomes, mechanisms, and acoustic reality. Cyrill Slezak (Phys., Utah Valley Univ., 800 West University Parkway, Orem, UT 84058, CSlezak@uvu.edu)

Extracorporeal shock wave therapy (ESWT) is an established non-invasive regenerative treatment with proven clinical effectiveness in common musculoskeletal indications such as plantar fasciitis, tendinopathies, and delayed bone healing, with growing use in soft-tissue and regenerative applications. Its increasing routine clinical adoption is paralleled by a rapid expansion toward novel and exploratory indications. While clinical outcomes are frequently favorable, systematic *insitu* evaluation of applied acoustic fields remains challenging due to reference point-limited measurements and the high computational demands of nonlinear shock wave modeling in heterogeneous tissues. Clinical outcome data across established and emerging indications motivate refined investigation of proposed modes of action, including mechanotransduction, cavitation-mediated effects, modulation of inflammation, and stimulation of angiogenesis and tissue remodeling. Investigation of contraindications and safety margins is constrained by ethical and methodological limitations, as direct experimental assessment in patients is rarely feasible. Consequently, sound-field characterization relies on cadaveric and preclinical reference measurements, which must be augmented by high-fidelity numerical simulations. Integrating clinical evidence, experimental acoustics, and computational modeling is essential for reliable reproducible ESWT research, mechanism-informed optimization of treatment parameters, and effective clinical translation toward improved patient outcomes.

8:40

4aBA3. Acoustic field mapping of an intravascular lithotripsy (IVL) catheter: Evidence for consistent, nondirectional output. Camilo Perez Saaibi (Circulatory Restoration, Johnson & Johnson Med. Tech., 3003 Bunker Hill Ln., Santa Clara, CA 95054, cperezs2@its.jnj.com), Adrienne Ockenhouse, and Thomas Hasenberg (Circulatory Restoration, Johnson & Johnson Med Tech, Santa Clara, CA)

Electrohydraulic intravascular lithotripsy (IVL) is an established therapy for vascular calcium modification, offering distinct advantages over electromagnetic, laser, and piezoelectric lithotripsy systems. This study characterizes the acoustic output and therapeutic consistency of a spherical, unfocused, electrohydraulic IVL emitter design. Acoustic field mapping demonstrates a nondirectional, homogeneous pressure and energy distribution along the catheter's working length, supporting its ability to treat diverse calcium morphologies—including nodular, eccentric, and circumferential (>180 deg) lesions—across varied anatomical geographies. Therapeutic penetration depth was demonstrated through pre- and post-treatment optical coherence tomography (OCT) and intravascular ultrasound (IVUS), confirming modification of both superficial and deep calcium layers. Design considerations, including multiple-pair versus single-pair emitter arrays, were analyzed for their impact on working length and treatment uniformity. Pulse-to-pulse variability remained minimal, indicating stable source performance throughout therapy. Mechanistic insights suggest shear forces as the dominant mode of action, supported by correlations between acoustic metrics, high-speed imaging, and preclinical (Micro-CT, OCT, and IVUS) data. These findings establish IVL as a consistent, nondirectional ultrasonic therapy with validated efficacy across complex calcium morphologies, reinforcing its role as a solution for challenging coronary and peripheral interventions.

9:00

4aBA4. Non-invasive ultrasound therapy of calcific aortic valve stenosis. Mathieu Pernot (Inst. Phys. for Medicine, Rue Jussieu, Paris 75004, France, mathieu.pernot@gmail.com)

Calcific aortic stenosis (CAS) is the most common heart valve disease. To date, surgical or transcatheter aortic valve (AV) replacement is the only effective treatment. However, 16% of patients with symptomatic severe CAS are rejected by local heart teams for surgical or AV replacement. 10 years ago, we proposed a novel non-invasive therapeutic approach based on the use of pulsed cavitational focused ultrasound to improve the function of calcified valves. The shock wave and cavitation induced by short, high-pressure ultrasound pulses focused on the valve generate a mechanical softening of the targeted calcified leaflet. This concept was first validated *in vitro* using degenerative aortic bioprosthetic valves and *in vivo* on a large animal model. A clinical non-invasive ultrasound image-guided therapeutic device (Valvosoft, Cardiawave, France) was designed based on this concept to improve the function of calcified AV. 100 patients with symptomatic severe CAS not eligible for AV replacement were treated in a European prospective study performed in 15 European centers. The valve area increased by 47% after treatment when compared to the natural progression of the disease and the stroke volume increased by 28%. Based on these results, the device received European medical device certification.

Contributed Papers

9:20

4aBA5. Frequency-dependent acoustic properties of urinary stones and their implications for shock wave-induced stress simulations. Xin Xu (Thomas Lord Dept. of Mech. Eng. and Mater. Sci., Duke Univ., 144 Hudson Hall, 100 Sci. Dr., Durham, NC 27708, xin.xu@duke.edu), Junqin Chen, Bavand Keshavarz, Arpit Mishra, Obed Isaac, and Pei Zhong (Thomas Lord Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Accurate quantification of acoustic attenuation in urinary stones is essential for predicting stress field development inside the stone during shock wave lithotripsy (SWL) and laser lithotripsy (LL). In this study, a broadband ultrasonic transmission spectroscopy method was employed to measure the frequency-dependent longitudinal and shear wave velocities and attenuation in artificial and human kidney stones. A theoretical correction was introduced to account for the Gaussian beam broadening of the incident pulse after transmission, which improved the accuracy of the measurement. The reliability of the method was verified through a comparison with reference measurements performed on a standard polymethyl methacrylate (PMMA) plate. The measured attenuation-frequency curve exhibited a linear relationship and was well described by a fractional Kelvin–Voigt model of near-zero order. The experimentally determined acoustic parameters were then incorporated into finite element simulations of shock wave propagation within cylindrical stones. The results revealed that stronger acoustic attenuation not only reduces the peak tensile stress but also shifts its location from the interior of the stone toward the surface, indicating a different damage initiation position. These findings highlight the critical role of attenuation in fracture initiation and provided a more accurate basis for future simulations of stress and damage evolution in kidney stones.

9:35

4aBA6. Shockwave transmission and biofilm disruption at knee implant interfaces. Veerle A. Brans (Inst. of Biomedical Eng., Univ. of Oxford, Botnar Res. Ctr., Old Rd., Oxford OX3 7LD, United Kingdom, veerle.brans@eng.ox.ac.uk), Sara Keller, Ramon Garcia Maset (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Abtin Alvand (Nuffield Orthopaedic Ctr., Oxford Univ. Hospitals, Oxford, United Kingdom), Michael Gray (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Dario Carugo (Nuffield Dept. of Orthopaedics, Rheumatology and Musculoskeletal Sci., Univ. of Oxford, Southampton, United Kingdom), Constantin Coussios, Robin O. Cleveland, and Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Orthopaedic knee implant-associated infections caused by biofilm-forming bacteria present clinical and economic challenges due to poor antibiotic penetration and immune evasion. Shock wave (SW)-based therapies are a promising non-invasive approach to biofilm disruption, and acoustic models of the complex propagation environment would aid translational efforts. We treated *Staphylococcus aureus* biofilms grown on ∼12-mm diameter stainless steel, titanium, and hydroxyapatite discs with either 50 SWs (focused on the centre of the disc; 3.5 MPa peak negative pressure (PNP), 0.5 Hz; 70 mJ incident energy) or 6 SWs (scanned over the disc; 3.5 MPa PNP; 5 mJ incident energy). Passive cavitation detection (PCD) characterized bubble cloud dynamics, including collapse time. The 50 centred SWs produced 3.8-log (95% CI: 3.1–4.5) reductions in biofilm-associated bacteria, and the six scanned SWs produced 2.5-log (95% CI: 1.8–3.2) reductions, demonstrating that shock waves have the potential to remove biofilm with relatively low energy. Optical profilometry was carried out on the disc to assess surface damage. Three-dimensional simulations were conducted using CT-derived anatomy for knee implants, identifying regions where SW energy can be effectively delivered to the knee joint. These results suggest that non-invasive SW therapy can be used to address implant-related infections.

9:50–10:05 Break

Invited Papers

10:05

4aBA7. Ultra-high-rate characterization of viscoelastic materials via bubble collapse. Jon B. Estrada (Mech. Eng., Univ. of Michigan, Ann Arbor, MI 48109, jbestrad@umich.edu)

Rapid, accurate characterization of soft, viscoelastic materials at high strain rates is of interest in biological and engineering applications. Examples include assessing the extent of tissue ablation during histotripsy procedures and developing injury criteria for the mitigation of blast injuries. Our own inertial microcavitation rheometry (IMR) allows for the characterization of local viscoelastic properties at strain rates up to 10^8 s^{-1}. However, IMR typically relies on bright-field videography of a sufficiently translucent sample at $\geq$1 million frames per second and a lengthy simulation-dependent fit optimization process. Multiple recent developments have taken the technique forward, including the development of analytical approximations for collapse time based on the initial approach by Rayleigh but accounting for much more of the physics and determining the likelihood of a model being optimal using Bayesian inference. We present the original technique and new developments in the space, such as the nearly real-time characterization of soft, viscoelastic hydrogels and biological materials with a numerical criterion assessing the correct choice of model.

10:25

4aBA8. Effects of transducer and pulsing parameters on histotripsy bubble cloud characteristics and ablation. Eli Vlaisavljevich (Dept. of Eng. Mech., Virginia Tech, Blacksburg, VA 24061, eliv@vt.edu)

Histotripsy is the first completely non-invasive, non-ionizing, and non-thermal ablation technology. Guided by real-time imaging, histotripsy uses focused ultrasound pulses delivered from outside the body in order to mechanically destroy tissue through the precise generation of acoustic cavitation. While acoustic bubble activity is often characterized as chaotic, short-duration histotripsy pulses can produce consistent cavitation clouds that render targeted tissue into acellular debris with high precision and tissue selectivity. However, although histotripsy has shown promising results for a wide range of applications in preclinical and early clinical studies, the optimal transducer and pulsing parameters for specific clinical applications remain unclear. Studies from our group have aimed to fill this gap in knowledge by investigating the physical thresholds to initiate cavitation and produce tissue damage in histotripsy. This talk will provide an overview of studies comparing the effects of various transducer (frequency, f-number) and pulsing parameters (cycles, pulse repetition frequency, pressure level) on histotripsy bubble cloud formation, spatiotemporal cavitation dynamics, and tissue damage. The effects of tissue mechanical properties on histotripsy bubble cloud dynamics and ablation will also be compared for both multi-cycle and single-cycle histotripsy regimes.

10:45

4aBA9. Histotripsy-induced tendon fractionation: Mechanistic insights and feasibility. Grace M. Wood (Graduate Program in Acoust., Penn State, University Park, PA), Jacob C. Elliott (The Penn State Univ., University Park, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu)

Histotripsy has been shown to liquefy most soft tissues and is now approved clinically to treat liver cancer. However, highly collagenous tissues, like tendon, have been resistant to histotripsy-induced fractionation, despite evidence of cavitation on B-mode ultrasound. Here, I will share our strategies towards successful fractionation of tendon, including a discussion of the possible mechanisms and monitoring modalities that may allow for controlled mild fractionation. Fueled by our prior work showing dual-frequency histotripsy causes complete tendon fractionation, although with significant thermal necrosis, we evaluated pressure-modulated histotripsy to treat tendons. Results indicate that localized heating increases the susceptibility of tendon to histotripsy-induced fractionation; however, there is a very narrow window between mechanical fractionation and thermal necrosis, necessitating a method for monitoring the treatment. Using hydrogels, we showed that histotripsy treatments of >90% dehydrated fibrin produce similar cavitation emission energies to tendon. Yet evaluation of cavitation energies, alone, was insufficient to predict fractionation in tendon. We therefore explored bubble color Doppler

and found better, but not complete, correlation with fractionation. These results suggest that histotripsy fractionation of tendon is feasible, but real-time monitoring is necessary to create the desired level of mechanical disruption. [Work supported by NIH R01EB032860.]

Contributed Papers

11:05

4aBA10. Characterization of flow and dose-dependent histotripsy bubble dynamics for DVT models. Eric J. Galindo (Radiology, Univ. of Chicago, P.O. Box 2404, 801 Leroy Pl., Socorro, NM 87801, ejgalindo@uchicago.edu), Erin S. Kim, Erik Saucedo, and Kenneth B. Bader (Univ. of Chicago, Chicago, IL)

Deep vein thrombosis (DVT) involves occlusion of the iliofemoral vasculature with risk of pulmonary embolization. The limitations of current interventions for DVT motivate the development of improved therapeutic strategies. Histotripsy is a non-thermal focused ultrasound technique that generates bubble clouds to fragment tissue mechanically, including thrombus. Prior benchtop investigations of histotripsy for DVT relied on static flow conditions and a fixed dose (i.e., pulses applied/mm), which potentially limits treatment efficiency. The goal of this study was to determine the relationship between histotripsy bubble activity, dose, and vascular flow rate. Histotripsy pulses (30 MPa peak negative pressure and 6.67 μs pulse duration) were applied to an *in vitro* model of venous flow (0 to 170 mL/min) to achieve doses of 2.5 to 4000 pulses/mm (50 to 400 Hz pulsing rate and 0.1 to 20 mm/s scan rate). The energy density of bubble activity was quantified with B-mode imaging and passive detection of acoustic emissions. The energy density was found to increase at low doses of histotripsy, but remained fixed beyond a threshold dose (42, 1117, and 75 pulses/mm at 0, 60, and 170 mL/min, respectively). These findings elucidate flow-dependent regimes that govern histotripsy bubble dynamics.

11:20

4aBA11. Histotripsy-induced activation of platelets in whole blood. Erik Saucedo (Radiology, Univ. of Chicago, 5842 S Ellis Ave., Chicago, IL 60637, erik.saucedo@bsd.uchicago.edu), Kevin Zhao (Radiology, Univ. of Chicago, Chicago, IL), Eric J. Galindo (Radiology, Univ. of Chicago, Socorro, NM), Everett Tate, David Leclerc (Cytometry & Antibody Technol. Core Facility, Univ. of Chicago, Chicago, IL), Timothy Carll (Pathol., Univ. of Chicago, Chicago, IL), Geoffrey Wool (Pathol., Medical College of Wisconsin, Milwaukee, WI), and Kenneth B. Bader (Radiology, Univ. of Chicago, Chicago, IL)

Histotripsy is a form of focused ultrasound that uses acoustic cavitation for tissue ablation. A primary advantage of the technology is that it spares major vessels. Recent clinical data indicate thrombosis within vessels in the treatment zone, suggesting there may be resulting shifts in blood chemistry. A primary hypothesis for thrombus formation is platelet activation under histotripsy exposure. To test this hypothesis, histotripsy-induced changes to platelets in human whole blood were assessed *ex vivo*. Whole blood samples were exposed to histotripsy for 10 min (10 μs pulse duration, 33 MPa peak negative pressure, and 80 Hz pulsing rate). Treated or sham samples were then analyzed with light transmission aggregometry and flow cytometry. Platelet aggregation was reduced in histotripsy arms relative to sham for 5 of the 13 agonists, with more than a 50% decrease for TRAP-6 and U46619. This finding was consistent with flow cytometry analysis, which indicated a 20%–50% reduction in individual platelets but increase in an activation for the histotripsy arm relative to sham. Subsequent size analysis indicates platelets aggregate under histotripsy exposure, thereby reducing the number of individual, isolated platelet cells. Overall, these findings suggest histotripsy causes a local hypercoagulability when applied to whole blood.

Session 4aCA

Computational Acoustics, Biomedical Acoustics, Underwater Acoustics, and Physical Acoustics: Ray Methods Across Acoustics

Jennifer Cooper, Cochair
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Subha Maruvada, Cochair
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Invited Papers

9:00

4aCA1. Ray and beam tracing methods in ocean acoustics. Michael B. Porter (Heat, Light, and Sound Res., Inc., 12625 High Bluff Dr. Ste 211, San Diego, CA 92130-2054, Porter@HLSResearch.com)

Ray tracing methods have a long history in ocean acoustics. A remarkable paper by H. Lichte, "On the Influence of Horizontal Temperature Layers in Sea Water on the Range of Underwater Sound Signals," dates back to 1919 covers German work in World War I. So-called full wave methods based on normal modes, parabolic equation, and wavenumber integration methods became increasingly popular in the 1970s; however, ray methods remained dominant until then. Many ray models were developed in the 1960s and 1970s including TRIMAIN, Grass, and RAYWAVE II to name a few. Broadly speaking, these models can be divided into two classes. One used "cells" or triangular patches with simple forms for which analytic solutions of the ray equations exist. The other class solved the ordinary differential equations (ODEs) that govern the rays using traditional ODE solvers such as Runge–Kutta methods. However, these traditional approaches have awkward challenges in finding the eigenrays to connect the source to the receiver. Beam tracing methods were introduced by Bucker in the 1970s. They provided a numerically convenient way to address this, and generalizations of this approach are predominant today. Most of the existing work considered 2-D propagation but recent developments address full 3-D effects. This talk will discuss the history and development of ray methods in ocean acoustics.

9:20

4aCA2. Ray tracing reconstruction of long-range acoustic vortex fields. Jianyu Hua (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, elihua@umich.edu) and Chengzhi Shi (Univ. of Michigan, Ann Arbor, MI)

Ray tracing is a great way for long-range propagation in complex, range-dependent media, yet it is often viewed as incompatible with fields whose structure is inherently three-dimensional (e.g., vortical phase patterns). We present a ray-based workflow that reconstructs complex pressure, amplitude, and phase of structured acoustic sources over long ranges while retaining the efficiency of established ray codes. Each transducer in a concentric-ring phased array is modeled as an individual point source; its contribution is propagated through an inhomogeneous environment and then coherently summed using prescribed phase relationships to recover the composite field. For wavefront-plane predictions, pressure samples computed along the range axis are projected onto an imaging plane using geometric path-length corrections, enabling high-resolution slices of the three-dimensional field from primarily two-dimensional ray outputs. The framework naturally incorporates refraction, boundary reflections, and multipath, and it supports post-processing such as spatial-frequency filtering to isolate direct-path energy and recover distorted phase. Example results show that vortex signatures and topological charge can be tracked and interpreted in ducted, refractive, and strongly multipath environments. The approach provides a scalable bridge between full-wave simulation and operational-scale prediction for structured sources.

9:40

4aCA3. Practical differences in statistical and raytracing methods for calibrating room models and acoustic simulation. Stefan Feistel (Ahnert Feistel Media Group, Berlin, Germany), Bruce C. Olson (AFMG Services North America LLC, 8717 Humboldt Ave., North, Brooklyn Park, MN 55444, Bruce.Olson@afmg.eu), and Ana M. Jaramillo (AFMG Services North America LLC, Brooklyn Park, MN)

The reverberation time and other important acoustic parameters of a space are based on the structure of the reverberant field. If the absorption is distributed evenly across the surfaces of the room and the room geometry is approximately convex, a homogeneous and isotropic sound field can be assumed that allows for use of statistical room acoustics according to Eyring/Sabine. However, for acoustically complex spaces, raytracing simulations need to be carried out which consider the geometrical details of the room configuration and the spatial distribution of absorption. Simulated, location-dependent impulse responses are then used to extract all relevant acoustic parameters for the room. When developing complex acoustic models, it is often important to calibrate the model. This process involves adjusting the acoustic properties of modeled materials within their range of uncertainty to match the calculation with the measured reverberation time of the real room and possibly other parameters. This is typically an iterative process in which a balance must be found between convergence speed and final accuracy. In this paper, we discuss the practical conditions for when raytracing simulations must be used instead of the simpler Eyring/Sabine approach. Our analysis is supported by theoretical considerations and application examples. We also discuss related errors and quantify uncertainties in a basic way. For the raytracing simulation, tuning absorption coefficients in comparison to scattering coefficients is considered as well.

10:00–10:15 Break

10:15

4aCA4. From rays back to waves—Models and challenges for room impulse response synthesis. Lukas Aspöck (IHTA, RWTH Aachen Univ., Aachen 52074, Germany, las@akustik.rwth-aachen.de), Michael Vorlaender, and Janina Fels (IHTA, RWTH Aachen Univ., Aachen, Germany)

In room acoustics, ray tracing algorithms and their variants are widely used to predict a room's energy decay. With the purpose of creating room auralizations, a room impulse response (RIR) is subsequently synthesized from the simulated decay. This synthesis can account not only for energy decay information but also for the rays' directions of arrival and geometric data of the room. To model the reflection density, this process also requires data such as the room's volume, the mean free path length or information of the boundary conditions. While existing models for this synthesis process can model RIRs adequately for many room situations, relevant deviations can be observed in more special situations such as in semi-open spaces, in the transition area between rooms and outdoor environments or in the vicinity of heavily diffusing surfaces. This work presents the process of generating RIRs from ray tracing results and discusses model adjustments that improve robustness for these challenging scenarios.

Contributed Paper

10:35

4aCA5. Ray-tracing investigation of atmospheric effects on measured noise variability during Starship launches. Nathan F. Carlston (Phys. and Astronomy, Brigham Young Univ., Campus Dr. N283 ESC, Provo, UT 84602, nfcarlston1@gmail.com), Kent L. Gee, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Accurate characterization of launch-to-launch variability in rocket noise is essential for interpreting acoustic measurements and improving predictive models. Two measurement campaigns conducted during Starship Flight 5 and Flight 6 observed notable differences in recorded levels in some regions, despite similar vehicle configurations and flight trajectories. Identifying the sources of this variability is necessary to distinguish vehicle-related effects from propagation effects driven by atmospheric conditions. An in-house ray-tracing code was used to model acoustic propagation along the rocket trajectories for both launches. Atmospheric profiles were obtained from a public radiosonde database, using weather balloon data within approximately one hour of launch. The ray-tracing results account for a portion of the observed differences between the two measurement campaigns, indicating that weather-related propagation effects contribute measurably to launch-to-launch variability. Notably, these effects are observed even at low altitudes and short-range distances, regimes in which acoustic propagation is often assumed to be relatively insensitive to atmospheric variability.

10:55–11:15

Session 4aEA

Engineering Acoustics: General Topics in Engineering Acoustics

Robert D. White, Cochair
Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155

Osama S. Tayel, Cochair
Mech. and Matr. Eng., Univ. of Cincinnati, 201 Klotter Ave. C3, Cincinnati, OH 45219

Contributed Papers

10:30

4aEA1. Lift and flow response of an airfoil to targeted high-intensity acoustic forcing. David E. Mayo (Appl. Res. Lab., Penn State Univ., 800 Stratford Dr., Apt 23, University Park, PA 16802, dem5987@arl.psu.edu), Riley K. Soule, and Zachary Berger (Appl. Res. Lab., Penn State Univ., University Park, PA)

The proliferation of unmanned aerial systems in modern conflicts has introduced the need to develop cost-effective counter unmanned aerial systems (cUAS) technologies, which use directed energy devices capable of disabling small fixed-wing or rotorcraft UAS platforms. In this presentation, the researchers aim to show the potential of using a long-range acoustic device (LRAD) to target the boundary layer over an airfoil, with the goal of increasing boundary layer separation over the foil to shift the center of lift to induce flight control instability and stall. Hot-wire anemometry, load cell measurements, and qualitative flow visualization will be used to observe changes in the steady and unsteady lift and drag of the airfoil, in addition to changes in the flow field around the airfoil, with and without external acoustic excitation. [Work prepared under ONR Contract N00024-25-F-8363.]

10:45

4aEA2. Towards acoustics-oriented aircraft design for highly integrated transport aircraft. Sabine C. Langer (Inst. for Acoust. and Dynam., TU Braunschweig, Langer Kamp 19, Braunschweig 38108, Germany, s.langer@tu-braunschweig.de)

A promising approach to increasing the efficiency of transport aircraft is the tight integration of the engine and airframe. While climate-impacting emissions remain central environmental concerns, aircraft noise has become a critical factor for crew and passenger health, community acceptance, and competitive positioning. Assessments for both external and internal noise depend on advanced models and simulation frameworks capable of representing the underlying physical processes with sufficient fidelity. It is essential to trace the complete "acoustic chain" in aircraft noise studies, from the sound generation of tonal and broadband noise at the source, through all propagation paths, to the immission. Hence, as in general product design, aircraft design also poses new challenges for acoustic engineers: Acoustic engineering is facing a paradigm shift—from isolated noise reduction measures to an integrated, acoustics-oriented design approach. This presentation outlines the prerequisites for embedding acoustics-oriented design into the aircraft development process. Adopting a system-of-systems perspective, we demonstrate how acoustics-oriented design connects to broader technology assessment. The presentation addresses how accurate modelling, reliable simulation, and insight-driven assessment pave the path for acoustics-oriented aircraft design for highly integrated transport aircraft.

11:00

4aEA3. Results from two long duration stratospheric balloon flights of a sonic anemometer. Robert D. White (Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155, r.white@tufts.edu), Rishabh Chaudhary (Mech. Eng., Tufts Univ., Medford, MA), Ian Neeson (VN Instruments, Brockville, ON, Canada), Don Banfield (NASA Ames Res. Ctr., Mountain View, CA), and Chris Yoder (NASA Wallops Flight Facility, Mooresville, NC)

Sonic anemometry allows high-speed, high-resolution measurement of wind fields using measured differences in the acoustic time of flight. In recent years, our group has developed a sonic anemometry system targeted at operation in low pressure (<20 mbar air, <10 mbar CO_2) and low temperature (−100 to 20 °C) atmospheres such as the surface of Mars or on stratospheric balloons. We have demonstrated 7.4 S/s measurement rates with a resolution (in quiet environments) of 1.5 cm/s. Here, we describe the results of two long duration stratospheric balloon test flights. The first flew out of McMurdo station, Antarctica from December 21, 2024 to January 2, 2025 (11 days in full sun) at an altitude between 35 and 38 km (∼4.5 mbar, 0 to 20 °C). The second flew from Wanaka, New Zealand to Chubut Province, Argentina from May 3, 2025 to May 13, 2025 (9 days, day and night) at an altitude near 33 km (5.8 to 8.3 mbar, −60 to 20 °C). Three-dimensional relative wind was successfully recorded throughout both flights. Relative wind speeds are generally less than 3 m/s, with radial and circumferential components faster and more variable than vertical winds. The Brunt–Väisälä buoyancy frequency is apparent.

11:15

4aEA4. Optimization of foster networks for acoustic propagation in thermoviscous Cylindrical Tubes. Ying Fu (Graduate Program in Acoust., Penn State Univ., 414 Eng. Collaborative and Res. Education Bldg. (ECoRE), University Park, PA 16802, yqf5143@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Modeling thermoviscous wave propagation in cylindrical tubes requires accurate time domain representations, particularly for broadband transient and system level acoustic simulations. The Foster networks, consisting of parallel RL and GC branches, provide compact, passive, and physically realizable approximations suitable for time domain simulation. However, the frequency range over which accuracy is required is not universal, but depends strongly on tube geometry and system level objectives. Motivated by this observation, this paper optimizes the Foster network representations of individual tube segments over targeted, physically relevant frequency ranges defined by the corresponding tube geometries. A frequency-weighted nonlinear least-squares approach is used to minimize complex residuals

while prioritizing accuracy within the selected bandwidth. The optimized networks exhibit substantially improved low-frequency performance while maintaining reasonable agreement outside the targeted range. These results demonstrate that specific bandwidth Foster optimization provides a flexible and efficient framework for time domain modeling of thermoviscous cylindrical tubes across diverse applications.

11:30

4aEA5. Time domain modeling of acoustic propagationin in a tube of any cross section. Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., University Park, PA 16802, sct12@psu.edu)

Recent work by several authors has described the use of analog circuit models that provide an approximation of arbitrary accuracy for acoustic propagation in tubes of constant cross section for several simple cross-sectional shapes. To date, the available shapes are those for which the analytical expressions for the series impedance and parallel admittance in the analog circuit can be written as a continued fraction expansion. For these cases, either a Cauer or Foster network configuration can be developed. Many useful tube shapes do not have this structure. The present work provides an alternative method to derive an analog circuit approximation for any tube shape regardless of the complexity of its analytical expressions. The method may also work for some shapes for which an analytical expression is not available.

11:45

4aEA6. Analysis of sound transmission loss in elements composed of polyurethane. Isabella T. Macedo (Mech. Eng., UFPR, Curitiba, PR, Brazil) and Alexandre P. Sarda (Mech. Eng., UFPR, Universidade Federal do Paraná, Avenida Coronel Francisco Heráclito dos Santos, No. 100, Curitiba, Parana 81531-980, Brazil, pescador@ufpr.br)

With the substantial growth of unwanted noise in modern society, the investigation of sound propagation mechanisms has become essential for the development of effective noise control strategies. Within this framework, sound transmission loss (TL) constitutes a fundamental parameter in the assessment of acoustic insulation performance, as it quantifies the capacity of structural elements to attenuate sound transmission, thereby contributing to acoustic comfort. Accordingly, this study experimentally investigated the TL of a medium-density fiberboard (MDF) panel incorporating three circular openings with a diameter of 100 mm under three distinct operating conditions: unsealed openings, openings sealed with rigid MDF plugs, and openings filled with flexible polyurethane (PU) foam. All measurements were carried out in the Vibration and Noise Laboratory at the Federal University of Paraná using a two-room transmission suite comprising a source chamber and a receiving chamber, in compliance with ISO 10140 standards. The highest transmission loss values were achieved with rigid MDF sealing, while the PU foam filling exhibited intermediate performance, limiting sound transmission to some extent through internal energy dissipation. Therefore, this work highlights the importance of evaluating sound transmission loss in systems with structural discontinuities and demonstrates the need for appropriate material selection when aiming to control noise in practical engineering applications.

Session 4aED

Education in Acoustics: SURIEA–Past and Future

Peggy Nelson, Cochair
Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455

Kimberly A. Riegel, Cochair
Phys., Farmingdale State College, 2350 Broadhollow Rd., Farmingdale, NY 10803

Invited Papers

9:00

4aED1. Breaking barriers in acoustics: Summer undergraduate research or internship experience in acoustics. Tyrone M. Porter (Biomedical Eng., Univ. of Texas at Austin, 107 W Dean Keetom St., TX 78712, tmp6@utexas.edu) and Keeta Jones (Acoust. Society of America, Melville, NY)

In June 2020, ASA members participated in a virtual meeting as part of the #ShutDownSTEM effort to identify systemic barriers hindering greater racial diversity in acoustics. Lack of access to academics and practitioners in acoustics was identified as a key factor in restricting the awareness of racial minorities to opportunities in the field. In response, the ASA established the Summer Undergraduate Research Internship or Experience in Acoustics (SURIEA) program to benefit both the mentors and the mentees. At the start of the program, mentees complete a week-long intensive short course that covers acoustic fundamentals and soft skills for navigating a professional work environment. The cohort bonds at the short course and maintains a connection throughout the summer via virtual meetings facilitated by graduate student coordinators. Mentors receive coaching on supporting mentees with minimal acoustics backgrounds and whose identities are historically underrepresented in the field. To date, 54 undergraduate students from a range of institutions have participated in the program. Survey results show increased awareness of acoustics and the ASA, with many students expressing interest in graduate study and careers in the field. Now entering its sixth year, SURIEA has become a signature ASA program.

9:20

4aED2. From SURIEA to SMMILE: Building inclusive pathways into acoustics-related fields. Kirby Parnell (Hawaii Inst. of Marine Biology, Univ. of Hawaii Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, keparnel@hawaii.edu) and Brijonnay Madrigal (Univ. of California San Diego, La Jolla, CA)

In this presentation, we reflect on our experiences serving as the first co-coordinators of SURIEA during its inaugural summers in 2022 and 2023. We highlight post-program pathways of several SURIEA alumni, sharing updates on their academic, research, and professional trajectories following participation. By revisiting SURIEA from the perspective of early program leadership and tracking its downstream impacts, this talk illustrates how SURIEA has functioned not only as a formative training experience, but as a catalyst for sustained engagement in acoustics for both program alumni and co-coordinators. SURIEA's structure, values, and community directly inspired the development of a complementary education initiative: the Summer Marine Mammal Intensive Learning Experiences (SMMILE). Drawing on core SURIEA elements—small cohorts, mentorship, hands-on training, and community-building—SMMILE is an immersive marine mammal science program rooted in experiential, place-based learning in Hawai'i. Its overarching goal is to expand access to acoustics-related fields for high school students from historically marginalized communities. Building on SURIEA's model, SMMILE adapts these elements for a younger audience, lab and field environments, and exposure-focused learning goals. Most recently, SMMILE is preparing to expand to California with support from the Acoustical Society of America (ASA), illustrating how SURIEA's impact continues to propagate through ASA-supported education efforts.

9:40

4aED3. Acoustics short course—Lessons learned from formal and informal instructions across disciplines. Adrian K. C. Lee (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, akclee@uw.edu), Eric Reuter (Reuter Assoc., LLC, Portsmouth, NH), and Kimberly A. Riegel (Phys., Farmingdale State College, Farmingdale, NY)

The field of acoustics crosses many traditionally defined disciplines, ranging from engineering and Earth sciences to life sciences and the liberal arts. Designing a ramp-up course for students with different academic backgrounds and prepare them for summer internships that span all these disciplines can be challenging. In the past 5 years, SURIEA organized a ramp-up course with three instructors covering a wide-ranging teaching content along with two teaching assistants helping mentor these undergraduates who have limited research / industry experience prior to joining the program. This shortcourse started during the pandemic-Zoom era but transitioned to a week-long instruction in a residential setting to create an immersive learning experience. In this talk, we will share lessons learned about what content area was selected to adequately prepare these trainees for their summer internship projects and what other informal learning opportunities that help trainees grow to be more confident in their eventual placements. Factors that would make this program more

sustainable in the future as well as how this course could be replicated in other settings (e.g., at other higher education institutions or in different global regions) to strengthen the pipeline for the acoustics workforce will also be discussed.

10:00

4aED4. Reflections on mentorship from graduate co-coordinators of the SURIEA program. Victoria A. Sevich (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, sevich.1@osu.edu) and Prajna Jandial (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Over the past five years, the Summer Undergraduate Research or Internship Experience in Acoustics (SURIEA) program has provided many undergraduate students with opportunities, resources, and confidence to pursue independent research or industry projects in acoustics. In addition to these undergraduates, the SURIEA program has also had a lasting impact on the graduate students who served as co-coordinators for each SURIEA cohort. In this talk, past and current SURIEA co-coordinators will share insights from their experiences working with the SURIEA students over the years and highlight the ways in which these experiences have shaped their mentoring strategies and career trajectories.

10:20

4aED5. Experiences of SURIEA students. Peggy Nelson (Ctr. for Appl./Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, nelso477@umn.edu) and Kayla Pierre (Montclair State Univ., Roselle Park, NJ)

Summer internships can change student's trajectories and have lasting impacts on student's success. In this presentation, we will present quotes from interviews with recent SURIEA students who reflect on their SURIEA experience. Individual stories will be reported. Student-generated ideas for the future will be discussed.

Contributed Paper

10:40

4aED6. Which emotional sounds captured attention? Insights on auditory emotional perception and processing in anxiety. Deanna V. Sharpe (Emory Univ., 201 Dowman Dr., Atlanta, GA 30322, deannas0730@gmail.com) and Mishaela DiNino (Univ. at Buffalo, Buffalo, NY)

Serving as a research intern at the University at Buffalo through the SURIEA 2025 Program was a formative experience that deepened my research and technical skills and prepared me for graduate pursuits. My biggest takeaway from this program was the opportunity to engage with an independent research project in a highly collaborative environment, while receiving regular guidance and support. I valued learning from my peers and mentors as I assisted with otoscopy, pure tone audiometric threshold testing, auditory brainstem testing, statistical analysis, and online behavioral experiment design, which equipped me for my SURIEA project. My project explored how young adults with different levels of anxiety symptomatology attend to stimuli in response to emotional or neutral stimuli. Although we initially predicted that people with higher anxiety scores would be distracted more easily by stimuli with emotional prosody, they were actually better able to attend to and ignore emotional sentences than people with lower anxiety symptoms did. This suggests that those with generalized anxiety were more likely to exhibit attention biases, congruent with visual literature findings. Overall, this research experience gave me insight into collaborative research and enhanced my adaptability as we further developed the study.

Session 4aMU

Musical Acoustics: Parameter Estimation for Musical Acoustics

Vasileios Chatziioannou, Cochair
Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna, 1030, Austria

Sarah R. Smith, Cochair
Elec. and Comput. Eng., Univ. of Rochester, 617 Comput. Studies Bldg., 160 Trustee Rd., RC 270231, Rochester, NY 14627

Invited Papers

9:00

4aMU1. Estimating wood density from X-ray computed tomographic scans. Mark Rau (Music and Theater Arts & Elec. Eng. and Comput. Sci., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Bldg W18-Rm., Cambridge, MA 02139), Jared Katz (Museum of Fine Arts, Boston, Boston, MA), and Benjamin Sabatini (Dept. of Mater. Sci. and Eng., Massachusetts Inst. of Technol., Cambridge, MA)

X-ray computed tomography (XCT) is increasingly used to study musical instruments. Computed tomography (CT) provides both the 3-D geometry of an object and relative density information. Quantitative CT densitometry is the process of measuring density by comparing the Hounsfield units (HU) or gray values of a CT-scanned object to calibrated standards. An initial investigation using a set of wood calibration standards to estimate the density of various parts of wooden stringed instruments is presented. The CT scans were made with a Lumafield Neptune commercial scanner. Violins were the target instrument group, so the calibration standards were constructed out of spruce, maple, and ebony. The CT densitometry process was tested with a set of known density woods to validate its accuracy.

9:20

4aMU2. Timbral feature characterization of the Caribbean steelpan. Andrew C. Morrison (Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431, amorriso@jjc.edu)

Although the distinct sound of the Caribbean steelpan is quickly recognized by ear, a definitive characterization of the timbre of this unique musical instrument remains elusive. The timbre of the steelpan depends in part on the vibration's time evolution after being struck by the player's stick. The vibration is usually dominated by the lowest three partials of the struck note, the sympathetic vibrations of other notes in the pan, and the vibration of the pan as a whole. The steelpan is usually tuned such that the lowest three partials of each note are harmonically related. A set of custom steelpans was commissioned to help better understand the effect of the sympathetic vibrations on the timbre of the instrument. A steelpan with a single note and another with a note plus its octave are compared to a similar steelpan with a complete set of notes in a traditional layout. Strikes on the steelpans using different sticks and dynamic levels were analyzed to determine which spectral features most contribute to the timbre of the steelpan.

Contributed Papers

9:40

4aMU3. Estimating envelope parameters for steelpan partials. Randall Ali (Music and Media, Univ. of Surrey, Guildford, United Kingdom) and Andrew Christian (Flight Vehicle Acoust. Branch, NASA Langely, 2 N. Dryden St., Hampton, VA 23681, andrew.christian@nasa.gov)

The steelpan is a percussive idiophone that originated in Trinidad in the early 20th century. Its sound is generated by striking an arrangement of notes etched into the sunken top of a steel oil drum. The arrangement of the notes on the pan appears to be critical to the timbre of the instrument, as it impacts the coupling strength between notes: If a note has a neighbor tuned an octave and fourth above, then the third overtone of the note is likely to have a ratio of 2.667 rather than the harmonic 3.0. Further, the temporal envelopes of the partials depend on the coupling strength, with lower-coupled partials having audible lags relative to the fundamental. To enable psychoacoustic testing of how these idiosyncrasies give the pan its "flava," a parameterization is needed that can morph between tone complexes that have these features, and those of a more typical instrument in which the overtones are essentially harmonic and have similar temporal envelopes. This presentation discusses a model that includes variable delay, onset, and decay times per partial (along with frequency and magnitude). It also describes methods to fit these parameters to recordings of both the steelpan and more traditional instruments.

9:55

4aMU4. Harmonic chime design using finite element analysis. Therese Malinowski (Purdue Univ., 401 N. Grant St., West Lafayette, IN 47907, theresemalinowski@gmail.com), Noori Kim, and Mark French (Purdue Univ., West Lafayette, IN)

The tubular bell chime is an orchestral percussion instrument with a naturally inharmonic frequency spectrum. Inharmonic overtones can cause perceptual discomfort to listeners by stimulating an overlapping cochlear region, but spectral tuning for perceptual optimization has not previously been explored for tubular bells. The research uses finite element analysis to solve the inverse spectral problem of designing a harmonic tubular bell by optimizing locations of diameter reductions to minimize spectral frequency-matching errors with a harmonic series. An initial design was generated using constrained minimization within Timoshenko beam theory. Then, the design was refined using gradient descent of simulated three-dimensional finite element analysis. The final design was prototyped using a manual lathe, and experimentally characterized via frequency response measurements. Prior to modification, the stock material had overtones up to the fifth mode between 24 and 158 cents from a fundamental-based harmonic series. Following shape optimization, the first five resonant frequencies of the manufactured chime were between 3 and 12 cents away from the target harmonic series, showing a substantial perceptible increase in harmonicity. The results prove that applied local mass and stiffness changes can modally tune a tubular bell, unlocking a new degree of musical freedom for chime-type instruments.

10:10

4aMU5. Modal analysis of a liberty bell replica. Sean Collier (Penn State Appl. Res. Lab., 218 Appl. Sci. Bldg., University Park, PA 16804, smc604@psu.edu), Jonathan Young, Aaron Stearns, and Eric Brown (Penn State Appl. Res. Lab., State College, PA)

Scant documentation exists for a bell so emblematic of the American ideology of freedom—its sound silenced by the infamous crack well before any of us were born. While it is true that computational tools exist to predict its sonance, the Liberty Bell poses a particularly challenging task due to the specific nature of "bell bronze" and the lack of recorded dimensions. Hence, modal analysis can be used to estimate the natural frequencies of the bell, so to tune the numerical model for acoustic prediction. To that end, this talk discusses the modal analysis of a Liberty Bell replica located at Penn State Behrend. In particular, we explore the planning, practical conversations, testing, and modal post-processing of the replica. Methodologies will be emphasized, exploring expansion techniques to increase spatial fidelity and frequency-domain filtering to limit errors caused by the long ringdown. Results presented in this talk describe the vibration characteristics of the replica, a comparison to classical results of Rossing and Perrin, and the story of how three enthusiastic acousticians worked to recreate the historic chime.

10:25–10:40 Break

10:40

4aMU6. Beyond cost and crisis: Towards an accessible and sustainable marimba sound. Amartya Bhattacharya (Wayzata High School, 716 Columbus Ave., Boston, MA 02120, amartya.bhattacharya007@gmail.com)

The marimba is a tuned idiophone (percussion instrument) whose keys are primarily made from Honduran rosewood—a rare resource—making the instrument expensive and inaccessible to many, including low-income individuals, school districts, and students. Building on existing materials research, this study systematically compares American wood species and synthetic materials as sustainable, cost-effective alternatives to Honduran rosewood with an emphasis on preserving musical quality. American hardwoods, softwoods, and polymers were evaluated for their density (ρ), Young's modulus (E), and loss factor (η). Density indicates bar durability, Young's modulus governs pitch and feel, and the loss factor determines the resonance of the bars. Initial analysis revealed that while no tested materials met pre-defined criteria based on rosewood reference values, cherry and hickory demonstrated promising potential, with thermal densification offering a viable pathway to enhance their damping performance. This research contributes foundational data for exploring more accessible and environmentally responsible materials for marimba construction, potentially broadening participation and ensuring the instrument's future.

10:55

4aMU7. Towards a robust estimation of the coherence parameters of musical signals. Sarah R. Smith (Elec. and Comput. Eng., Univ. of Rochester, 617 Comput. Studies Bldg., 160 Trustee Rd., RC 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

Adapting methods from optical coherence theory holds significant promise for the analysis of musical signals; however, reliably estimating audio coherence time is challenging in practice. In particular, audio coherence parameters are sensitive to the acoustic environment, and the typical coherence times of audio signals (on the order of tens to hundreds of milliseconds) are comparable to both room decay times and analysis frame lengths. This work introduces a set of estimators for audio coherence that can be tracked over time, while remaining robust to small variations in recording conditions or analysis parameters. The proposed framework could provide a common basis for describing how perceptual parameters, such as sound localization and pitch estimation, vary with source characteristics and listener position. The metrics are demonstrated using binaural signals generated from anechoic sources and head-related transfer functions, followed by examples illustrating the effects of small to moderate reverberation in the monaural case.

11:10

4aMU8. Physical factors that affect violin sound and performance: Part 1. Gordon P. Ramsey (Phys., Loyola Univ. Chicago, Chicago, IL 60660, gramsey@luc.edu) and Katie Bennett (Phys., Loyola Univ. Chicago, Chicago, IL)

Violinists have considerable choices when selecting the style of playing to create a particular sound. Two of these include (1) the placement and angle of the bow on the string and (2) the use of mutes to modify the sound. We investigated the spectra and sounds for bowing at three different locations along the string and for two sections of the bow. Bowing near the bridge enhanced spectral overtones while increasing the intensity. Exciting near the fingerboard yields a softer, purer sound, while bowing at the normal position in between yields the average. Violin mutes not only reduce the intensity of the sound but also act as frequency filters, due to the decreased bridge oscillation. We tested a concert mute and a "cash currency" mute often improvised by players. Most of them acted as low pass filters while lowering the sound intensity. Our data indicate some of the key factors that create different sounds created by violinists.

11:25

4aMU9. Physical factors that affect violin sound and performance: Part 2. Gordon P. Ramsey (Phys., Loyola Univ. Chicago, Chicago, IL) and Katie Bennett (Phys., Loyola Univ. Chicago, Chicago, IL 60660, kbennett@luc.edu)

For violinists, one of the greatest decisions one can make is choosing a bow. There is a need for more research on bow properties, including materials and amounts of rosin used. Despite this lack of empirical evidence, the differences in sound produced from variations of such are obvious to musical experts. Data were taken from a one-piece-backed violin using a standard-quality wooden bow and a similar-quality carbon fiber bow. The spectra were examined for various amounts of rosin on the wooden bow, as well as the differences in sound for the wooden and carbon fiber bows. A "standard" amount of rosin was proved to be favorable due to continuity in harmonic structures of the spectra. The carbon bow produces a less-organized spectra compared to the wooden bow; along with the possibility of the introduction of bow resonances into the tone and quality of sound produced by the actual instrument.

11:40

4aMU10. Modal parameter estimation for simulating cello bridge admittance. Coline Girard (École normale supérieure Paris-Saclay, Paris, France) and Vasileios Chatziioannou (Dept. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Vienna 1030, Austria)

When simulating string vibration in musical instruments, such as guitars or cellos, it is customary to use simply supported boundary conditions at both ends of the string. The bridge velocity is then obtained via convolution of the force that the string applies to the bridge with the bridge impulse response. The latter can be obtained numerically, e.g., via finite element modeling, or experimentally. On the other hand, it is possible to incorporate the effect of a moving bridge boundary condition directly into the string simulation. This requires obtaining the parameters that govern the oscillations of the bridge. In this work, this is achieved using the filter diagonalization method, which extracts bridge modal parameters from a measured impulse response. These parameters can be used in numerical simulations to predict the motion of the bridge, subject to the vibrations of the string. Using such simulations, the effect of the moving bridge on the string oscillations is analyzed for the case of plucked and bowed strings.

THURSDAY MORNING, 14 MAY 2026 SALON D, 8:30 A.M. TO 11:55 A.M.

Session 4aNS

Noise and Architectural Acoustics: Methods in Soundscape Assessment and Design

Bennett M. Brooks, Cochair
Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062

Brigitte Schulte-Fortkamp, Cochair
HEAD Genuit Foundation, Ebert Straße 30 a, Herzogenrath, 52134, Germany

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

Chair's Introduction—8:30

Invited Papers

8:35

4aNS1. Outlining the purpose and goals of soundscape intervention and soundscape design. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and Brigitte Schulte-Fortkamp (HEAD Genuit Foundation, Herzogenrath, Germany)

This paper presents practical guidance on how to systematically identify, structure, and evaluate soundscape interventions and new designs in real-world contexts. Drawing on established definitions, a soundscape intervention is defined as any site-specific implementation, process, outcome, or installation derived from soundscape design that aims to preserve or improve the acoustic environment. A soundscape intervention design is defined here as a structured plan or specification to preserve or enhance an existing acoustical environment. A new soundscape design creates an acoustical environment that provides a desired perceptual outcome in a new use for a building site, neighborhood, or city. Interventions and new designs are developed through co-creation, a collaborative design process that integrates knowledge and perspectives from local experts, practitioners, and other soundscape stakeholders. Part 4 of the ISO 12913 soundscape standard is currently under development, with a key objective being the operational integration of soundscape approaches into urban planning and building design practice. This paper focuses on this objective by positioning the soundscape method as a functional and actionable planning tool, with practical examples. The means by which soundscape interventions and designs can be embedded within existing city planning and building design workflows and decision-making processes are discussed.

8:55

4aNS2. An emerging process for urban soundscape investigations. Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein (Siebein Acoust., Gainesville, FL), Marylin Roa (Siebein Assoc., Gainesville, FL), Gary Siebein Jr. (Consulting, Siebein Assoc. Inc, Gainesville, FL), and Martin Gold (School of Architecture, Univ. of Florida, Gainesville, FL)

Many urban communities in the United States struggle to achieve a balance among people moving back into cities, commercial activities, and entertainment venues that people find desirable in live, learn, work, and play environments. Dynamic documentation, analysis, design strategies, codes, and enforcement activities are required to achieve a balance in rapidly evolving, sustainable cities. Active engagement of the full range of participants in the acoustical communities in each case was essential in understanding and addressing soundscape design challenges. Multiple meetings with the dwellers in the soundscape in organized, community fora, focus groups, and discussions with individuals provided ways for all involved to understand the points-of-view of others and the blocks to work towards consensus. Soundwalks with community members were used to understand the issues in the specific context where they occurred. Subsequent soundwalks at multiple times of day and night allowed immersion of the team in the sounds and life of the cities. Simple, but sophisticated, measurement and modeling of the soundscapes were necessary elements of the method. Case studies in three communities are presented, including goals, challenges, methods, results, and proposed solutions to the variety of acoustical issues encountered in the urban contexts.

9:15

4aNS3. Urban soundscape design: Possible strategies and solutions. Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein (Siebein Acoust., Gainesville, FL), Marylin Roa (Siebein Assoc., Gainesville, FL), Gary Siebein Jr. (Consulting, Siebein Assoc. Inc, Gainesville, FL), and Martin Gold (School of Architecture, Univ. of Florida, Gainesville, FL)

Case studies of soundscape projects in two cities are used to demonstrate methods and potential interventions that can be applied in these environments. Both cities had initially described acoustical issues related to loud vehicles and amplified music disturbing residents and evading enforcement policies. Interviews with stakeholder groups, including residents, city officials, business owners, and others; individually, in small focus groups and in larger public fora; soundwalks at multiple times of day and night; long-term and short-term acoustical measurements, and mapping were used to identify itineraries, calendars, taxonomies, and acoustical communities involved in the soundscapes of the cities. Conceptual structures, acoustical rooms, and sonic niches were identified. Strategic computer model studies were used to explore potential solutions and stimulate dialogue among soundscape participants. Student design projects explored imaginative possibilities for future development. These soundscape elements were transformed into planning interventions to reduce, buffer, and mitigate the unwanted sounds, preserve and enhance the desired sounds, and add multi-tiered interventions to expand the social and cultural richness of the cities, including infrastructure such as program, verticality, texture and material; administrative such as ordinance development and policies; and operational controls such as continued public engagement and education as methods to move forward.

9:35

4aNS4. Intervention results of a soundscape study to develop a new noise ordinance. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com)

A follow-up soundscape study was conducted to assist with the development of a new noise ordinance in the City of Delray Beach, Florida. Delray Beach has an active entertainment district that is popular with both tourists and locals. Integrated into the district and the surrounding area are recent residential and business developments. During the pandemic, activity in the entertainment district decreased. Since then, activity has resumed, causing increased noise complaints to the city. A soundscape study was conducted to determine the existing conditions and to develop data-driven, perception-based objective standards. Goals assigned for the new ordinance were to create certainty for stakeholders, balance the interests of businesses and residents, manage expectations, and ensure fairness and consistency in application. Long-term and short-term acoustical measurements were made, several public workshops and hearings were held, and two soundwalks were conducted. A new ordinance was drafted and is in the final stages of implementation. The outcome of this effort will be presented.

9:55

4aNS5. Dependency of results on chosen soundscape scales in the assessment of auditory perception by cross-comparison of scales in an indoor office space. Merve Eşmebaşı (Architecture, National Univ. of Singapore, Singapore, Singapore) and Siu-Kit Lau (Architecture, National Univ. of Singapore, 4 Architecture Dr., NUS, Singapore 117566, Singapore, slau@nus.edu.sg)

In this study, the 2019 and 2025 versions of the standardized soundscape scale, along with the suggested indoor soundscape scales in the literature for residential and office spaces, are used in combination. Although the ISO and indoor soundscape scales are highly correlated, their relations with other perceptual attributes, such as preference, satisfaction, and appropriateness, exhibit statistically significant differences, especially for the Eventfulness and Content axis from the ISO and indoor soundscape scales, respectively. Pleasantness is evaluated as more sensitive than the Comfort axis on the indoor scale. These results highlight the influence of the chosen scales on the interpretation of perceptual results. The evaluated indoor soundscape condition is an actual office space with open windows and birdsong masking, presented at 45–55 dBA. The suggested circumflex coordinate analysis in the recent ISO/TS 12913:3-2015 version indicates that birdsong effectively masks intrusive traffic noise in open office space, provided it does not increase the overall sound level by more than 3 dBA. This study underscores the importance of the selected scales for the results obtained and contributes to the semantic assessment of the proposed soundscape scales by revealing the differences they elicit.

10:15–10:30 Break

10:30

4aNS6. Changes to rural and suburban soundscapes due to data centers and natural gas power plants. David S. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

The current boom in the creation of data centers and their associated natural gas power sources to support them impacts the local soundscape. Rural residential and suburban areas have unique soundscapes, in some cases where the A-weighted sound levels do not change significantly with the introduction of data centers and power plants, but the soundscape can be radically reshaped. Several examples will be discussed.

Contributed Papers

10:50

4aNS7. Beyond odor intensity: Assessing sensory congruence in the restorative soundscape of Beitou thermal valley. Yosua W. Tedja (Architecture, National Taiwan Univ. of Sci. and Tech., 43 Keelung Rd. Sec. 4, Taipei 10607, Taiwan, yosuatedja@gmail.com), Lucky S. Tsaih, Khaing Thinzar, Veerin Udomsopakit, Sanh T. Diep (Architecture, National Taiwan Univ. of Sci. and Tech., Taipei, Taiwan), and Shiang-I Juan (Landscape Architecture, Chung Yuan Christian Univ., Taoyuan, Taiwan)

Standard soundscape assessment protocols often isolate acoustic factors, yet real-world experiences are inherently multisensory. This study presents a mixed-methods assessment strategy to investigate cross-modal interactions in Beitou Thermal Valley, a geothermal site characterized by a dominant waterfall soundmark and distinct sulfurous odors. The assessment protocol triangulated *in situ* acoustic measurements with a subjective field survey ($N = 34$) to evaluate how non-acoustic factors modulate soundscape perception. Objective profiling established the acoustic baseline using repeated 5-min equivalent continuous sound level measurements (mean $L_{Aeq,\ 5min}$), identifying a mean level of 61.6 dB with a characteristic mid-frequency peak at 630 Hz. Subjective data, collected via 7-point Likert scales, assessed odor intensity, soundscape quality, and the specific construct of smell–sound congruence. Spearman's rank correlation analysis revealed that while physical odor intensity did not correlate with sound pleasantness ($\rho = -0.07$), the perceived congruence between smell and sound was a strong predictor of both sound enjoyment ($\rho = 0.70$) and overall restorativeness ($\rho = 0.65$). These findings validate an assessment model that accounts for sensory fit rather than isolated stimuli. Consequently, the study proposes a design principle for restorative environments: practitioners should prioritize multisensory coherence over simple noise or odor reduction, ensuring that olfactory signals conceptually align with the acoustic context.

11:05

4aNS8. Survey: Making hospitals quiet. Joanne Solet (Harvard Med. School, The Cambridge Hospital 1493 Cambridge St., Cambridge, MA 02139, Joanne_Solet@HMS.Harvard.edu)

The Making Hospitals Quiet (MHQ) survey presents multiple categories for review, assessment, and intervention to enhance healthcare facility acoustic environments. Goals include the improvement in communication accuracy and privacy, limiting medical errors, ameliorating staff fatigue and burnout, and protecting patient sleep. In the current climate of austerity and uncertainty in healthcare finance, this survey is primarily directed toward policy analyses and operational upgrades rather than alterations in facility design or building materials. Assessments and interventions can be undertaken unit by unit and tailored to the needs, specific characteristics, and diagnostic categories of the patient population. Where it is determined that additional interventions are required, for example, due to special patient sensitivities, high intensity of care, or site exterior noise such as helicopter landing pads or ambulance bays, installation of high-performance acoustic ceiling tiles, sound-absorbing flooring, and double pane windows may provide additional protection and be cost effective. Indicators of improvement may include direct sound monitoring, as well as patient quality of care ratings (HCAPS), medical error tracking with outcomes, patient length of stay and readmissions, staff turnover, and exit interviews.

11:20

4aNS9. Characterizing and predicting vessel noise in Glacier Bay National Park to inform soundscape management strategies. Rebecca Cohen (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ritter Hall 101, 8635 Kennel Way, La Jolla, CA 92037, rec022@ucsd.edu), Dimitri Ponirakis (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Christine Gabriele (Glacier Bay National Park and Preserve, Gustavus, AK), Davyd H. Betchkal (Natural Sounds and Night Skies Div., National Park Service, Denali Park, AK), Tania Lewis (Glacier Bay National Park and Preserve, Gustavus, AK), and Aaron N. Rice (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

Vessels are a pervasive source of anthropogenic noise fragmenting natural soundscapes in Glacier Bay National Park, Alaska, USA. More than 90% of human visitors to this largely marine Park travel by vessel, and many areas are accessible only by vessel. To effectively manage Park use and preserve natural soundscapes, there is a need to quantify the effects of vessel speed modulation for management of sound in air and water. Accomplishing this goal entails (1) characterizing source levels of vessel categories present in the park, (2) investigating relationships between source level and vessel/operational characteristics, and (3) developing a manager-friendly coding-free acoustic propagation modeling application to simulate Park soundscapes under various vessel scenarios. In 2024 and 2025, acoustic data were collected from various vessel types in the Park via a partnership with vessel operators and opportunistic data collection. Here, we will present our findings regarding vessel source levels across and within vessel categories and how they relate to vessel/operational factors. As in past studies, vessel speed was found to have a significant positive relationship with source level. The modeling application functionality, which leverages existing, widely accepted acoustic propagation models to run complex vessel scenario simulations, will also be demonstrated.

11:35–11:55 Panel Discussion

Session 4aPAa

Physical Acoustics: Infrasound I

Philip S. Blom, Cochair
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Invited Papers

8:00

4aPAa1. New infrasound applications facilitated by small quick-deploy sensors. Jacob F. Anderson (Geoscience, Boise State Univ., 1910 Univ. Dr., 1535 Geosciences, Boise, ID 83725-1535, jacobanderson152@boisestate.edu), Jeffrey Johnson, Owen Walsh, Madeline Hunt, and Jerry Mock (Geoscience, Boise State Univ., Boise, ID 83725-1535)

Infrasound monitoring is a rapidly growing field with established use for monitoring volcanoes, nuclear tests, and avalanches, and emerging applications to fluvial and wildfire hazards. These hazards are increasingly common and damaging, and non-traditional instruments, new sensor configurations, and recording techniques can help infrasound accomplish these monitoring needs. We highlight the Gem infrasound logger's capabilities, being a small, self-contained, low-power, quick-deploy sensor that is easy to conceal and deploy in large numbers with flexible geometry; it contrasts with conventional infrasound instruments that are generally larger, heavier, more difficult to hide, more time-consuming to deploy, and whose deployment geometry in arrays is limited by fixed cables. Ten years of fieldwork with Gem infrasound loggers has shown the value of this style of instrument in theft-prone or hazardous sites, in rapid-response roles, when conducting spatial surveys of steady infrasound, and in large-N arrays. We summarize research projects related to emerging infrasound monitoring applications and discuss how the Gem infrasound logger has facilitated these studies. Finally, we discuss our development of a new instrument to carry the Gem's advantages into the next decade.

8:20

4aPAa2. Towards a deep learning-based realistic operational tracking of the infrasound IMS network performances. Alice Janela Cameijo (CEA, DAM, DIF, 2, Chemin du Ru, Bruyères-le-Châtel 91680, France, alicam2510@gmail.com), Alexis Le Pichon, Julien Vergoz (CEA, DAM, DIF, Arpajon, France), Youcef Sklab (UMMISCO, Sorbonne Université, IRD, Bondy, France), Souhila Arib (Thema, CY Cergy Paris Université, Cergy-Pontoise, France), and Samir Aknine (LIRIS, Université Lyon 1, Villeurbanne, France)

Global-scale infrasound observations show how the strong spatio-temporal variability of the middle atmosphere impacts the detection capability of the International Monitoring System infrasound network, which has been deployed to monitor compliance with the Comprehensive Nuclear-Test ban Treaty. Past studies used infrasound station- and time-independent statistical noise models, climatological atmospheric specifications, and scaling laws to estimate the minimum detectable explosive yield. More recently, a Bayesian approach has been proposed to predict, in high resolution, the infrasound network probability to detect any explosive yield for near real-time monitoring application. We propose an updated version of this approach, leveraging a deep learning-based infrasound propagation modeling tool. The deep learning algorithm can map realistic atmospheric specifications to range-dependent infrasound transmission losses associated with uncertainty estimates. We demonstrate the algorithm's robustness to new atmospheric scenarios and its consistency with the underlying physics of infrasound propagation via a data-centric explainability framework.

8:40

4aPAa3. The use of modal group velocities for infrasound arrival time modeling. Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrechtseweg 297, Utrecht 3731GA, Netherlands, jelle.assink@knmi.nl) and Roger M. Waxler (Univ. of MS, Univ., MS)

While infrasound localization primarily relies on back-azimuth information rather than arrival time, travel-time modeling remains essential for the correct association and interpretation of infrasound arrivals—and, ultimately, for accurate event characterization. To support this, various celerity models have been developed, generally falling into two categories: (1) empirical models based on ground-truth events, and (2) stochastic models derived from ray-based propagation through ensembles of model atmospheres. To date, most efforts have focused on ray-theoretical approaches (e.g., Blom, 2020). This work explores an alternative framework based on modal group velocities (Lonzaga, 2015). We investigate the possibility that group velocity distributions derived from the vertical-mode distributions can provide reliable estimates of the range of arrival times. A key advantage of this approach is its ability to incorporate

frequency-dependent effects, including diffraction, which are typically neglected in ray theory. These effects become especially relevant in the presence of small-scale atmospheric structure in temperature and wind (i.e., from gravity waves), which can significantly affect infrasound travel times. The theoretical values are compared to traveltimes from ground-truth events.

9:00

4aPAa4. Infrasound waveform comparisons from approximate propagation models. Jordan W. Bishop (Los Alamos National Lab., P.O. Box 1663, Los Alamos, NM 87545, jwbishop@lanl.gov) and Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., Los Alamos, NM)

Numerical models for infrasound propagation have a fundamental tradeoff between physical complexity and computational speed. Well-validated software packages exist to numerically solve infrasound propagation models for a variety of physical approximations, such as geometric acoustics, normal modes, and the parabolic equation. Here, we present time domain waveform predictions for benchmark comparisons between broad-band normal mode reconstruction (NCPAprop *Modess*), reconstruction from the parabolic equation (NCPAprop *ePAPE*), and a finite-difference time-domain solver, termed *FinAc*. The *FinAc* software leverages GPU computing through the Kokkos C++ Performance Portability Ecosystem to solve linearized Euler equations for a horizontally stratified medium. We will examine waveform outputs across regional distances (<500 km) for tropospheric and stratospheric returns.

9:20

4aPAa5. Predicting atmospheric effective sound speed using synthetic infrasound and machine learning. Alex Witsil (Appl. Res. Assoc., 8537 Six Forks Rd., Ste. 200, Raleigh, NC 27615, awitsil@ara.com), Stefan Awender, David Fee (Univ. of Alaska Fairbanks, Fairbanks, AK), Roger M. Waxler (Univ. of MS, Univ., MS), and Mark Leidig (Appl. Res. Assoc., Houston, TX)

We introduce a machine learning (ML) model that reconstructs atmospheric conditions at the location of an explosive source using regionally recorded infrasound. A convolutional neural network (CNN) inputs raw waveforms as well as source–receiver geometries and outputs a predicted effective sound speed profile. The model is trained on synthetic data generated by propagating a source time function through modeled atmospheres out to random station locations. Prediction accuracy is high, with an average root mean squared error (RMSE) of 8 m/s. Importantly, we find that poorly predicted profiles nonetheless reproduce the original waveforms with high accuracy, indicating a nonuniqueness between atmospheric conditions and recorded waveforms. Unlike traditional physics-based inversions that rely on array deployments, the ML model presented herein can analyze single-channel data and, once trained, make predictions within milliseconds. Given the performance on synthetic data, future tests should include real-world data to further assess the benefits of ML-based frameworks for infrasound analysis.

9:40–10:00 Break

10:00

4aPAa6. Infrasound generation by ground shaking from underground explosions: Insights from a multi-physics experiment for low-yield nuclear explosion monitoring. Keehoon Kim (Geophysical Monitoring Program, Lawrence Livermore National Lab., 7000 East Ave., L-103, Livermore, CA 94550, kim84@llnl.gov), David N. Green (AWE Blacknest, Brimpton, United Kingdom), Daniel Bowman (Pacific Northwest National Lab., Albuquerque, NM), Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., Los Alamos, NM), and Jordan W. Bishop (Los Alamos National Lab., Los Alamos, NM)

Infrasound from underground explosions plays an important role for explosion monitoring, particularly in determining the explosion depth. To understand infrasound generation from seismoacoustic coupling, many chemical explosion experiments have been conducted. Previous experimental data showed that gas venting and spall generation at the epicenter play a key role in infrasound generation. However, those detonations were conducted at shallow depths and may not represent large overburied events like nuclear explosions or earthquakes. Here, we report a unique infrasound observation from a large, overburied chemical explosion and describe the seismoacoustic coupling mechanisms. A multi-physics experiment (PE1-A) for low-yield nuclear explosion monitoring was conducted at the Nevada National Security Site in 2023 in support of the US National Nuclear Security Administration. The PE1-A experiment was a fully coupled, chemical detonation with a TNT-equivalent yield of 16.3 tons at a 254 m depth. This overburied explosion produced unusually long-duration infrasound signatures. Physics-based numerical modeling and analysis indicate that the extent of the seismoacoustic coupling area and local topography substantially affect the infrasound waveforms in duration and amplitudes. Seismoacoustic coupling at this depth may be governed by linear elastic-to-acoustic conversion along the surface, and local topography can play a critical role in infrasound generation.

10:20

4aPAa7. Estimating surface explosion yield from single-channel infrasound signals at regional distances with random-forest regression. Stefan Awender (Wilson Alaska Tech. Ctr., Univ. of Alaska Fairbanks, Fairbanks, AK, sawender@alaska.edu), David Fee (Wilson Alaska Tech. Ctr., Univ. of Alaska Fairbanks, Fairbanks, AK), and Alex Witsil (Appl. Res. Assoc., Raleigh, NC)

When detonated on the Earth's surface, large explosions emit distinctive low-frequency acoustic signatures that can be detected by infrasound stations hundreds of kilometers from the source, atmospheric conditions permitting. Here, we present a method to estimate the yield of real surface explosions from regional infrasound signals with a random-forest regressor. The random forest is trained entirely on synthetic data generated using scaled-yield source models, realistic atmospheric specifications, and normal-mode wave propagation. Scalar features from the propagated waveform, the point of origin, and propagation effects are input for the random forest that estimates the yield of the source between 1 and 1500 tonnes of equivalent TNT. When applied to five surface shots from the Humming Roadrunner (HRR) experiment, our method performs reasonably well and compares favorably with other methods despite not being trained on real data. Additionally, we explore how well the model generalizes to a different season and a different location using infrasound records

from the Bouncing Bilby and Sayarim II explosion tests. By leveraging broadband propagation techniques with machine learning, we seek to enhance the effectiveness and applicability of explosion monitoring and build on the growing trend of deep learning applications within the seismoacoustic community.

10:40

4aPAa8. Fundamental frequency normalization for target identification and classification. Lorin Hendricks (GSL, ERDC – USACE, 3909 Halls Ferry Rd., Vicksburg, MS 39180, Lorin.Hendricks@usace.army.mil)

During operation, most types of vehicles produce fundamental frequencies in the infrasound (below 20 Hz) and low-end audible ranges (20–400 Hz). These fundamental frequencies are then repeated as harmonics at integer multiples of the fundamental frequency. These frequencies are generally tied to the revolutions per minute (RPM) of the engine and are mostly produced by the ignition of the cylinders in an internal combustion engine and the air impact of the blades from propellers or rotors. However, the constant variation of the environment, the engine's RPM, and velocity changes (and resulting Doppler shift) during normal operation result in difficulties in identifying and classifying based on acoustical signals. Fortunately, the ratio of the first few harmonics remains relatively stable. Additionally, atmospheric absorption is roughly proportional to frequency, allowing these first few harmonics to propagate much farther. This paper will demonstrates techniques for identification of tonal sources by utilizing an ambient estimation, refinement of the fundamental frequency estimation by combining harmonics, and normalization of the spectrum for future identification and classification. [Permission to publish was granted by Director, Geotechnical & Structures Laboratory, U.S. Army Engineer Research and Development Center.]

11:00

4aPAa9. Investigation of the acoustic signals generated by the OSIRIS-REx re-entry. Elizabeth Silber (Sandia National Labs., 1515 Eubank Blvd. NE, Albuquerque, NM 87123, esilber@uwo.ca) and Vedant Sawal (Sandia National Labs., Albuquerque, NM)

The OSIRIS-REx sample return capsule re-entry provided a unique, controlled opportunity to study atmospheric shock wave propagation from a high-altitude source. Unlike natural meteoroids, which often undergo complex fragmentation and ablation, the capsule offered a stable source for characterizing specific acoustic generation mechanisms. We utilize infrasound data recorded by a regional network of ground-based sensors to analyze the acoustic signature associated with the descent. This study employs a semi-analytical weak-shock approach developed for a cylindrical line source to evaluate signal evolution as the wavefront propagates through the atmosphere. We examine the applicability of established shock theories to the recorded data, comparing theoretical predictions with the observed waveforms. The analysis explores the relationship between source characteristics and observations, providing a framework for a better understanding the physics of non-fragmenting and non-ablating entries. These findings have broader implications for the monitoring and characterization of space debris, artificial re-entries, and meteoroids using infrasound stations. [Work managed by SNL and operated by NTESS under DOE NNSA contract DE-NA0003525. Cleared for release.]

11:20

4aPAa10. From sound waves to security: Analyzing global infrasound for treaty verification. Kyler Goddard (Air Force, 10989 South Patrick Dr., Patrick AFB, FL 32925, kyler.goddard.2@us.af.mil), Thomas VanDeMark, and Jeffrey Miller (Air Force, Patrick AFB, FL)

The Air Force Technical Applications Center (AFTAC) is the sole organization within the United States government tasked with detecting and identifying foreign nuclear explosions to monitor compliance with international treaties. A primary tool for this mission is the International Monitoring System (IMS), a global network of sensors that includes an extensive array of infrasound sensors. This presentation will detail AFTAC's methods for analyzing events of interest captured by the IMS infrasound network. Specifically, we will discuss the application of open-source algorithms and geophysical data analysis techniques to process and interpret atmospheric acoustic data. The use of these open-source tools facilitates reproducible research and enhances our capabilities to distinguish between naturally occurring and man-made explosive events.

11:40

4aPAa11. Close-range signal characteristics of a gas-combustion infrasound source. Chad M. Smith (The Penn State Univ., P.O. Box 30, State College, PA 16804, chad.smith@psu.edu) and Thomas B. Gabrielson (Penn State Univ., State College, PA)

A precisely controlled and well-characterized source is an invaluable tool in characterization of any receiver, propagation path, or detection system. Design of a transportable sound source within the infrasound band is a difficult engineering challenge due to the incredibility large air volume (and mass) that must be moved to create useful signal levels. Prior work developed a novel source method taking advantage of the large energy density available in gas combustion to displace the atmosphere with the gaseous combustion constituents of propane and atmospheric oxygen. This method has been demonstrated to be a safe, scalable, and well-controlled technique, and engineering models have been developed for the acoustic output. However, to date, signal characteristics such as spatial coherence and repeatability lack thorough evaluation. A recent collaboration between the French Alternative Energies and Atomic Energy Commission and The Pennsylvania State University allowed measurements of this source using a densely populated array of microbarometers at close range (300 m), but within the acoustic farfield ($kr >> 1$, where k is the acoustic wavenumber and, r is the range) of this highly complex fluid-displacement source. Close-range signal characteristics and their comparison with those of an ideal point source between 0.5 and 10.0 Hz will be discussed.

Session 4aPAb

Physical Acoustics, Structural Acoustics and Vibration and Biomedical Acoustics: Ultrasonic Characterization Across Scales and Applications I

Andrea P. Arguelles, Cochair
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Olivia J. Cook, Cochair
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Haley N. Jones, Cochair
Mater. Sci. and Eng., Penn State Univ., N-225 Millenium Science Complex, State College, PA 16802

Invited Paper

8:05

4aPAb1. Laser-ultrasonic characterization of anisotropic materials utilizing the ultra-transient grating phenomenon. Hanus Seiner (Inst. of Thermomechanics, Czech Acad. of Sci., Dolejskova 5, Prague 18200, Czechia, hseiner@it.cas.cz), Tomas Grabec, Jakub Kusnir, Pavla Stoklasova, David Mares, and Petr Sedlak (Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia)

Transient grating spectroscopy (TGS) is a materials characterization method based on the use of coherent laser optics to create a spatially harmonic array of pulsed thermoacoustic sources on the free surface of the examined material. The array generates counter-propagating surface acoustic waves (SAWs) that superpose into a standing-wave pattern known as the transient grating, and the frequency of the standing waves, determined by the wave speeds of the SAWs, is optically detected. When using highly sensitive photodiodes and a high degree of signal averaging, finer features of the recorded TGS spectrum become visible, revealing that the surface response to the spatially harmonic loading is more complex than just the dominant peaks associated with SAWs. In the early-field regime, immediately after the pump-laser pulse, bulk acoustic waves also create standing-wave patterns that, however, diminish much faster than those from SAWs and can therefore be termed ultra-transient. The resulting experimental response is strikingly similar to the Green's function of the examined surface, with edges, shoulders, and acoustic sinks marking the wave speeds of bulk acoustic waves. Owing to this ultra-transient response, TGS experiments on a single free surface allow the determination of all independent elastic coefficients of an anisotropic solid.

Contributed Paper

8:25

4aPAb2. Ultra-transient grating spectroscopy and extraordinary elastic anisotropy. Tomas Grabec (Inst. of Thermomechanics, Czech Acad. of Sci., Dolejskova 1402/5, Praha 18200, Czechia, tomas.grabec@it.cas.cz), Pavla Stoklasova, Kristyna Zoubkova, Petr Sedlak, and Hanus Seiner (Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia)

Elastic anisotropy introduces complexities to ultrasonic-wave propagation: wave modes couple, phase and group velocities diverge, and energy focuses into preferred directions. While conventional techniques already struggle at the anisotropy level of copper (Zener anisotropy ratio $A \approx 3$), phase-transforming materials go far beyond these values, commonly reaching $A > 10$, demanding specialized diagnostics. We employ ultra-transient grating spectroscopy (UTGS), a dedicated modification of transient grating spectroscopy, to map the angular dispersion surface-guided acoustic modes, without energy-steering bias. Two interfering laser beams create a tunable pattern that excites a transient grating formed by counterpropagating surface acoustic waves (SAWs). With optimized detection, UTGS resolves subtle spectral features formed by various wave modes, residing in the earliest portion of the time trace. These ultra-transient signatures agree with frequency-domain elastodynamic surface Green's-function predictions, enabling detailed interpretation. Applying the UTGS setup to the modulated-martensite Ni–Mn–Ga crystal reveals highly unusual ultrasonic-wave behavior, offering a clear fingerprint of lattice instability and extreme elastic anisotropy—as a single highly compliant crystallographic direction drives the anisotropy ratio to $A \approx 250$. The results illustrate how UTGS enables new insights into surface-wave physics in the limit of extraordinary anisotropy.

Invited Paper

8:40

4aPAb3. Probing solid-state battery degradation with through-transmission ultrasonic testing. Simon Montoya-Bedoya (Walker Mech. Eng. Dept., The Univ. of Texas at Austin, Austin, TX), Avtar Singh, Kaustubh G. Naik, Donal P. Finegan (National Lab. of the Rockies, Golden, CO), Hadi Khani, Ofodike A. Ezekoye, and Michael R. Haberman (Walker Mech. Eng. Dept., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX, haberman@utexas.edu)

Silicon solid-state batteries (Si-SSBs) promise a significant increase in energy density and safety compared to conventional lithium-ion batteries. Nevertheless, the Si-SSB technologies face practical challenges like sustained stack pressure to maintain contact between layers and high volumetric expansion which yields poor cyclability. Ultrasonic testing (UT) is uniquely suited to track interlayer coupling and microstructural evolution during cycling. We present the challenges associated with measuring the properties of acoustically thin samples which require a custom experimental apparatus that ensures repeatable transducer location and pressure uniformity during testing while minimizing measurement error. In addition, we analyze the measurement results from 40 degradation cycles performed on the Si-SSBs, during which we monitor cell behavior with contact UT, and we perform periodic immersion UT to evaluate spatial heterogeneities. An overall softening of the Si-SSBs is observed during aging, indicated by an increase in the time-of-flight together with an overall reduction in the energy transmitted through the cell. The latter behavior is hypothesized to be due to damage accumulation from gas/void formation in electrodes and/or debonding between layers. A complementary modeling analysis of ultrasonic transmission through these Si-SSB multilayered systems and the experimental apparatus is presented, providing insights into internal mechanical changes occurring due to aging.

Contributed Paper

9:00

4aPAb4. Non-contact ultrasound metrology to enable advanced battery electrode manufacturing. Wesley Chang (Mech. Eng. and Mech., Drexel Univ., 3141 Chestnut St., Philadelphia, PA 19104, weschang@drexel.edu), Andre Tayamen, and Sam Amsterdam (Mech. Eng. and Mech., Drexel Univ., Philadelphia, PA)

Advanced metrology techniques enable faster and more accurate detection of defects in battery electrodes to improve device performance, lower production cost, and enhance operational safety. We present a non-contact ultrasonic metrology method that utilizes air-coupled transducers to map important physical properties of battery electrodes. The technique generates acoustic impedance images by scanning across electrode surfaces, enabling the calculation of structural information from acoustic wave physics. We demonstrate the technique's capability to detect buried defects, resolve micron-scale thickness variations, and quantify density uniformity resulting from calendaring. A simple analytical model illustrates how these physical properties are linked to the measured acoustic signals. We conclude by discussing the method's potential for both R&D and in-line manufacturing quality control.

Invited Paper

9:15

4aPAb5. Quantitative ultrasound imaging of soft tissues from 1 to 1000 MHz. Jonathan Mamou (Radiology, Weill Cornell Medicine, 416 East 55th St., B1, New York, NY 10025, jom4032@med.cornell.edu)

Quantitative ultrasound (QUS) is an established research field using physical principles and inverse-model approaches to measure a wide range of soft-tissue properties, i.e., QUS parameters. From this definition, QUS parameters are powerful for ultrasound-based tissue characterization because they are user- and system-independent. This presentation focuses on QUS parameters associated with fundamental acoustical (e.g., attenuation, impedance, speed-of-sound) and microstructural properties (e.g., backscatter coefficient, envelope statistics). In biomedical ultrasound applications, QUS parameters are expected to change with disease progression and therefore are powerful tools to non-invasively improve diagnosis, treatment guidance and monitoring, and overall patient management in a wide range of organ systems, diseases, and spatial resolutions. After a brief overview, this presentation will review several QUS applications in biomedical applications ranging from 1 to 1000 MHz: myocardium, lymph nodes, tendons, placenta, nail bed, and eyes. [Work supported by NIH grants R01CA277038, R01GM143388, R21EB028084, and R01HL175513.]

9:35

4aPAb6. Sum- and difference-frequency generation by shear wave beams in tissue-like media. Philip G. Kaufinger (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78758, pkaufinger@utexas.edu), John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Plane nonlinear shear waves in isotropic media are subject only to cubic nonlinearity at leading order and therefore generate only odd harmonics during propagation. Wavefront curvature in shear wave beams breaks the symmetry in the material response and yields quadratic nonlinearity, such that a second harmonic may be generated at second order by a shear wave beam depending on the polarization of the wave field. Recent analyses of quadratic nonlinearity in shear wave beams [Spratt *et al.*, AIP Conf. Proc. **1685**, 080007 (2015); Kaufinger *et al.*, Wave Motion **139**, 103595 (2025)] have been concerned with second-harmonic generation by a single-frequency primary beam. The inclusion of a second collinear source-frequency beam leads to the generation of sum- and difference-frequency wave fields via quadratic nonlinear interactions between the two source-frequency beams, which corresponds to a shear wave parametric array. Here, closed-form analytical solutions and numerical simulations are presented for a bifrequency collinear source with radial polarization and Gaussian amplitude shading. Examples of the sum- and difference-frequency wave fields are presented using parameters consistent with tissue-like media for applications to ultrasound shear wave elastography. [P.G.K. is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

9:50–10:10 Break

10:10

4aPAb7. Numerical simulation of harmonic generation in nonlinear shear wave beams using a frequency-domain algorithm. Cassidy A. Christie (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78758, cchristie@utexas.edu), John M. Cormack (Div. of Cardiology, Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

A time-domain numerical algorithm was presented at the Spring 2011 ASA meeting for simulation of radially polarized shear wave beams in a viscous medium with both quadratic and cubic nonlinearity [Spratt *et al.*, JASA **129**, 2679 (2011)]. The present work focuses on harmonic generation in soft elastic media with arbitrary absorption and dispersion, and therefore a frequency-domain numerical algorithm is desired. Such an algorithm was recently developed to model second-harmonic generation due to weak quadratic nonlinearity in soft tissue-like media, motivated originally to confirm analytical perturbation solutions for this field and to include source conditions for which analytical solutions are not available [Kaufinger *et al.*, JASA **157** (2025); doi: 10.1121/10.0037533]. Here, a frequency-domain algorithm is presented with both quadratic and cubic nonlinearity evaluated via convolution of the harmonics. For simplicity only radially polarized (axisymmetric) shear wave beams are considered, which is common in shear wave elastography. Simulations are presented for cases with purely quadratic, purely cubic, and combined nonlinearities to examine how nonlinear and diffraction phenomena affect harmonic generation and waveform distortion in shear wave beams. Simulations will also be presented for parameters relevant to soft tissue. [C.A.C. is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

Invited Paper

10:25

4aPAb8. Measurement of crystallographic texture within engine alloy components. Michael J. Lowe (Dept. of Mech. Eng., Imperial College London, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk) and Bo Lan (Mech. Eng., Imperial College London, London, United Kingdom)

The manufacturing processes for metal components for high-value safety-critical applications often create significant preferred orientations of their polycrystals, resulting in components with orientation-dependent properties, including stiffness, thermal expansion, strength, and fatigue and creep resistance. Knowledge of the crystallographic texture is important for assured performance and quality control. Texture can be measured on exposed surfaces using, for example, Electron Back-Scatter Diffraction (EBSD), or in thin samples using neutron diffraction, but until recently there has been no capability in industry to measure texture within significant volumes of real components. Work by the authors over the past 10 years has pursued a method of measuring texture within large volumes of material from the speed of ultrasound propagated through the component in multiple selected directions. This has been demonstrated to work reliably on samples of a specific regular shape in some alloys of importance to industry, which offers value at early stages of the manufacturing process. The talk will present some of the ongoing research to take that principle towards application, including some specific characterization targets for industry and the pursuit of the capability to measure texture in components of arbitrary shape, which would allow them to be characterized much later in the manufacturing process.

Contributed Papers

10:45

4aPAb9. Interpreting scattering-induced ultrasonic attenuation during phase transformations in NiTi shape memory alloys. Olivia J. Cook (Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci., University Park, PA 16801, ojc3@psu.edu) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Nickel titanium (NiTi) shape memory alloys derive unique deformation properties from reversible phase transformations initiated by temperature changes. Specifically, during cooling through the transformation range, martensitic plates form within grains of the high-temperature austenite phase. Previously, ultrasonic testing has been used to monitor such transformations through changes in wave velocity and attenuation, though only velocity changes are well-understood through their connection to evolving elastic constants. Reported attenuation peaks during transformation lack a mechanistic explanation. In this investigation, longitudinal wave speed and attenuation are monitored during thermal phase transformations in near-equiatomic, vacuum arc-melted NiTi using immersion ultrasonic testing at 15 MHz. The results show that alongside the well-known velocity signature

of the martensitic transformation, attenuation decreases, in contrast to most prior reports. Analytical attenuation estimates depending on published temperature-dependent single crystal elastic constants indicate this behavior is consistent with a reduction in elastic anisotropy preceding martensite formation, weakening scattering strength. This study highlights elastic wave scattering as an important contributor to ultrasonic attenuation during phase transformations. Further, the results suggest that ultrasonic testing can provide a window into a transformation phenomenon usually only accessible via specialized microscopy conditions or using synchrotron facilities.

11:00

4aPAb10. Temperature-dependent elasticity in platinum compared with iridium. Shaofei Wang (Material Sci. and Technol. Div., Oak Ridge National Lab., 1 Bathel Valley Rd., P.O. Box 2008, MS-6064, Oak Ridge, TN 37831, wangs1@ornl.gov), James R. Torres (Neutron Scattering Div., Oak Ridge National Lab., Oak Ridge, TN, USA), Andrew F. May, and Raphael Hermann (Material Sci. and Technol. Div., Oak Ridge National Lab., Oak Ridge, TN, USA)

Platinum (Pt) and iridium (Ir) are face-centered-cubic noble metals with strong electron–phonon coupling. Although the complete elastic tensors of Pt have been reported above room temperature, below room temperature only c_{44} and have been available. Here, we present temperature-dependent resonant ultrasound spectroscopy (RUS) measurements on Pt from 5 to 300 K, that resolve the full set of cubic elastic constants. Compared with literature data for Ir, Pt shows slower and non-monotonous softening in c_{44} but faster softening in the shear modulus c'. This mode-selective behavior indicates stronger and more broadly distributed anharmonicity and electron–phonon coupling associated with [110] transverse-acoustic phonons in Pt, whereas anomalies in Ir are more localized in reciprocal space. The resulting evolution of elastic anisotropy (e.g., increasing Zener anisotropy in Pt) highlights Pt as a promising platform for probing the interplay among lattice elasticity, phonon anomalies, and electron–phonon coupling. [This work is supported by the US Department of Energy (DOE), Office of Science, Basic Energy Sciences, Materials Sciences and Engineering Division. Research conducted at Oak Ridge National Laboratory's (ORNL) neutron sources was sponsored by the Scientific User Facilities Division, Office of Basic Energy Sciences, US DOE.]

11:15

4aPAb11. An alternative method of resonance sweep testing using optical microphones. Olivia C. Van Fleet (Acoust., Penn State, 556 White Course Dr., State College, PA 16801, ocv5015@psu.edu), Lovejoy Mutswatiwa (Eng., Penn State Univ., University Park, PA), Andrea P. Arguelles, and Christopher M. Kube (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Ultrasonic fatigue testing is typically performed on dog-bone specimens with circular crosssections and resonances in the 20–100 kHz range. A strong longitudinal vibrational mode is usually selected. In this work, the approach is extended to much higher frequencies (approximately 500 kHz–1.5 MHz) using smaller cylindrical tensile specimens. To develop the method, the longitudinal resonance of each specimen must be measured in the same configuration used for fatigue loading, with the flat specimen face in contact with a 1 MHz longitudinal transducer. This constraint limits the applicability of conventional resonance ultrasound spectroscopy setups and places greater demand on the sensing technique used during frequency sweeps. A laser interferometer was first used to detect surface motion during excitation. Although it identified the resonant frequencies, the instrument used was not ideal below 1 MHz, and the resulting signals did not provide reliable absolute displacement amplitudes. An optical microphone was therefore evaluated as a supplemental method for detecting acoustic pressure fluctuations near the specimen surface. This pressure-based signal produced a cleaner resonance response, though it likewise does not yield absolute displacement. The comparison highlights practical considerations for resonance identification in constrained ultrasonic test configurations and informs the development of high-frequency fatigue methods.

11:30

4aPAb12. Real-time sensing of microstructural evolution in polycrystalline materials for laser-based additive manufacturing. Nathan Kizer (Penn State Univ., 212 Earth and Eng. Sci. Bldg., University Park, PA 16802, njk19@psu.edu), Jordan Lum, Rosa E. Morales, Gabe Guss (Lawrence Livermore National Lab., Livermore, CA), David M. Stobbe (NDE, Lawrence Livermore National Lab., Livermore, CA), and Christopher M. Kube (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Laser-based metal additive processes are susceptible to microstructural defects such as deleterious phases or unpredictable grain evolution. *In situ* X-ray diffraction provides valuable real-time information on phase formation during solidification and subsequent solid-to-solid phase transformations; however, it is limited to specialized facilities, narrow material selection, and lower sampling rates that may miss rapid transient events. *In situ* ultrasonic monitoring offers an alternative with superior temporal resolution and material flexibility through grain scattering analysis. This technique utilizes high-frequency shear waves (25 MHz) with wavelengths comparable to grain size to characterize microstructural evolution via forward-scattered signals during continuous monitoring from melt to ambient temperature. Multiple Fe-Cr-Ni samples with varying Cr-Ni ratios and laser parameters were evaluated to assess their impact on grain scattering response and microstructural pathways. The results demonstrate sensitivity to phase transformations through distinct grain scattering signatures, with future work focusing on refining this methodology and extending it to other material systems as a broadly applicable additive manufacturing characterization tool. [This work was performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344 and was supported by the LLNL-LDRD Program under Project No. 23-ERD-009. The document release number is LLNL-ABS-2002220.]

11:45

4aPAb13. Microstructural scattering bounds for ultrasonic evaluation of internal features in LPBF SS316L. Harshith Kumar Adepu (Mech. Eng., Purdue Univ., 500 Central Dr., B36, West Lafayette, IN 47906, adepu@purdue.edu), Partha Pratim Pandit, and Luz D. Sotelo (Mech. Eng., Purdue Univ., West Lafayette, IN)

Laser powder bed fusion (LPBF) is a metal additive manufacturing technique (AM) used to manufacture complex geometries with high resolution. However, LPBF process parameters lack standardization and lead to microstructural heterogeneity, causing high ultrasonic scattering amplitudes, which limit the detectability of internal features. The objective of this work is to determine the extent to which porosity-driven ultrasonic scattering can be reliably distinguished from microstructure-driven scattering in LPBF SS316L and to understand how background scattering interferes with locating internal features. For this purpose, cubic samples were fabricated using different process parameter combinations to produce distinct microstructures while maintaining comparable nominal energy input. Ultrasound measurements were used to quantify the experimental scattering baseline and define threshold-based bounds, which were then used as a denoising framework to suppress microstructure-driven contributions. Porosity and internal geometry, as well as grain morphology, were quantified via X-ray computed tomography and electron backscatter diffraction, respectively. The results show how detectability limits are affected by process-induced micro and mesostructured and the improvements resulting from thresholding methods. This work highlights the need for microstructure-aware ultrasonic NDE protocols for AM parts and provides a practical pathway to improve internal feature characterization under realistic microstructural variability.

Session 4aPP

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

Z. Ellen Peng, Chair
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All posters will be on display from 8:00 a.m. to 12:00 noon. Authors of odd-numbers papers will be at their posters from 8:00 a.m. to 10:00 a.m.and authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aPP1. Usability of VR exergaming with 3-D audiovisual feedback in rehabilitation engineering. György Wersényi (Széchenyi István Univ., Egyetem ter 1., Gyor 9026, Hungary, wersenyi@sze.hu)

Exergaming combines physical exercise with interactive virtual environments, where motion tracking, spatial acoustic, and visual cues enhance sensory engagement and motor learning. A randomized clinical trial involved 100 patients assigned to four intervention groups: exergame, cycling, agility, and robot therapy. The exergame system utilized the Microsoft Xbox 360 with Kinect, employing *Reflex Ridge*, *Space Pop, and Just Dance* games incorporating synchronized 3-D audiovisual feedback. Spatial audio cues were integrated to simulate dynamic sound localization and directional feedback and to enhance timing, balance control, and sensorimotor coordination. Heart rate monitoring confirmed the safety and adaptability of high-intensity exergaming. Significant improvements were observed in gait velocity, Functional Independence Measure scores, and Berg Balance Scale outcomes. Exergame and cycling therapies yielded the most comprehensive benefits, demonstrating that multimodal 3-D feedback can contribute to neuroplasticity, postural control, and cognitive–motor integration. Results suggest that gamification using 3-D VR solutions with spatial audio rendering can increase engagement and rehabilitation efficiency.

4aPP2. Factors affecting self-tuning benefits: Personalized scenes and prior hearing-aid experience. Bertan Kursun (Speech and Hearing Sci., Univ. of Washington, 332 24th Ave., E, Seattle, WA 98112, bertanks@uw.edu), Brianna Chessik, Hailey Robinson, Iris Choe, Isabella E. Cunio, and Yi Shen (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Self-tuning enables hearing aid users to personalize fittings, but its outcomes differ across individuals. This study investigated whether prior hearing aid experience moderates self-tuning outcomes in older adults with mild-to-moderate hearing loss and whether outcomes differ when self-tuning is conducted in personalized versus generic scenes. In Experiment 1, 40 participants stratified by prior hearing-aid experience were compared on speech-in-noise outcomes and real-ear insertion gains (REIG) between self-tuned and prescription-based fittings. In Experiment 2, 16 participants self-tuned hearing aids in either personalized or generic scenes after two weeks of real-world use. Personalized scenes were acoustically reconstructed from daily listening environments collected via ecological momentary assessment. Generic scenes included 60 dBA speech in either babble (+5 dB SNR) or traffic (+10 dB SNR) noise. Two groups were compared on sound quality ratings and speech-in-noise recognition. In Experiment 1, practiced hearing aid users showed greater reductions in perceived mental demand than new users, and no reduction in average REIG was observed for new users. In Experiment 2, self-tuned programs received higher sound quality ratings than prescription-based programs, with no additional benefit of personalized scenes. Whereas prior hearing aid experience was associated with greater self-tuning benefit, personalized scenes did not enhance self-tuning outcomes under the current protocol.

4aPP3. Robust denoising and parameter estimation of electrically evoked compound action potentials under low signal-to-noise ratio conditions. Fan-Jie Kung (Elec. Eng., National Taipei Univ. of Technol., No. 1, Sec. 3, Zhongxiao E. Rd., Integrated Technol. Complex, Rm. 215, Taipei 10608, Taiwan) and Kai-Hsiang Hsu (Elec. Eng., National Taipei Univ. of Technol., Taipei, Taiwan)

Electrically evoked compound action potentials (ECAPs) are used to assess auditory nerve function in cochlear implant users, providing an objective, noninvasive means of assessing auditory nerve responses without requiring active participation from patients. However, ECAP recordings are often corrupted by noise, particularly under low signal-to-noise ratio (SNR) conditions, leading to unreliable estimation of neural activity patterns and current spread. In addition, two important challenges remain insufficiently addressed: robust noise reduction of ECAP recordings and reliable estimation of neural response amplitudes. Many existing studies assume noise-free ECAP signals or predefined amplitudes, which limits their applicability in practical measurement conditions. Accurate modeling and parameter estimation of auditory nerve fiber responses based on ECAP measurements are essential for characterizing neural health and current spread along the cochlea. This work investigates integrated spatial, adaptive, and learning-based signal processing approaches for ECAP denoising and auditory nerve parameter estimation. By combining spatial image-based filtering, adaptive filtering techniques, and data-driven models, the proposed framework aims to enhance ECAP signal quality and enable more accurate estimation of neural response amplitude, neural health, and variance under adverse recording conditions.

4aPP4. Adding interaural level differences improves fusion and lateralization precision, but degrades lateralization accuracy. Karla G. Rodriguez, Stephanie Romero, Prajna BK (Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, jaronoff@illinois.edu)

Previous research suggests that adding an interaural level difference (ILD) under conditions of decreased interaural coherence improves binaural fusion. These findings raise the question of whether improvements in binaural fusion reflect increased localization precision. This study aims to address that question. Seven typical hearing listeners were tested using sounds processed in a simulated reverberant environment, which reduces the interaural coherence of the signal, with an RT60 of either 0.005 or 0.1 seconds, and with either no ILD or a fixed ILD (12 dB) added after the simulation of the reverberant environment. Participants were instructed to indicate the perceived location, size, and number of the auditory image(s) using a dial. Location discrimination was also assessed, with listeners detecting a change in source location. Binaural fusion improved when an ILD was added, consistent with previous research. This corresponded to a lateralized shift in the perceived sound source location. Localization variability decreased in most conditions when an ILD was added to the stimuli, suggesting improved

localization precision. Location discrimination accuracy was degraded by the addition of an ILD. The results suggest that adding an ILD both increases localization precision and decreases localization accuracy. [Work funded by NIH/NIDCD Grant R01-DC018529.]

4aPP5. Abstract withdrawn.

4aPP6. Abstract withdrawn.

4aPP7. Preserving natural reverberation decay improves speech intelligibility in cochlear implant users. Prajna BK (Speech & Hearing Sci., UIUC, 901 S 6th St., Champaign, IL 61820, prajnab2@illinois.edu) and Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

In reflective environments, reverberation alters the spectrotemporal characteristics of speech. While typical-hearing (TH) listeners maintain robust intelligibility in low- to moderate-reverberant conditions, cochlear implant (CI) users often experience substantial degradation. TH listeners adapt by perceptually down-weighting the exponentially decaying portions of the temporal envelope associated with reflections. To determine whether this temporal decay cue is preserved following CI processing, room impulse responses were analyzed. Results showed that the processor's logarithmic compression stage artificially increases and sustains reverberant energy, distorting the natural exponential decay and potentially limiting CI users' perceptual benefit from down-weighting decaying reflections. Behavioral testing examined whether reducing the aggressiveness of logarithmic compression—thereby better preserving the natural decay of reverberation—improves speech intelligibility in simulated reverberant environments. Preliminary results from two Cochlear-brand CI users indicate that softer compression improves intelligibility in reverberation. Further benefits were observed when this manipulation was applied selectively to frequency channels dominated by reverberant energy. These findings suggest that preserving the natural exponential decay of reverberation is a critical factor for improving speech intelligibility for CI users in reverberant environments. [Work funded by NIH/NIDCD Grant R01-DC018529.]

4aPP8. Bilingualism and cognitive resilience: Online episodic memory for auditory verbal material. Maureen Samokomgboeso (Kingsborough Community College, 2001 Oriental Blvd., E329, Brooklyn, NY 11235, MAUREEN.SAMOKOMGBOESO47@students.kbcc.cuny.edu), Aspen Shearer (Brooklyn College, Brooklyn, NY), Courtney Stathis, Jennifer Edwards-Saul (Kingsborough Community College, Brooklyn, NY), and Laura M. Muscalu (Psych., Ithaca College, Ithaca, NY)

Cognitive reserve refers to the ability to maintain cognitive functioning despite aging or neuropathology. Episodic memory is thought to better represent the neural basis of cognitive reserve than other cognitive abilities or static proxies such as education. The present study examines whether bilingual language experience is associated with enhanced episodic memory for auditorily presented verbal material, using an online paradigm modeled on established clinical memory measures. Participants complete an episodic memory task via Pavlovia involving auditory encoding of word lists, followed by immediate and delayed recall. This task demands auditory perception, phonological encoding, and memory consolidation, closely paralleling verbal episodic memory measures commonly used in cognitive reserve research. Language experience is assessed using background questionnaires capturing age of acquisition, daily language use, and proficiency tests. The experiment is underway, with a target sample of about 60 adult participants. We predict that bilingual participants will demonstrate higher recall accuracy and reduced forgetting over the delay compared to monolinguals and that greater balance and intensity of bilingual language use will be associated with stronger episodic memory performance. By adopting an episodic memory structure shown to predict cognitive reserve and implementing it with auditory verbal stimuli, this student-led project contributes to acoustically grounded approaches to bilingual cognition using scalable online methods.

4aPP9. Loudness compression in electronic hearing protection can degrade speech intelligibility in noise. William E. Audette (Benchtop Eng. LLC, 281 Sweet Pond Rd., Guilford, VT 05301, chipaudette@benchtopeng.com), Nicole Lyons, Sarah E. Zwick, Eric M. Desjardins, and Jed C. Wilbur (Creare LLC, Hanover, NH)

Hearing Protection Devices (HPDs) are used to reduce ambient noise to safe levels. Unfortunately, they also attenuate speech. To mitigate this loss, many HPD manufacturers have added microphones and speakers to their HPDs to allow you to hear speech and other sounds through the hearing protection (hear thru). To prevent loud sounds from being presented too loudly, the hear-thru system will also include some form of loudness compression. In this paper, we present how loudness compression processing in electronic HPDs can change the intelligibility of speech when listening in noise. Specifically, we used an objective speech intelligibility analysis using the objective tool "Articulation Band Correlation—Modified Rhyme Test" (ABC-MRT) to estimate how speech intelligibility (SI) is affected by different loudness compression algorithms and different compression speeds. Our results show that SI is degraded whenever the ambient noise level is above the SPL where the loudness compression begins (knee point). Crucially, we find that the slower compression (i.e., longer "attack" and "release") results in much less SI degradation than fast compression. We confirmed key elements of these findings using a small human subject study. We hope that these findings are helpful to industrial hygienists when choosing electronic hearing protection and to engineers when designing new hear-thru systems.

4aPP10. Computational modeling of spectral contrast effects in speech perception. Braden N. Maxwell (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, maxwe318@umn.edu), Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Christian E. Stilp (Speech Pathol. and Audiol., Marquette Univ., Milwaukee, WI)

Auditory sensitivity for spectral peaks is magnified by precursor sounds that have spectrally contrastive peaks. This sensitivity affects the interpretation of speech sounds. For example, a precursor sentence with prominent energy in low-F3 frequencies (1700–2700 Hz) encourages perception of the high-F3 target "da"; precursor context with prominent energy in high-F3 frequencies (2700–3700 Hz) encourages perception of the low-F3 target "ga". Adaptation of auditory nerve fibers was hypothesized to underlie these perceptual shifts. Here, neural responses to stimuli previously used in behavioral experiments were simulated using an auditory nerve model (Zilany *et al*., 2014). Center frequencies of high-spontaneous-rate fibers covered tonotopic regions of interest where precursors should affect contrastive components of the target sounds. Rates were averaged over a 50-ms window at target syllable onset and then normalized by the mean of the population response. Then, perception was simulated by categorizing the population response as "da" or "ga" on each trial. Neural responses reflected human listeners' tendency to magnify spectral contrast: simulations estimated more high-F3 "da" responses following precursor sentences with amplified low-F3 frequencies, and more low-F3 "ga" responses following high-F3-amplified sentences. Results support the hypothesis that neural adaptation in the auditory periphery can explain SCEs in speech perception. [Work supported by NIDCD.]

4aPP11. Complex pitch perception as a function of age and hearing loss: Comparing time and place predictions. Braden N. Maxwell (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, maxwe318@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Fundamental-frequency discrimination of harmonic complex tones is often an order of magnitude more accurate when the stimulus includes low-numbered, spectrally resolved harmonics than when only high-numbered, unresolved harmonics are present. It remains unclear whether peripheral spectral resolution (place) and/or peripheral temporal encoding fidelity (time) underlie this phenomenon. Recent machine-learning approaches have suggested that temporal is more important than spectral encoding, but that the spectral profile of the training stimulus set, relative to the training noise, may be more critical than auditory processing limitations. This study provides an empirical test of this prediction, as listeners with congenitally normal hearing initially had similar access to the spectral profiles, but may now have limited access to temporal or spectral cues, due to age and/or hearing loss. Fundamental-frequency discrimination was measured as a function of the lowest harmonic number for a fixed high-pass cut-off frequency of 1200 Hz. Overall performance was assessed, along with the transition between good and poor performance in listeners spanning a wide range of age and hearing loss. Independent measures of spectral and temporal resolution were obtained in the same listeners. The results will provide an important large-scale empirical test of competing theories of complex pitch. [Work supported by NIH Grant R01DC005216.]

4aPP12. Notches in level functions of distortion-product otoacoustic emissions and intracochlear distortion products in mice. Lauren Ysais (USC, Los Angeles, CA) and James Dewey (USC, Zilkha Neurogenetic Inst., Rm. 407, 1501 San Pablo St., Los Angeles, CA 90033, james.dewey@med.usc.edu)

Distortion-product (DP) otoacoustic emissions (DPOAEs) provide a noninvasive window onto nonlinearities in outer hair cell (OHC) function and cochlear mechanics. However, it remains unclear whether certain features in the frequency and level dependence of DPOAEs arise from the behavior of DPs generated by individual OHCs, or if they are instead attributed to interference between spatially distributed DP sources. Here, in mice, we specifically examined whether notches in DPOAE level functions are also present in DPs measured in the vibrations of the OHCs. We found that DPOAEs often exhibited notches at stimulus levels above 60 dB SPL, with the position and depth of the notches depending on both stimulus frequency ratio and stimulus level separation. In contrast, DPs measured from the OHC region typically grew monotonically before saturating at high levels, regardless of stimulus parameters. While notches could be observed in DPs measured from the nearby basilar and tectorial membranes, these usually did not occur at the same stimulus levels as the notches in DPOAEs. The data therefore indicate that notches in DPOAEs and intracochlear DPs are not produced by the underlying DP-generating nonlinearity but instead result from complex interference phenomena. [Work supported by NIH/NIDCD grant R01DC021006.]

4aPP13. Urban noise level measurements and surveyed perceptual responses in Rabat, Morocco Medina. Andrew T. Pickering (Dept. of Mech. and Industrial Eng., Univ. of Minnesota Duluth, 1730 E 5th St., Duluth, MN 55812, andrewpickering2005@gmail.com) and David A. Anderson (Elec. Eng., Univ. of Minnesota Duluth, Duluth, MN)

Noise pollution is an increasingly prevalent environmental issue in urban areas, yet its impact on residents remains understudied in countries like Morocco due to no permissible noise limits and no regulatory definition of noise. This study investigates the extent and perception of noise pollution in Rabat, Morocco, using a combination of objective sound level measurements in the Rabat Medina and subjective public perception surveys. Using phone-based decibel readers, noise levels were measured across different points in the Medina in Rabat. Simultaneously, residents were surveyed to assess their perception and effects of noise. While the average sound level values never exceed 80 dB, peak sound level measurements reached up to 88 dB, which is above the 85 dB threshold for hearing damage according to the Centers for Disease Control and Prevention (CDC). The survey shows that while 55.2% of people reported difficulty concentrating, 42.1% also reported that urban noise created an increased sense of community. While objective findings (or sound level measurements) show that on average noise levels in the Rabat Medina are not high enough to cause permanent hearing loss, urban noise levels can create additional types of social and economic impacts that warrant further study.

4aPP14. Multi-dimensional auditory performance in adults with normal audiometric thresholds. Lindsey Kummerer (Commun. Sci. and Disord., Univ. of South Florida, 4202 East Fowler Ave., PCD1017, Tampa, FL 33620, lkummerer@usf.edu), Angelina Natalie (CSD, Univ. of South Florida, Tampa, FL), Robert A. Lutfi, and Jungmee Lee (Univ. of South Florida, Tampa, FL)

Audiologists have several diagnostic tools available to evaluate why some individuals with normal audiograms experience difficulty understanding speech in noise. However, these measures likely capture different aspects of listening in noise. This study used principal component analysis (PCA) to examine how common diagnostic measures organize into linearly independent dimensions in adults with normal hearing. The premise was that different components would reflect different aspects of listening in noise. Measures for an initial PCA included standard and extended high-frequency pure-tone thresholds (PTA and HFPTA), otoacoustic emissions (OAEs), QuickSIN, and self-report questionnaires of listening difficulty in quiet and noise. The PCA (59 participants) revealed three principal components explaining ∼69% of the total variance with the strongest loading on: OAEs (PC1), QuickSIN and questionnaire scores (PC2), and HFPTA (PC3). A second PCA incorporated behavioral efficiency metrics including cue-weighting and noise efficiency, which are not currently used in the clinic, from a speech separation task. This analysis (34 participants) revealed three principal components explaining 61.0% of the total variance with the strongest loading on: OAEs (PC1), QuickSIN and cue-weighting efficiency (PC2), and PTA/HFPTA (PC3). The results underscore the challenge of identifying why many individuals with clinically normal hearing have unusual difficulty understanding speech in noise. [Work supported by NIH R01 DC001262-31.]

4aPP15. The importance of vocal pitch in spatial release from speech-on-speech masking. Pinar Ertürk (Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, perturk@bu.edu), Virginia Best, Christine Mason, and Gerald Kidd (Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA)

Separating multiple talkers in azimuth improves the intelligibility of a single voice (the "target") relative to a colocated presentation. The magnitude of this spatial release from masking (SRM) depends on a number of factors that may interact in complex ways. Here, we assessed the role of vocal-pitch information in SRM by adding noise-vocoded speech back to the natural speech from which it was derived, which varied the integrity of the vocal pitch information in the stimulus while maintaining the temporal envelope. Speech-reception thresholds were measured for colocated and spatially separated conditions as a function of the relative level of the natural-to-vocoded speech (N/V). Colocated thresholds were found to be roughly constant with N/V while separated thresholds declined (improved) with increasing N/V yielding increasing SRM. Acoustic analysis revealed that preserving vocal pitch information was key to producing large SRM.

4aPP16. Review of human auditory localization and the effects of wearing hearing protection. Derek W. Schwaller (Warfighter Health and Protection, Naval Submarine Medical Res. Lab., Bldg. 141, Trout Ave., Groton, CT 06349-5900, derek.w.schwaller.civ@health.mil)

Auditory localization researchers are generally familiar with Interaural Level Difference (ILD), Interaural Phase Difference (IPD), Interaural Time Difference (ITD), and spectral cues. However, these cues are not the only factors that should be considered when assessing the effects that head-worn devices have on auditory localization. Additional factors for consideration include conflicting cues, spatial release from masking, hearing loss or functional differences, sound level, sound duration, motion, cross-sensory inputs, familiarity with the sound, and other top-down effects. Each factor is briefly presented. Most publications focus on azimuth determination and elevation determination, and very few focus on distance determination. The cues for sound localization in the azimuth and elevation are reviewed, with an emphasis on which cue is dominant in each frequency range. The cues used for distances are also reviewed, with an emphasis on which cue is dominant in each distance range. Wearing a hearing protection device can decrease localization performance by attenuating the sound, blocking the pinnae, and introducing non-linear effects. However, when a device initially disrupts sound localization, by just attenuating the sound, a listener can relearn the altered localization cues. The amount of function that can be relearned and the time it takes vary in the literature.

4aPP17. Cross-laboratory validation of the Acoustical Society of America and the American National Standards Institute published S3.71-2019 "Methods for measuring the effect of head-worn devices on directional sound localization in the horizontal plane". Derek W. Schwaller (Warfighter Health and Protection, Naval Submarine Medical Res. Lab., Bldg 141, Trout Ave., Groton, CT 06349-5900, derek.w.schwaller.civ@health.mil), Eric R. Thompson (711th Human Performance Wing, Human Systems Directorate, Air Force Res. Lab., Wright-Patterson AFB, OH), Heath Jones (U.S. Army Aeromedical Res. Lab., Fort Rucker, AL), Stephanie Karch, and Jeremy Federman (Warfighter Health and Protection, Naval Submarine Medical Res. Lab., Groton, CT)

The Acoustical Society of America and the American National Standards Institute standard S3.71-2019, "Methods for Measuring the Effect of Head-worn Devices on Directional Sound Localization in the Horizontal Plane," describes the three main methods for measuring how head-worn devices affect localization in the horizontal plane: (1) a low-complexity method with eight loudspeakers, (2) a more complex method with 36 loudspeakers, and (3) an auditory-guided visual search method with 36 loudspeakers. To cross-validate the standard, three Department of Defense laboratories completed measures on five different hearing protection devices (HPDs). Out of 41 comparisons, 21 reached a statistically significant difference between the three laboratories. For the long duration stimuli, the percent correct was >90 (method 1), error was <5 deg (method 2), and Front -Back reversals occurred $<1\%$ of the time. Method 3 was insensitive to the HPD. Some participants reported that they could not hear some of the stimuli. All three labs evaluating the standard had difficulty finding participants who could pass the required performance screening. As a result, the performance requirement was altered. In addition, there were differences in how the laboratories interpreted the language in the standard. Changes to the standard are needed to make the measures more repeatable across different laboratories.

4aPP18. Identifying areas for improvement of the Acoustical Society of America and the American National Standards Institute S3.71-2019 "Methods for measuring the effect of head-worn devices on directional sound localization in the horizontal plane". Derek W. Schwaller (Warfighter Health and Protection, Naval Submarine Medical Res. Lab., Bldg 141, Trout Ave., Groton, CT 06349-5900, derek.w.schwaller.civ@health.mil)

This presentation discusses potential areas for improvement of the Acoustical Society of America and the American National Standards Institute S3.71-2019, "Methods for Measuring the Effect of Head-worn Devices on Directional Sound Localization in the Horizontal Plane." This presentation is based on a literature review and the use of the standard during a U.S. Department of Defense tri-service project to assess the consistency of results across laboratories. The standard is critiqued from the beginning to the end of the standard. Three areas are assumed to be the main reasons the tri-service cross-validation produced poor results. Those three areas for improvement are (1) the different subject direction indication techniques allowed by the standard, (2) the stimuli approaching inaudible levels with highlyattenuating hearing protection, and (3) the performance screening criteria being too difficult for the subjects. These areas for improvement are discussed, along with several others.

4aPP19. An micro electro mechanical systems-based probe for ultrasound range distortion-product otoacoustic emissions measurements. Suna Ro (Purdue Univ., 420 Central Dr., DUDL, West Lafayette, IN 47907, ro10@purdue.edu) and Noori Kim (Purdue Univ., West Lafayette, IN)

Distortion-Product Otoacoustic Emissions (DPOAEs) provide an objective, noninvasive measure of cochlear outer-hair-cell function, yet extended high-frequency (EHF) range targeted portable instrumentation remains limited. We present a prototype probe for DPOAE acquisition in the 20–40 kHz band. This prototype integrates two Micro Electro Mechanical Systems (MEMS) speakers for primary-tone delivery and a MEMS microphone for in-ear recording. DPOAEs are elicited using a fixed primary ratio of $f2/f1 = 1.2$ with $L1/L2 = 65/55$ dB ratio, and analyzed in the frequency domain at 2f1–f2, with the noise floor estimated from neighboring FFT bins. System characterization includes calibration using a custom coupler. Initial measurements confirm detectable distortion-product components with the discrete testing frequencies within the targeted frequency band, and ongoing work is quantifying noise floor and repeatability. Motivated by the speed and accessibility of OAE-based newborn screening in humans, this platform targets a compact, potentially lower-cost direct cochlear measure for canines and is positioned as a practical supplement to Brainstem Auditory Evoked Response (BAER) and Auditory Brainstem Response (ABR) for veterinary hearing screening and longitudinal monitoring in the EHF range.

4aPP20. Beyond aging: Local sensitivity of hearing health disparities to socioeconomic and environmental factors across the United States. Tianle Duan (Purdue Univ., West Lafayette, IN), Suna Ro (School of Eng. Technol., Purdue Univ., 420 Central Dr., DUDL, West Lafayette, IN 47907, ro10@purdue.edu), Dorde Sabatus, Qingchun Li, and Noori Kim (Purdue Univ., West Lafayette, IN)

Aging is the primary determinant of human hearing loss; however, the roles of socioeconomic and environmental factors on hearing health outcomes remain insufficiently understood. This study integrates explainable machine learning and geospatial analysis by employing a Geographical Random Forest model coupled with SHapley Additive exPlanations (GRF-SHAP) to investigate contributors to hearing health outcomes across the contiguous United States. Results indicate spatial heterogeneity in dominant predictors across countries. Globally, racial minority social vulnerability, the percentage of veterans, and point-of-interest density are the three most

influential non-aging predictors. Higher racial minority vulnerability and higher point-of-interest density are generally associated with better hearing health outcomes, whereas higher veteran concentration tends to be associated with worse hearing health outcomes. Locally, the influence of these variables varies across counties. For example, point-of-interest density emerges as the predominant predictor across counties in New York State, whereas racial minority social vulnerability dominates in counties across Florida. The proposed GRF-SHAP framework offers a scalable and interpretable approach for identifying globally and locally dominant determinants of hearing health. By moving beyond aging-centric explanations, this study demonstrates that hearing health is shaped by a complex interplay of social vulnerability, infrastructure conditions, and community characteristics that vary geographically.

4aPP21. Remote-frequency masking on speech perception in neurotypical and neurodivergent young adults. Yingjiu Nie (Commun. Sci. and Disord., James Madison Univ., 235 Martin Luther King Jr. Way, MS 4304, Harrisonburg, VA 22807, nieyx@jmu.edu), Husna Firdose (California State Univ., Northridge, CA), and Destiny Bruce (Commun. Sci. and Disord., James Madison Univ., Harrisonburg, VA)

This study examined how narrow band maskers affected perception of narrow band speech in spectrally remote regions, comparing neurotypical and neurodivergent young adults. Coordinate response measure sentences and the maskers were filtered into two $^1/_2$-octave-wide frequency bands: low-band (500 Hz) and high-band (2500 Hz). Speech reception thresholds (SRTs) and speech recognition scores (SRSs) were measured under four conditions: (1) low-band speech in quiet, (2) low-band speech with high-band masker, (3) high-band speech in quiet, and (4) high-band speech with low-band masker. Maskers were presented at 70 dB SPL when applicable; SRSs were obtained at 7 dB above each participant's respective SRTs. Preliminary results indicate: (1) Frequency-band effect: High-band speech yielded higher SRSs than low-band speech, while SRTs were similar across bands. (2) Group effect: Neurodivergent participants showed elevated SRTs compared to neurotypical participants across conditions. (3) Interaction: Neurodivergent listeners exhibited distinct SRT patterns as listening environments shifted between quiet and masked conditions and across frequency bands. These findings suggest that SRT and SRS capture different sensitivities to spectral separation and to perceptual discrepancies between neurodivergent and neurotypical listeners. Psychometric functions were fitted to derive slope and lambda parameters, which will be compared across frequency-band conditions, listening environments, and participant groups.

4aPP22. The effect of vibrotactile input on binaural fusion. Nicholas Margolies (Behavioral and Systems Neurosci., OHSU, 3181 SW Sam Jackson Rd., Portland, OR 97239, margolie@ohsu.edu), Holden D. Sanders (Oregon Hearing Res. Ctr., OHSU, Portland, OR), Jonas Brodsky (Oregon Hearing Res. Ctr., OHSU, Salt Lake City, UT), and Lina Reiss (Oregon Hearing Res. Ctr., OHSU, Portland, OR 97239)

Binaural fusion is the perception of one single sound when two different sounds are presented, one to each ear (dichotically). Normal hearing listeners typically fuse tones 0.1 to 0.2 octaves apart. The range over which binaural fusion occurs can be much broader in people with hearing loss, who may fuse tones differing by 3 to 4 octaves. Broad fusion leads to spectral averaging of speech from different talkers, and thus difficulties with understanding speech in noise. This experiment investigated the effect of vibrotactile input on binaural fusion. Binaural fusion, bilateral vibrotactile (bimanual) fusion, and binaural fusion with vibrotactile input were measured in listeners with normal hearing (NH) and bilateral cochlear implants (CI). Binaural fusion was generally broader in CI than NH listeners, consistent with previous findings. Bilateral vibrotactile fusion was similar across groups, and broader than binaural fusion in NH listeners. Preliminary findings suggest that vibrotactile input may moderate binaural fusion. [Work funded by a grant from the Medical Research Foundation of Oregon and an NIH/NIDCD grant R01 DC022777.]

4aPP23. Effects of two types of transcranial electric stimulation on speech understanding in listeners who struggle with speech-on-speech masking. Elin Roverud (Boston Univ., 635 Commonwealth Ave., Dept. of Speech, Lang. & Hearing Sci., Boston, MA 02215, erover@bu.edu) and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Many individuals, both with and without sensorineural hearing loss, have difficulty understanding talkers in speech-on-speech (SOS) masking environments. Noninvasive transcranial electric stimulation (tES) may facilitate neural tracking of the attended speech envelope and improve SOS understanding. In this study, we compared the effects of two tES paradigms: transcranial alternating current stimulation (tACS), which delivers an oscillating current phase-aligned to the target speech envelope to enhance neural entrainment, and transcranial direct current stimulation (tDCS), which delivers a tonic low-level current to enhance neural plasticity. Matching tACS to the target speech envelope has been shown to improve speech-in-noise intelligibility in both younger and older adults during stimulation, and both tACS and tDCS have been shown to facilitate motor learning, cognitive training in older adults, and speech-language therapy in aphasia, sometimes with effects persisting beyond the stimulation period. We examined whether tACS or tDCS enhanced speech understanding in listeners specifically struggling with SOS, whether these benefits are trainable, and whether they persist beyond the stimulation period. Here, we describe preliminary results from our ongoing studies in which participants with demonstrated SOS difficulties undergo multiple-day tACS, tDCS, or sham stimulation during a SOS task. [Work supported by AHRF and NIH NIDCD R21DC021763.]

4aPP24. The art of stereophonic reproduction—A test engineer's perspective. James Weir (Hamilton Audio Design @ Consulting, 3079 Premire Pkwy, Ste. 120, Duluth, GA 30097, jim.weir@bksv.com)

The stereophonic illusion is a miracle. Practitioners of the hobby and even the most casual listener are amazed that just two loudspeakers can create a rendering of a music performance in your living room. More than just a radio playing the song with a beat, melody and lyrics, a left and a right loudspeaker in a triangular relationship with the listener, and a recording that exhibits some reasonable production skills, allow you to hear a "phantom" center image along with the characteristics of space around the artist and accompanying instrumentalist ranging to the left and the right, with height, depth and breadth. There are flaws in the illusion, and in the process of rendering this psychoacoustic illusion passionate participants of this hobby can develop belief systems that conflict with the electro-acoustical engineer analysis and testing of high fidelity products. This paper presents an explanation and proposes that accommodation of both belief systems, "subjectivist" and "objectivist," need not be at odds with each other.

4aPP25. Integrating spatial audio and 360 deg time-lapse imagery for spatial-temporal data exploration. Isaiah V. Mercer (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, mercei@rpi.edu) and Jonas Braasch (School of Architecture/Future of Computing Inst., Rensselaer Polytechnic Inst., Troy, NY)

Immersive audiovisual environments offer new opportunities for communicating long-term, multidimensional data through perceptually meaningful representations. This work presents a method for sonifying and visualizing time-varying data using panoramic time-lapse techniques within Rensselaer Polytechnic Institute's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab). The facility includes a 128-loudspeaker array capable of wave-field synthesis and higher-order ambisonics, paired with an eight-projector system for 360° panoramic video. The system was configured to present both symbolic and recreated data sets, with a focus on techniques that enable direct comparison of temporal patterns. A year-long series of daily 360° photographic images of a single outdoor location was assembled into a panoramic time-lapse to examine seasonal and weather-related changes. Corresponding soundscape

recordings were integrated by presenting ten-second excerpts for each hour of the day, spatialized in 15° increments around the listener. This configuration allows participants to navigate the environment physically and perceptually, selectively attending to different temporal perspectives. The study demonstrates how spatial audio and panoramic visualization can be combined to support the intuitive exploration of long-term environmental data.

4aPP26. Quantifying the perceptual contribution of additional side reflections in the precedence effect. Noel Gutierrez (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY) and Jonas Braasch (School of Architecture/Future of Computing Inst., Rensselaer Polytechnic Inst., School of Architecture, 110 8th St., Troy, NY 12180, braasj@rpi.edu)

The precedence effect enables robust localization of sound sources in reverberant environments by suppressing spatial information from reflections and prioritizing cues from the direct sound. At the same time, early reflections provide perceptually relevant information related to auditory spaciousness, contributing to estimates of room size and reverberance. A recently proposed precedence model (binaurally integrated cross-correlation/auto-correlation mechanism, BICAM) formalizes this dual processing by separating localization cues associated with the direct signal from those associated with multiple reflections. An open question concerns the extent to which the auditory system can extract distinct spatial and room-acoustic information from additional early side reflections beyond the first pair typically considered most perceptually salient. This study investigates the perceptual contributions of two additional early reflections—both specular and diffuse—to robust source localization and room-acoustic parameter estimation. Psychophysical experiments were conducted to quantify changes in localization performance and perceived room attributes as a function of these added reflections. The results provide insight into how additional early reflections influence the balance between localization accuracy and spatial impression within the precedence-effect framework.

4aPP27. Uncovering the latent structure of hearing and speech perception across the lifespan. Sita Carraturo (Dept. of Otolaryngol., Univ. of Iowa, 278 PBSB, Iowa City, IA 52242, sita-carraturo@uiowa.edu), Sarah E. Colby (Univ. of Iowa, Iowa City, IA), Samarium Knight (Dept. of Psych., Univ. of Illinois Urbana–Champaign, Champaign, IL 61820), McKenna Revis, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Adverse listening conditions are common and contribute to hearing health and quality of life. Yet, it is unclear the extent to which there is a single, latent ability to cope with challenges, or if perceptual/cognitive mechanisms are specific to distinct types of adverse listening (noise, fast speech, *etc*.). An impediment to addressing this is that differences in challenging listening may simultaneously reflect age, hearing and cognition in addition to perceptual/cognitive differences. A latent-variable/structural-equation (SEM) approach may offer a unique way to disentangle this web. We thus apply an SEM to characterize the underlying structure of hearing and speech perception in a group of 210 adults ages 30–86. The model demonstrated good fit (CFI = 0.942, TLI = 0.921, RMSEA = 0.078). We found support for two latent auditory factors—detection (audiometric thresholds), and discrimination (gap detection; spectral and temporal resolution)—and one speech perception factor that captured covariance between quiet speech, speech in noise, and time-compressed speech measures. The structural model showed that detection ($p = 0.033$) and age ($p = 0.006$) significantly predicted speech perception, while discrimination and general cognition did not ($p > 0.1$). This approach demonstrates the utility of SEM for reducing the dimensionality of large datasets, and unpacking the causal links between aging, hearing, cognition and speech outcomes.

4aPP28. Spectral weighting functions for localization in simulated and chronic unilateral hearing loss. Sara Momtaz (Ctr. for Hearing Res., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, sara.momtaz@boystown.org) and G. Christopher Stecker (Ctr. for Hearing Res., Boys Town National Res. Hospital, Omaha, NE)

Spectral weighting functions (SWFs) for free-field sound localization were measured in participants with normal hearing (N = 11), simulated unilateral asymmetry via ear plugging (N = 11), and unilateral hearing loss (UHL; N = 12). Methods followed those of Folkerts and Stecker [2022, JASA **151**, 3409]. Stimuli were complexes of seven tonal or noiseband components (100–6400 Hz), with azimuth varying independently across components, and overall level roving ±6 dB across trials. SWFs were computed by multiple regression of localization judgments onto component azimuth. Normal-hearing listeners exhibited SWFs that peaked around 800 Hz, as in previous studies. Listeners with unilateral asymmetry showed more variability in SWFs, with peak weights that differed across groups. SWFs for plugged listeners favored lower frequencies (200–400 Hz), whereas UHL listeners weighted higher frequencies (1600–6400 Hz) more. These results suggest that chronic unilateral hearing loss drives compensatory utilization of higher-frequency spatial cues that remain accessible via monaural mechanisms (e.g., spectral shape cues from head-related transfer functions), whereas temporary ear plugging uniformly degrades spatial cue availability without eliciting adaptation. The divergent spectral weighting strategies have important implications for understanding spatial hearing deficits in UHL and may inform frequency-specific rehabilitation approaches. [Work supported by NIH R01-DC016643, P20-GM109023, U54-GM115458, and Boys Town National Research Hospital.]

Session 4aSA

Structural Acoustics and Vibration: Advanced Modeling and Computational Methods

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Contributed Papers

9:00

4aSA1. Causality preservation test for acoustic flow monitoring: Optimizing gauge length. Peyman Moradi (Baker Hughes, 1600 Springwoods Plaza Dr., Apt 616, Spring, TX 77389, peyman.moradi@bakerhughes.com)

Gauge length is a key configuration parameter in Distributed Acoustic Sensing (DAS) that governs the ability to reconstruct and track flow-induced traveling waves in wellbore fluid, enabling acoustic flow monitoring. This paper utilizes a causality preservation test that converts gauge-length selection for flow monitoring from a heuristic choice to an objective, repeatable optimization. The test uses two fiber stretchers at a known spacing stimulated by an identical waveform, while prescribed inter-stretcher time offsets emulate wave propagation delays within ranges of interest for gas-, oil-, and water-dominated flow regimes. Controlled sweeps of gauge length and stimulus amplitude are performed, and for each setting recovered inter-channel delays are compared against the prescribed offsets to quantify delay preservation and the resulting apparent-velocity accuracy. The reported metrics include delay bias, delay variability, apparent-velocity error, and detection reliability as functions of gauge length and signal-to-noise ratio. The tests are conducted on both standard fiber and enhanced backscatter fiber (EBF) under consistent test conditions. The outcome is a gauge-length selection map that identifies operating regions where wavefront timing is preserved with sufficient fidelity for flow monitoring, enabling a defensible, application-driven configuration for real-time wavefront tracking and related diagnostics.

9:15

4aSA2. Principled inference for isotropized flexural and vibrational characteristics of paired thin films. Kent Eng (Dept. of Civil and Environ. Eng., Princeton Univ., Princeton, NM), Nakul Deshpande (Los Alamos National Lab., Casa Grande Dr., Los Alamos, NM 87544, ndeshpande@lanl.gov), Kane Bennett, Darrel Beckman, Amanda Graff, and Eric Cushman (Los Alamos National Lab., Los Alamos, NM)

In practice, vibrometry-based inverse methods for the mechanical characterization of multi-body systems require substantial specialization for thin polymer films. Thin films are particularly affected by spatial variability of thickness and density, as well as elastic parameters, imposed loads, and anisotropy thereof. This work combines experimental and numerical methods to determine the effective isotropized flexural rigidity of paired thin films, based on parameter inversion across a range of imposed in-plane stresses. To do so, the effects of tensile load on viscoelastic film vibrations are formulated by a "hybrid" variational principle, for the films' out-of-plane displacement, linked to classical plate-buckling theory. The forward problem for parameter inversion, modeled using frequency-domain Finite Element Analysis (FEA), approximately quantifies the influence of variable tensile load on observed resonant frequencies, as shown by comparison to and validation against an experimental dataset. In this manner, material characteristics for paired thin films are inferred from consistent device performance over a range of imposed tensile loads.

9:30

4aSA3. Robust time-domain nonlinear finite element method modeling for cochlear mechanics. Weiyuan Fan (Mech. Eng., Univ. of Michigan, G.G. Brown Lab., 2350 Hayward, Ann Arbor, MI 48109, weiyuanf@umich.edu) and Karl Grosh (Mechancial Eng., Univ. of Michigan, Ann Arbor, MI)

The cochlea is the inner-ear organ responsible for hearing, converting mechanical vibrations into neural signals processed by the brain. It may be viewed as a fluid–structure interaction system that supports traveling waves, while micromechanical processes within sensory cells introduce nonlinearities. To study the cochlea, many linear and nonlinear numerical models have been created and solved in the frequency domain. However, time-domain analysis not only provides a benchmark for nonlinear frequency-domain methods but also offers a more natural framework for studying this nonlinear system, albeit with increased numerical challenges. In this work, we investigate numerical issues that hinder time-domain cochlear simulations and develop a robust time-domain algorithm. Solution uniqueness is examined for both compressible and incompressible fluids using acoustic formulations based on pressure, displacement potential, and velocity potential, with particular emphasis on DC loading effects arising in nonlinear cochlear mechanics. A formulation that decomposes the pressure into static and dynamic components is shown to resolve the non-uniqueness associated with static loading. A simple and effective time-marching method from the literature is implemented to improve numerical performance. The proposed approach is incorporated into a nonlinear cochlear mechanics framework, enabling a more reliable investigation of the biophysics and biomechanics of the mammalian ear.

9:45

4aSA4. Asymptotic modal analysis to predict the forced vibration response of a structural system—String application. Donald J. Nefske (DJN Consulting, LLC, Troy, MI) and Shung H. Sung (SHS Consulting, LLC, 4178 Drexel Dr., Troy, MI 48098-4310, ssung1972@gmail.com)

The Asymptotic Modal Analysis (AMA) method has previously been developed to predict the high-frequency dynamic response of structural and acoustical systems (Sung *et al.*, *Asymptotic Modal Analysis of Structural and Acoustical Systems*, Springer Nature, Switzerland AG, 2021). In the AMA method, the dynamic response is obtained as the asymptotic limit of the Classical Modal Analysis (CMA) solution as the averaged response in frequency bands. In previous work, the AMA method has been applied to uncoupled individual structural and acoustical systems, including a flat plate and a rectangular cavity. In the asymptotic limit, the AMA results for an individual structural or acoustical system have been shown to approach the corresponding Statistical Energy Analysis (SEA) results. The AMA method has also been previously applied to predict the sound pressure and vibration

responses of a coupled cavity-plate system for uniform pressure excitation (white noise excitation) of the plate. In this paper, the AMA method is applied to a simple spring subjected to random and frequency dependent force excitation at the center of the string to predict the vibration of the spring. The asymptotic limit of the AMA results for the spring is shown to approach the corresponding CMA and SEA results. In addition, the AMA method is shown to predict the spatial distribution of the spring vibration response, including the Chladni-type vibration responses at high frequencies.

10:00

4aSA5. Numerical prediction of total harmonic distortion in piezoelectric-driven diaphragms using COMSOL multiphysics. Jomar C. Morales (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Rm. 270, Montreal, QC H3A 0C3, Canada, christopher.morales@mcgill.ca) and Luc G. Mongeau (Mech. Eng., McGill Univ., Montreal, QC, Canada)

Synthetic jet actuators (SJAs) generate jet-like flows without the need for an external fluid supply. The flow is produced by the periodic compression and expansion of a fluid cavity using piezoelectric-driven diaphragms, resulting in alternating in- and out-flow through a dedicated orifice. This capability to produce a continuous-like jet without additional hardware makes SJAs attractive for applications such as sensor cleaning in the automotive industry. However, despite their functional advantages, SJAs can generate unwanted high-frequency noise, which may be a disturbance to users in close proximity. Noise generation originates from structure-borne and fluid-borne monopole sources associated with the vibrating diaphragm and the pulsating flow, respectively. The piezoelectric disks are composed of ferroelectric PZT-5A material, whose inherently nonlinear electrostrictive electromechanical response contributes to the observed total harmonic distortion (THD). COMSOL Multiphysics was used to predict THD in the structural response as observed in experiments by modifying the electromechanical coupling definition and coupling an electrical circuit to the piezoelectric boundary.

10:15

4aSA6. Geometric phase of two coupled Su–Schrieffer–Heeger chains. Abhirup Basu (Mathematics, The Univ. of Arizona, 424 E Waverly St., Tucson, AZ 85705, abhirup.basu56@gmail.com), Pierre A. Deymier, Keith Runge, and Wataru Takeda (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ)

The future integration of topological acoustic materials into technology requires a better understanding of composites of topological systems. We consider two coupled Su–Schrieffer–Heeger chains and analytically find conditions governing the topology of the composite system. The conditions show that coupling two topologically trivial chains can give rise to a topologically non-trivial system. Furthermore, two topologically non-trivial chains can couple to form a topologically trivial system. We can even tune the coupling spring constant to go from trivial to non-trivial topology and *vice versa*. Trivial topology can be visualized as the parallel transport of a vector along a Moebius strip-type manifold having an even number of twists, whereas, in case of non-trivial topology, the manifold will have an odd number of twists. Each contribution of π to the Berry phase corresponds to a twist in the manifold. Our model serves as a prototype for composite topological acoustic structures that can be translated into practical technology.

10:30

4aSA7. Interface coupling modes. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, anthony.l.bonomo.civ@us.navy.mil)

Interface coupling modes are introduced as a means to analyze the interaction of coupled dynamic substructures. Loosely analogous to acoustic radiation modes, interface coupling modes are obtained at each frequency by performing the singular value decomposition of the interface impedance matrix (i.e., the inverse of the sum of the interface mobilities of the substructures being coupled). The right singular vectors define the interface velocity profiles that couple independently, the left singular vectors are the associated force profiles, and the singular values quantify the coupling gain. The coupling gain corresponding to a given Interface Coupling Mode can be tracked over frequency through the use of the modal assurance criterion and the Hungarian matching algorithm. Peaks in plots of the coupling gains versus frequency indicate modes of the coupled system. The additional quantities of interface coupling mode participation factors and transfer functions are defined in the context of frequency-based substructuring. The usefulness of these concepts is illustrated through application to some simple canonical examples.

Session 4aSC

Speech Communication: Exploring Creaky Voice/Glottal Fry Across Disciplines

Susannah V. Levi, Cochair
Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012

Lisa Davidson, Cochair
Linguist., New York Univ., 10 Washington Pl., New York, NY 10003

Jody Kreiman, Cochair
Head and Neck Surgery, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794

Chair's Introduction—8:00

Invited Papers

8:05

4aSC1. Acoustic variability in creaky voice. Jody Kreiman (Head and Neck Surgery, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu), Patricia A. Keating, and Marc Garellek (Linguist, UC San Diego, La Jolla, CA)

Creaky voice is common in human communication and is studied across a very wide range of academic disciplines. At least seven different functions have been attested: marking phonemic contrasts on vowels and/or on consonants; serving as an allophone of a glottal stop; a result of (and a cue to) coarticulation with adjacent consonants; in the production and perception of tonal contrasts; as a marker of prosodic boundaries; as a cue to group membership, personal identity, or emotional state; and as a symptom of a phonatory disorder. This talk will focus on the acoustics of creak across these many functions, in the belief that such descriptions will improve our understanding of the vocal phenomena in question and the meanings they express. These analyses may allow us to compare degrees of creakiness across functions, and may help establish clear descriptions of different types of creakiness.

8:25

4aSC2. Investigating acoustic profiles, perceptual qualities, and distributional properties of period doubling and vocal fry. Yaqian Huang (Univ. of AB, Dept. of Linguist, Edmonton, AB T6E 2G8, Canada, yaqian.huang@ualberta.ca)

In this talk, I present my research on two types of creaky voice in Mandarin, vocal fry (or "creak") and period doubling. Vocal fry is a characterized by low and quasi-regular f0, prominent glottal constriction (resulting in low H1–H2), and highly-damped resonances between pulses. Period doubling consists of alternating glottal pulses that differ in period and amplitude, in which multiple fundamental frequencies (f0s) co-exist. This often leads to an indeterminate pitch percept with a low, rough quality. I first describe their physical and physiological differences using acoustic and electroglottographic (EGG) analyses, showing that period doubling involves a covariation of harshness and breathiness. Next, I discuss the perceptual implications of these voice qualities, focusing on their effects on the perceived naturalness of Mandarin tones. Finally, I argue that, apart from their distinct acoustic profiles, period doubling and vocal fry likely arise from different aerodynamic configurations and serve distinct prosodic functions. Period doubling concentrates at utterance edges, suggesting vocal instability, while vocal fry signals weak prosodic elements around focused content.

8:45

4aSC3. Acoustic correlates of two types of glottalized phonation in Danish. Jailyn M. Pena (Inst. for Phonet. and Speech Processing, Ludwig Maximilians Universität, Akademiestraße 7, Munich, Baveria 80799, Germany)

Danish has multiple linguistic sources of glottalization, including stød and word-initial glottal stops. Word-initial glottal stops are non-contrastive and can surface as a full closure or creak, e.g., [ˈʔɛltʁɐ ~ ˈɛ̄ltʁɐ] "older." In contrast, stød is a phonological voice quality that contrasts with modal voice and often manifests with creak, though sometimes also with full glottal closure, e.g., [vɛnā(ʔ)] 'Wait!' vs [vɛn] "friend" (Fischer-Jørgensen, 1987; Peña, 2024; Schachtenhaufen, 2013). In this study, sequences of words with modal voice were compared to sequences with (1) word-initial glottal stops, (2) stød, or (3) both, to investigate whether there are acoustic differences in glottalization based on its underlying source in Danish. The results of a random forest classification analysis show that both stød creaky voice and creak from word-initial glottal stops are differentiated from modal voice by strength of excitation (SoE), H1* residual, and cepstral peak prominence (CPP). Furthermore, generalized additive mixed models (GAMMs) revealed significant differences in the time-course trajectories of these correlates, reflecting different alignment patterns between creak and its underlying linguistic source. Finally, the GAMMs also revealed that sequences with both stød creak and word-initial glottalization were produced with lower SoE and CPP that sequences with only a single source of glottalization, suggesting that the effects may be cumulative.

9:05–9:15 Break

Contributed Paper

9:15

4aSC4. Temporal relations between two creaky voices in Zapotec. Mykel L. Brinkerhoff (Dept. of Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195-2425, mlbrinke@uw.edu)

Voice quality research primarily focuses on the overall quality of phonation during production. This quality is often assumed to exist throughout the duration of the entire vowel. In conducting acoustic analyses, the researcher often examines the average of various acoustic measures. One area that is sometimes ignored is the temporal relationships between these phonations, especially in linguistic uses of voice quality. In most Zapotec languages, there are two types of creaky voice, which researchers descriptively call checked and rearticulated. These two types of creaky voice are only differentiated by the temporal relationship of the creaky voice during vowel production. Evidence from a generalized additive mixed model on data from 10 speakers (five male and five female, all over the age of 18) of Santiago Laxopa Zapotec confirms a temporal difference between these two types of creaky voices. Checked voice is characterized by creaky voice in the final third of the vowel, and rearticulated voice is characterized by a creaky voice in the first half of the vowel. These results suggest that temporal relationships need to be considered during acoustic analyses.

Invited Papers

9:30

4aSC5. Examining vocal fry in clinical research: History and future directions. Maxine Van Doren (English, Writing, and Lang. Sci., California State Univ., San Bernardino, 5500 University Parkway, San Bernardino, CA 92407, maxine.vandoren@csusb.edu)

Definitions and sentiment of "vocal fry" have shifted over time in the clinical literature. Although recent clinical voice research often lists vocal fry as pathological or potentially phonotraumatic, this view has not been consistently supported by empirical evidence. The purpose of this invited talk is twofold: first, we examine the history of "vocal fry" and related terminology (i.e., creaky voice) in the clinical voice literature, including how definitions and sentiment have changed with advances in technology and research methods. Second, this talk considers future directions for clinical research and practice, focusing on the application of linguistic theory to clinical voice analysis. This method accounts for linguisticallymotivated sources of vocal fry in order to examine quantitative and qualitative differences between typical and dysphonic speakers. Preliminary results of a case series of two speakers suggest that some speakers with dysphonia may use vocal fry in qualitatively similar ways to typical speakers. This finding is also important for clinical practice: it suggests that what is identified as pathological vocal fry is not necessarily different from typical speakers' voicing patterns. Together, these observations motivate a reexamination of how vocal fry is defined, measured, and interpreted in clinical contexts.

9:50

4aSC6. Creak as an automated acoustic measure in speakers with adductor laryngeal dystonia. Katherine L. Marks (Otolaryngol., Emory Univ., 550 Peachtree St., Ste. 1135, Atlanta, GA 30308, katie.marks@emory.edu), Manuel E. Díaz Cadiz (Speech Lang. and Hearing Sci., Boston Univ., Boston, MA), Saul Frankford (Speech, Lang., and Hearing, UT Dallas, Richardson, TX), Daria Dragicevic (Speech Lang. and Hearing Sci., Boston Univ., Boston, MA), Maria Eugenia Castro (Otolaryngol., Univ. of Southern California, Los Angeles, CA), and Cara E. Stepp (Speech Lang. and Hearing Sci., Boston Univ., Boston, MA)

Creak has been defined by Keating and colleagues as an umbrella term for irregularities in fundamental frequency. Creak was found to be the most prevalent acoustic feature in speakers with adductor laryngeal dystonia (AdLD). This finding is critical, as there are currently no widely accepted measures that aid a differential diagnosis of this rare neurological disorder. In this invited talk, the author will cover several studies which investigated creak as an acoustic measure in speakers with AdLD. In a series of studies, an open-source creak detector (COVAREP; Drugman & Kane) was implemented via custom MATLAB script to yield % creak as creak time divided by total voiced time. Creak (%) was statistically compared between groups of speakers with and without laryngeal dystonia via analyses of variance and receiver-operating characteristic curves. Across three studies, there have been statistically higher percentages of creak in speakers with AdLD compared to controls, speakers with primary muscle tension dysphonia, and speakers with glottic insufficiency ($p < 0.05$). Results will be presented by group comparisons and speech stimuli used. Creak has demonstrated discriminant validity in several studies differentiating speakers with and without AdLD. [Work was supported by NIDCD grants R01DC015570, F32DC015570, and R21DC023280.]

10:10–10:20 Break

10:20

4aSC7. An update on the social evaluation of creaky voice in American English. Sarah R. Bellavance (Communicative Sci. and Disord., New York Univ., 665 Broadway, 6th Fl., New York, NY 10012, srb664@nyu.edu), Lisa Davidson (Linguist, New York Univ., New York, NY), and Susannah Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

The social evaluation of creaky voice (vocal fry) in American English has held the interest of the public and researchers alike for nearly 100 years and has seen increased interest over the last decade (Van Doren & Bellavance, 2025). In more recent years, this interest has been especially focused on its use in young women. Despite previous work showing that creaky voice is often negatively evaluated when produced by young women, recent work has shown that this may be too broad of a generalization. This talk will provide an overview of two studies. The first study found that when young women produce creaky voice on the last syllable of a sentence, only some listeners evaluate these sentences negatively (Bellavance & Levi, under review). The second study found that when creaky voice occurs

over more syllables of the sentence, listeners negatively evaluate *both* women and men. Taken together, these two studies challenge the implications from previous work that young women should avoid producing creaky voice.

10:40

4aSC8. Rethinking "young women's creak": Piecing together production, perception, and social evidence. Jeanne Brown (Linguist, McGill Univ., 1085 Ave., du Docteur-Penfield #104, Montreal, QC H3A 1A7, Canada, jeanne.brown@mail.mcgill.ca)

Since the early 2010s, popular narratives alongside influential academic publications have presented creaky voice as a hallmark of young women's speech. Drawing primarily on sociophonetic work on Canadian English, this talk integrates acoustic, perceptual, social, and causalmodeling approaches to argue for a more nuanced understanding of creaky voice. Acoustic analyses reveal that men and older speakers exhibit more creak than young women, with fundamental frequency (f0) partially mediating these gender- and age-related differences. A complementary perceptual study shows that listener judgments of creakiness are driven more by acoustic cues such as low f0 than by social cues like perceived female gender, highlighting a divergence between experimental and popular perceptions of creak. I argue that, together with previous work on gender and creak, these results provide little empirical support for the notion that young women are creakier than other speakers—contrary to popular belief. Capturing the complexity of creak requires an integrative approach that considers interactions between acoustic, perceptual, and social factors, rather than treating any single dimension or demographic as explanatory. More broadly, this work challenges widely held sociocultural assumptions about speech patterns and offers directions for future research on voice quality. [Work supported by FRQSC Grant 373124.]

11:00

4aSC9. Cross-linguistic comparison of creaky voice prevalence across German and English. Hannah White (Ludwig Maximilians Universität, Akademiestraße 7, München 80799, Germany, hannah.white@lmu.de)

This study investigates the extent to which non-contrastive creaky voice is used in German compared to American English. Evidence suggests that creak is used as a form of boundary strengthening vowel-initially and as a finality-marker in both English and German (Davidson, 2020), but when it comes to studies of creak prevalence, languages other than English have received relatively little attention. In English, there is a strong association between creak and women's speech, although empirical findings are mixed (e.g., Podesva, 2013; Abdelli-Beruh *et al*., 2016). In German, preliminary work has indicated that creak may serve as a sociolinguistic marker of gender (Penney *et al*., 2025). The relationship between creak and gender, as well as language background, is investigated further here with speech data from 53 monolingual German and American English adolescent women and men from the RUEG corpus (Wiese *et al*., 2024). Creak was identified using an automatic method optimized by-speaker. The output of a GLMER showed a significant interaction between gender and language: both English-speaking women and German-speaking men were creakier than German-speaking women, providing further evidence that creak may be playing a socio-indexical role in German. Future work will explore whether creak plays a salient role in perception in German.

11:20–11:50 Panel Discussion

Session 4aSP

Signal Processing in Acoustics: Signal Processing Potpourri – II

Contributed Papers

8:30

4aSP1. Two-point statistics for signals with intermittent scattering and incomplete saturation. D. Keith Wilson (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Morse Hall, Rm. 213, 8 College Rd., Durham, NH 03824, Keith.Wilson@unh.edu) and Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Previously, it was shown that the probability distribution for the product between partially correlated signals (phasors) at two receivers, under conditions of full saturation, is given by a *variance gamma* distribution. Mathematically, this distribution models the cross correlation between two zero-mean, normally distributed variables. Furthermore, when the strength of the scattering process varies in space and/or time, as commonly occurs in real scenarios, the variance gamma can be extended to a new distribution called the *compoundvariance gamma*. This presentation provides an approach to extending the treatment to randomly scattered signals with nonzero mean, i.e., to incomplete saturation. The approach is based on expanding the distribution as a series in terms of the Rice factor, which represents the ratio of unscattered to scattered signal intensity. Each term in the series (mixture) is a complex variance gamma distribution, with the first term corresponding to full saturation. The full distribution, called the *noncentral compound variance gamma*, is applicable to a broad range of scattering phenomena and propagation geometries, with most existing, commonly used statistical models for random scattering being special cases.

8:45

4aSP2. Exploring effects of tone mapping on buried targets using synthetic aperture sonar. Miah Miner (Arizona State Univ., 17020 W Soft Wind Dr., Surprise, AZ 85387-2011, mminer4@asu.edu), Thomas E. Blanford (Univ. of New Hampshire, Durham, NH), and Suren Jayasuriya (Arizona State Univ., Tempe, AZ)

Synthetic aperture sonar (SAS) is a sonar imaging modality that uses the motion of a platform for synthesizing an array in order to create high-resolution images of the scene being scanned. In particular, SAS with low frequency has the capability to penetrate the seafloor and observe partially and fully-buried objects in the sediment. However, visualizing these objects can be difficult due to the scattering effects of the sediment surrounding them. This project investigates using tone mapping techniques to better visualize buried targets. A custom dataset was collected from a controlled in-air SAS (10–30 kHz) consisting of nine different targets of varying geometries and angles in a pit of plastic pellets to simulate diffuse background scattering of the seafloor. Pellets were incrementally added to the scene to simulate various levels of burial, and each scan had a newly disturbed and unique sediment pattern. A comparison of tone mapping algorithms was performed with respect to this dataset, and it was observed that trigonometric difference tone mapping was very effective in suppressing the noise induced by the seafloor while preserving and visualizing the geometric scattering of the target.

9:00

4aSP3. Performance weighted mixtures of latent adaptive filters. Kanad Sarkar (Elec. and Comput. Eng., Univ. of Illinois at Urbana Champaign, B10 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801-2447, kanads2@illinois.edu), Manan Mittal (Elec. and Comput. Eng., Stony Brook Univ., Stony Brook, NY), Diego A. Cuji (Elec. and Comput. Eng., Stony Brook Univ., Boston, MA), Ryan M. Corey (Elec. and Comput. Eng., Univ. of Illinois Chicago, Chicago, IL), and Andrew C. Singer (Stony Brook Univ., Stony Brook, NY)

Latent adaptive filters improve convergence by performing adaptation in a low-dimensional latent space that serves as the decoder input, learned from acoustic impulse responses using an autoencoder. This latent-space adaptation can improve system identification and tracking performance relative to conventional adaptive filtering methods. In this work, we investigate two strategies for blending latent adaptive filters to improve spatial robustness and training efficiency. The first approach blends multiple latent adaptive filters with different spatial constraints, thereby incorporating more spatial information into an existing latent adaptive filtering framework. The second approach blends latent adaptive filters with traditional adaptive filters, allowing effective adaptation while reducing the number of samples required to train the decoder. Simulation results indicate that these blending strategies improve robustness to spatial variation and reduce decoder training requirements without sacrificing adaptation performance.

9:15

4aSP4. An adaptive implementation of the variable resolution and detection receiver. Radienxe Bautista (Sensors and Sonar, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, rbautista@umassd.edu) and John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

Active sonar systems perform range estimation using pulse compression, where a high-bandwidth pulse is transmitted, and echoes are typically processed with matched filters. Matched filters optimize output signal-to-noise ratio (SNR) for a single target in white noise while range resolution is limited by the pulse bandwidth. This work explores methods to adaptively improve range resolution over the matched filter. Sharma and Buck (2011) proposed the variable resolution and detection receiver (VRDR), which adjusts its range resolution between the matched filter and the inverse filter with one parameter, trading off detection gain for range resolution. In practice, the VRDR requires prior knowledge of the target and noise background to choose the best resolution and detection gain tradeoff. A new receiver is proposed to blend VRDR receivers with different detection and resolution tradeoffs to adapt to the environment. Each VRDR receiver output is weighted based on performance, using a mixture of experts' approach inspired by universal linear prediction [Singer and Feder, 1999] to implement the blending and minimize the regret of using a fixed individual VRDR filter. Preliminary results show the blended VRDR can improve range resolution while maintaining detection performance for many scenarios by adapting to the background SNR. [Work supported by NUWCDIVNPT.]

9:30

4aSP5. Real time gunshot detection using intermediate fusion of blind source separated early and late arriving muzzle blast waves. John Irungu (Comput. Sci., Univ. of the District of Columbia, 4200 Connecticut Ave., NW, Washington, DC 20008, john.irungu@udc.edu)

Real time gunshot detection, particularly in high-crime areas, could provide stronger, more localized signal intelligence to first responders and enhance threat surveillance and crime mapping. In this study, a fusion strategy of blind source-separated muzzle early and late arriving muzzle blast waves is investigated for gun caliber classification. Using blind source separation (BSS),raw early and late arriving acoustic waves from the muzzle blast are decomposed, measured, and separated. Intermediate fusion is then applied to model the early and late arriving features for the proposed binary classification. Evaluation from the experimental study demonstrates that fusion of muzzle blast features enhance the robustness of gunshot detection by differentiating between high- and low-caliber gunshot threats.

9:45

4aSP6. Real-time detection of microplastics in water using ultrasonics. Christopher Verlinden (Appl. Ocean Sci., Fairfax Station, VA), Theodore F. Argo (Appl. Res. Assoc., Inc., Littleton, CO), EeShan Bhatt (Appl. Ocean Sci., 5242 Port Royal Rd. # 1032, Springfield, VA 22151, eeshan.bhatt@appliedoceansciences.com), Jennifer Brandon (Appl. Ocean Sci., San Diego, CA), Kathryn Verlinden (Appl. Ocean Sci., Fairfax Station, VA), and James Murray (Appl. Ocean Sci., Stafford, VA)

Applied Ocean Sciences has developed a Hyperspectral Ultrasonic MicroPlastic Bulk Acoustic Characterization (HUMPBAC) instrument to detect and characterize the presence of microplastics in water in real-time. Microplastics are defined as particles smaller than 5 mm and have been found in ocean and river water, sea ice, soil, atmosphere, drinking water, our food supply, and accumulating in our human body, causing a growing list of human health concerns. However, current detection methods rely on expensive methods (in terms of time and money) such as FTIR and Raman Spectroscopy. HUMPBAC's patent-pending method provides for an affordable, flow-through, real-time system that fits on a benchtop and will characterize the microplastics by density, concentration, size, and shape. HUMPBAC can detect particles down to 5 μm. HUMPBAC employs broadband acoustics that range from 1–100 MHz to analyze the response of the waveguide (water and plastics). Utilizing machine learning (ML), the data can be quickly organized to characterize the microplastics in the sample. This paper illustrates the methodologies associated with the ultrasonic method, addresses the variables, such as EF interference and temperature dependence, and results of the instrument will be shown. Finally, the commercialization possibilities for drinking water plants, water treatment plants, and limnology/oceanography will be discussed.

10:00–10:15 Break

10:15

4aSP7. HelmholtzNet: Physics-informed graph neural network for acoustic pressure estimation. Jhon A. Castro-Correa (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, 3rd Fl., Newark, DE 19716, jcastro@udel.edu) and Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

We present a method for predicting the ocean acoustic pressure field from partially observed pressure magnitudes across range–depth locations. Our framework leverages a physics-informed graph neural network (PIGNN), in which graph connectivity encodes the local spatial structure of the waveguide and enables stable learning of highly oscillatory fields. The PIGNN is trained to map range and depth coordinates to complex acoustic pressure while enforcing the partial differential equation (PDE) governing underwater sound propagation. In large-scale environments, pressure fields often exhibit rapid phase fluctuations over kilometer-scale distances, even at frequencies below 100 Hz, leading to highly complex solution landscapes that standard neural networks struggle to approximate. A key contribution of this work is demonstrating that graph-based message passing, combined with the envelope function derived from the parabolic equation (PE) model, effectively mitigates this challenge by capturing local phase relationships and yielding a substantially smoother learning target. By jointly incorporating the governing PDE and the spatial structure of the environment, the PIGNN achieves accurate pressure predictions with limited training data and improved generalization. [Work supported by ONR (Code 322).]

10:30

4aSP8. Abstract withdrawn.

10:45

4aSP9. Advancing adaptive Limaçon methods for improved direction of arrival in reverberant environments. Adam D. Kingsley (Sentient Digital Inc., 6593 Merchant Pl., Warrenton, VA 20187, adam.kingsley@rdainc.com)

In reverberant underwater environments, it is desirable to increase the directivity of the receive system and thus improve the signal-to-noise ratio. An existing architecture, which increases the directivity of a sensor, involves a combination of isotropic and vector sensors that are leveraged to obtain a limaçon beampattern. At Sentient Digital Inc. (SDi), researchers have implemented established adaptive limaçon methods to minimize noise, including the interference from reverberant environments. SDi has built upon these adaptive methods to improve direction of arrival estimates of coherent signals by weighting the resulting set of limaçons by their individual ability to minimize the noise. [Work funded by the U.S. Navy.]

11:00

4aSP10. Development and debugging of a real-time convolution system to assess vocal health of teachers. Bryce A. Lundstrom (Phys. & Astronomy, Brigham Young Univ., BYU Dept. of Phys. & Astron., N284 ESC, Provo, UT 84602, bal83@student.byu.edu), Brian E. Anderson, Bethany Wu (Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Brian A. Mazzeo (Elec. & Comput. Eng., Brigham Young Univ., Provo, UT)

This presentation describes the creation of a real-time convolution system with off-ear headphones and a cheek microphone, and the efforts made to clean up and improve its performance. This system utilizes a multi-board processing approach to improve the latency of the system, making it possible to convolve speech for a talker in real-time in different environments. The convolution computation is split into manageable chunks by having different boards convolve the same speech segments with different time segments of the impulse response and then superposing their results. The efforts and problems encountered in the development and optimization of this system are explained. This system is being developed to assess the vocal effort of teachers in simulated K-12 classrooms with varying acoustical properties, due to the prevalence of vocal problems in that profession. This system allows the talker to stay in the same physical location (an anechoic chamber) rather than move them from one physical room to another room.

11:15

4aSP11. Realizing gains provided by pulse compression and beamforming in active sonar processing. Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com)

Active sonar processing benefits from having to detect a known waveform, the transmitted waveform. Such waveforms are typically wideband, enabling pulse compression via a matched filter at the receiver. When the receiver is an array, how can we simultaneously realize the benefit of both temporal processing gain (via a matched filter) and spatial processing gain (*via* some form of beamformer)? Conventional beamforming and matched filtering are both linear processes, so they commute (we can apply them in any order). However, more powerful beamformers, such as adaptive beamformers, are non-linear and typically require multiple snapshots to be averaged to form the covariance matrix used to calculate the adaptive weights. Furthermore, reverberation may not be stationary. We will assess how well different beamformers reduce various forms of interference in typical active sonar scenarios when integrated with matched filtering.

11:30

4aSP12. Analyzing the effects of physical task stress on speech production characteristics. ZAHRA OMIDI (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas—Dallas, 800 W Campbell Rd., Comput. Sci., EC 31, Richardson, TX 75080, zahra.omidi@utdallas.edu) and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas—Dallas, Richardson, TX)

Speech produced under physical task stress differs from speech produced in resting conditions due to changes in respiratory activity, vocal effort, and overall physiological load. Such conditions commonly arise in operational and physically demanding environments. This work focuses on analyzing how physical task stress affects speech production characteristics across acoustic, prosodic, and temporal dimensions. The study is motivated by speech data collected under physically demanding conditions, including resources such as the UT-Scope corpus, which capture speech produced across varying levels of physical exertion. Rather than treating stress as a generic source of variability, this analysis considers physical task stress as a physiological factor that reshapes speech production mechanisms. Key aspects of interest include stress-related changes in intensity, pitch behavior, spectral structure, articulation rate, and temporal organization of speech. The goal of this work is to provide a clearer characterization of how speech patterns evolve under increasing physical load and to identify speech attributes that are most sensitive to task-induced stress. By improving understanding of these effects, this study aims to motivate the development of stress-aware speech representations and evaluation frameworks for robust speech systems operating under physically demanding conditions.

11:45

4aSP13. Reduction of total harmonic distortion in the acoustic response of synthetic jet actuators using a spectral least-mean square adaptive filter. Jomar C. Morales (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Rm. 270, Montreal, QC H3A 0C3, Canada, christopher.morales@mcgill.ca) and Luc G. Mongeau (Mech. Eng., McGill Univ., Montreal, QC, Canada)

Synthetic jet actuators (SJAs) are a promising technology in the automotive industry, particularly for sensor-cleaning applications. One limitation of SJAs is poor noise quality caused by total harmonic distortion (THD), in which significant acoustic energy is generated at harmonic frequencies. Synthetic jet actuators are inherently complex multiphysics systems that couple structural, fluid, electrical, and acoustic phenomena, making it challenging to derive an analytical methodology for reducing THD. Adaptive filtering techniques are commonly employed for noise cancellation, with the Filtered-x Least-Mean-Square (FxLMS) algorithm being widely applied in active noise control. However, FxLMS requires additional hardware such as microphones and secondary sound sources, making it impractical for sensor-cleaning applications. Alternatively, a spectral least-mean-square (SLMS) algorithm was previously developed to minimize THD by directly shaping the input signal such that the response is dominated by the fundamental harmonic, while the response at higher-order harmonics is suppressed. This study extends the application of the SLMS algorithm to improve the noise quality of SJAs through THD reduction. This is achieved in two ways: first, by directly reducing THD in the acoustic response using the noise signal as the output, and second, by reducing THD in the electrical current draw of the actuator.

Session 4aUW

Underwater Acoustics and Physical Acoustics: Session in Honor of Thomas G. Muir

Megan Ballard, Cochair
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Preston S. Wilson, Cochair
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Chair's Introduction—8:30

Invited Papers

8:35

4aUW1. The combustive sound source and high amplitude alternatives inspired by Tom Muir. Andrew R. McNeese (Appl. Res. Labs, The Univ of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu) and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Tom Muir inspired an extensive history of underwater acoustics research at the Applied Research Laboratories at the University of Texas at Austin. Much of this work includes his efforts in developing novel high-amplitude underwater acoustic sound sources for use in underwater acoustic experiments. This work began with the development of the Combustive Sound Source (CSS) in the early 1990's, which led to the basic understanding of the CSS and the parameters required to modify its output. In the subsequent decades after the 1990's, Tom remained involved in the CSS development, with work to tune the acoustic output of the source to a variety of experimental needs. This talk will reflect on the guidance Tom provided throughout the development of the CSS for several decades, and it will discuss how much of his early work laid the groundwork for future iterations of the CSS as well as other alternative underwater sound sources. Stories will be shared about field tests and time spent with Tom during this time.

8:55

4aUW2. Physics-based interpretation of broadband modal dispersion in flat and gently sloping shallow-water waveguides. Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, Rm. 140, Evans Hall Rm. 140, Newark, DE 19716, USA,badiey@udel.edu), Sebastian Rojas-Ortega, and Christian D. Escobar-Amado (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

Broadband propagation in shallow-water waveguides produces modal dispersion curves whose geometry reflects both seabed layering and oceanographic variability. We develop a physics-based framework to interpret these dispersion relationships for flat and gently sloping waveguides, and we compare theoretical predictions to field observations from the Seabed Characterization Experiments (SBCEX) in 2017, 2021, and 2022. For the range-independent (flat-bottom) case, the acoustic field is represented as a normal-mode superposition with depth-dependent mode functions and frequency-dependent horizontal wavenumbers determined by layered water–seabed boundary conditions. The dispersion relationship is interpreted through mode-dependent group slowness, yielding frequency–time arrival branches that map directly to observed interference ridges in broadband spectrograms. To extend the interpretation to gently range-dependent bathymetry, we introduce a Webster-type formulation in which gradual depth variation acts analogously to an acoustic horn, producing adiabatic evolution of modal wavenumbers and predictable deformation of dispersion-curve trajectories. The theory explains the robustness of low-frequency modal structure and the increased sensitivity at higher frequencies to sound-speedprofile fluctuations and seabed stratification. These results establish dispersion-curve observations as compact, physically interpretable diagnostics for shallow-water environmental structure and variability. [Work supported by ONR Code 322.]

9:15

4aUW3. Porous medium model for under-ice propagation loss. Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, chotiros@utexas.edu)

The underside of the Arctic ice contains ice keels (downward protruding extensions) that have a significant impact on under-ice sound propagation. Current models attempt to model the propagation loss as either the deflection of sound rays by the facets of solid keels or an equivalent reflection loss that compensates for their presence. An alternative model is proposed in which the keels are treated as porous structures. Keels are generated when ice floes collide. Blocks of ice that break away from the colliding ice floes are swept above to form ridges, and below to form keels. They are porous structures consisting of randomly assembled ice blocks. A finite element simulation is used to estimate the effective sound speed and attenuation in a field of random ice blocks. The results are used to produce an effective medium layered model of the average sound speed and attenuation as a function of depth below the ice floe for estimating the average acoustic propagation loss. [Work supported by the US Navy's Office of Naval Research, Code 322OA, Grant N00014-23-1-2522.]

9:35

4aUW4. A decade of direct sediment acoustic measurements inspired by Tom Muir. Megan Ballard (Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu), Kevin M. Lee, Andrew R. McNeese (Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Tom Muir played an instrumental role in initiating our investigation of direct measurements of sediment acoustic properties, a line of research we have continued to develop over the past decade. Our work began with controlled laboratory experiments that produced a unique set of measurements of shear-wave speed and attenuation in kaolinite-based muds, using circa-1970 bender elements provided by Tom. From the laboratory, we moved to shallow-water estuarine environments, employing spear-like probes that could be deployed by hand or mounted in a small frame, again using instrumentation derived from Tom's early work in sediment acoustics. In the next iteration, we developed the acoustic coring system, which incorporates compressional- and shear-wave probes mounted on the penetrating tip of a gravity core. This system was also directly inspired by Tom, who provided us with a nose cone designed to mount acoustic sensors—another artifact from the earlier work in sediment acoustics. Most recently, we have developed an acoustic multicore system that enables *in situ* measurements of surficial sediment acoustic properties with minimal disturbance. This presentation describes the evolution of these measurement systems, highlights Tom's influence throughout their development, and presents representative results obtained using each approach.

9:55–10:10 Break

10:10

4aUW5. Why not hit it with a hammer? Tom Muir's radical approach to investigating slow dynamics. John M. Cormack (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA 15621, cormackjm@upmc.edu) and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Slow dynamics in natural and man-made rocks such as sandstone, limestone, and concrete is characterized by a reduction in the stiffness due to a large-amplitude vibration. After the vibration ends, the rock stiffness remains reduced but recovers to the original value over the course of hours. Conventional laboratory studies of slow dynamics investigated the material response to narrowband stimuli, e.g., nonlinear resonant spectroscopy or propagation of a tone burst. In 2017, Tom Muir obtained a block of sandstone with a novel and "impactful" approach in mind: investigating slow dynamics due to impulsive stimuli. Tom built an apparatus in which the heaviest hammer he had in his garage impacted one end of a long sandstone bar. Longitudinal motion of the sandstone along its length was measured with a laser vibrometer as the resulting unipolar strain pulse bounced back and forth. Slow dynamics manifested as an initial reduction in the round-trip time of the pulse from one end to the other that slowly recovered with each round trip. Follow-up dynamic acoustoelastic testing revealed that the softening was due almost entirely to the tensile phase of the impulsive deformation. This presentation is a summary of Muir *et al.* [JASA **147**, 1006–1014 (2020)].

10:30

4aUW6. Thomas G. Muir's impact on nonlinear acoustics. Mark F. Hamilton (Appl. Res. Labs., Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Tom Muir was known in nonlinear acoustics while still a graduate student in the 1960s. Working for Applied Research Laboratories at UT Austin (ARL:UT), in February 1969, Tom and his colleague Joe Blue published in JASA the first open-water measurements of the parametric array, confirming the key features of Westervelt's theory. This publication resulted in Tom becoming chair of what is now called the second International Symposium on Nonlinear Acoustics (ISNA), held at ARL:UT in November 1969. ISNA currently circulates between North America, Europe, and Asia every three years. Tom earned his Ph.D. in Mechanical Engineering at UT Austin in 1971 under the co-supervision of David Blackstock, who started at ARL:UT in 1969. Over the years, the combination of Tom's experimental work and David's theoretical work made ARL:UT an international center for nonlinear acoustics, attracting visiting scientists from around the world. Tom's contributions to nonlinear acoustics included measurements of acoustical saturation, focused nonlinear sound beams, arrays of parametric receiving arrays, finite-amplitude reflection, difference-frequency imaging in sediments, nonlinear absorption in tissue, and acousticallyinduced slow dynamics in sandstone, among others. Tom's approaches to experiments in acoustics, both linear and nonlinear, were innovative, and his technical presentations inevitably engaged and entertained his audiences.

10:50

4aUW7. Parametric acoustic array and T. G. Muir: Singular contributions. Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu)

A seminal contribution to parametric array acoustics was made by T. G. Muir, presumed collaboratively with J. E. Blue in 1969 [T. G. Muir and J. E. Blue, J. Acoust. Soc. Am., **46**, 227 (1969)]. They answered the question first posed by Westervelt at the moment of discovery of the parametric acoustic array. When walking in front of a transducer designed for underwater use but sitting on a bench in a laboratory and transmitting in air at its resonant frequency of 18 kHz, Westervelt heard sound at hundreds of hertz. Where exactly did the nonlinearity originate that accomplished this profound demodulation? The question was answered to the theoretician's satisfaction by the theory published in 1963. However, the empiricist in Muir demanded demonstration by experimental measurement. This was accomplished in *Op. cit*., reviewed here. A consequence for sonar development was confidence in understanding the phenomenon, hence in the design of a 100-kW parametric transmitter for underwater application.

Contributed Paper

11:10

4aUW8. Some reflections on Tom Muir as a mentor. 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)

Tom Muir was one of my MS thesis advisors and hence one of my early scientific mentors. After receiving my MS, I worked for Tom as a research engineer and began to learn from him on a different level, ultimately finding my place in underwater acoustics. After a few years of helping Tom pursue his research program, he was one of the people who encouraged me to pursue a Ph.D. (at a different institution) and hence greatly influenced my later life. Finally, many years later, I worked with Tom again, where he continued to mentor me and also mentored junior and not-so-junior members of my own research group. I will share some of the mentoring practices and characteristics of Tom that helped me, and that I have continued to use when mentoring my own students and junior colleagues.

11:25–11:55 Panel Discussion

THURSDAY AFTERNOON, 14 MAY 2026 SALON C, 1:00 P.M. TO 4:25 P.M.

Session 4pAA

Architectural Acoustics, Musical Acoustics and Psychological and Physiological Acoustics: Audio for Architectural Acoustics, Indoors and/or Outdoors

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

Bennett M. Brooks, Cochair
Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062

Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. New methods for evaluating the quality of audio support for speech in lecture rooms. Tiago F. Mattos (Brooks Acoust. Corp., Pompano Beach, FL), Francisco Monteiro (CrossLite+, F. MonteiroScience, Ottawa, ON, Canada), and Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., #121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com)

Proposed loudspeaker layouts for a new lecture room design were compared: (1) a distributed ceiling-speaker concept and (2) an alternative based on a compact front cluster above the main displays. For both speaker layouts, the room's architectural ceiling and wall finishes provide substantial absorption at mid and high frequencies, and calculated metrics of reverberation time and speech transmission indices fall within a favorable range, indicating a controlled teaching space. However, those standard metrics do not address the design goals of spectral neutrality, spatial coherence, and a stable, intuitive auditory image tied to the teacher and visual displays. Using novel

4p THU. PM

techniques to analyze the room response to audio inputs, including frequency-dependent coverage, seat-to-seat tonal shifts, comb-filtering, and visual-to-audio focus matching, the distinct advantages of the central cluster became apparent. While both system layouts provide effectively uniform coverage over the seating area, the cluster offers a slightly higher overall level and additional headroom, with the front-to-back gradient well within accepted practice for speech reinforcement in classrooms. The key distinction between the two options is not SPL coverage, but the comb-filtering, arrival-time structure, and psychoacoustic behavior, all of which favor the coherent front-cluster approach for a higher-quality learning environment.

1:25

4pAA2. Recreating music performance environments for live music ensembles using architectural modeling and real-time audio processing. Veronika Pavlova Vorel (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, pavlov@rpi.edu) and Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

This paper presents a real-time auralization system for loudspeaker-based immersive environments. The goal is to recreate music performance environments rather than just recreating the sound of a specific hall. The system uses a combination of close-mic-driven auralization techniques simulating both early reflections and late reverberation, as well as reverberation augmentation techniques using a microphone position outside the critical distance. The latter is important to provide a captivating experience for musicians and the audience alike, in particular, to provide the feel of presence for the audience. Authentic sound fields are generated from floor plans and CAD models of the simulated venues using virtual microphone control as a real-time auralization engine (Braasch *et al*., 2008). The acoustic experience was recreated in Rensselaer's Collaborative Research Augmented Immersive Virtual Environment (CRAIVE-Lab). The CRAIVE-Lab uses 134 full-range loudspeakers (JBL 308, 128 speakers along the lab's perimeter at ear height, plus six ceiling speakers). Eight short-throw front projectors reproduce congruent immersive 360deg images on a screen that encloses the listening area up to a height of 4.3 m. The screen is microperforated for improved acoustic performance. The listeners can walk through these live recreations over an extended user area (120 m^2) and experience the recreated performance environments.

1:45

4pAA3. Enhancing and streamlining broadcast audio with digital microphone arrays. Yuri Lysoivanov (Shure Inc., 5800 W Touhy Ave., Niles, IL 60714, lysoivy@shure.com) and Brent Shumard (Shure Inc., Niles, IL)

Covering a wide acoustic space while isolating key elements has been an ongoing challenge in broadcast, often involving strategic placements of microphones to try to best capture detailed audio while minimizing unwanted ambient noise. A new approach to audio capture systems incorporating beamforming, integrated auto-mixing and additional signal processing techniques can augment current analog workflow and increase isolation for capturing audio sources using steerable, highly directive capture in a variety of environments.

2:05

4pAA4. Suites for audio research, engineering, and showcasing. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com), Zachery O. L'Italien (McKay Conant Hoover Inc., Westlake Village, CA), and Chris Hansen (Harman, Northridge, CA)

A world-renowned manufacturer of high-end loudspeakers and audio products sought to update and improve their facilities via the design and construction of demonstration/listening rooms and new labs within their existing building shell. Major spaces include rooms for studio listening, 3-D listening, sound-reinforcement, and ceiling loudspeakers, among others. The HVAC system went through some major design changes, always requiring very quiet operation. Of course, interior treatments and room sizing were especially important. This presentation will review highlights, with some emphasis on the studio listening room, which has generated enthusiastic praise from the end users.

2:25

4pAA5. Adaptive 3-D reverb processing in immersive audio systems—Clear imaging and directional envelopment. Dai Hashimoto (Spatial Audio Group, Yamaha Corp., 10-1, Nakazawa-cho, Naka-ku, Hamamatsu, Shizuoka 4308650, Japan), Hiromu Ogi (Spatial Audio Group, Yamaha Corp., Shizuoka, Japan), and Hideo Miyazaki (Spatial Audio Group, Yamaha Corp., Hamamatsu, Japan)

Spatial control of reverberation is essential for delivering a natural or augmented sound experience in immersive audio systems. This presentation introduces two core technologies integrated into the updated object-based audio platform, which incorporates a 3-D reverberation effect. The first is an algorithm of spatial reverberation that measures sound intensity in real venues and estimates virtual source distributions to optimize reverberation processing based on sound object positions and loudspeaker layout. The second is a control method that applies spatial gain and delay weighting to reverberant signals, creating three-dimensional gradients to manipulate attributes, such as reverberation directionality and perceived source width. These techniques are designed to operate across diverse loudspeaker configurations and architectural scales. An evaluation framework combining physical acoustic metrics and subjective listening tests will also be outlined, demonstrating how these approaches influence spatial impression and listener experience.

2:45–3:00 Break

3:00

4pAA6. Electronic enhancement for control of loudness in a concert hall. David Kahn (Acoust. Distinctions, 110 8th St., Troy, New York 12180, Stamford, CT 06901, dkahn@ad-ny.com) and Arthur W. van der Harten (Acoust. Distinctions, Stamford, CT)

We present the design concept for a new concert hall at the University of Florida tailored to host large and powerful ensembles (up to 200 pieces), while accommodating only a modest audience size of 500 seats. The challenge lies in creating an acoustical environment that avoids excessive loudness without sacrificing reverberation, especially given the limited physical volume available. The hall area

that results from the occupancy is such that the room would require an excessively high roof deck (90 ft above the stage platform) in order to incorporate the volume needed for adequate control of loudness (around 650000 cubic feet). Constructing such a tall room to compensate for the limited floor area is often not feasible due to the high costs involved and concerns with the visual aesthetics of such a tall building. In this presentation, we describe our approach to the design of the hall to accommodate the loudest ensembles with the enhancement system, as well as more modest ensembles, which will still have the option to use the natural acoustics of the room. Our approach integrates (passive) adjustable acoustic absorption with an (active) electronic acoustic enhancement system to provide good playing conditions for all ensembles.

3:20

4pAA7. Interaural differences in rooms and the 1000-Hz puzzle. William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu)

Interaural time differences (ITDs) and interaural level differences (ILDs) are the basis for human sound localization. These differences are greatly affected by a room environment, often in the same way, independent of reverberation time. Pure tone experiments with headphones show that ITDs are highly effective at 1000 Hz, though they rapidly become useless for slightly higher frequencies. But in any room environment, ITDs become worse than useless at 1000 Hz. Headphone experiments show that listeners are sensitive to ILDs down to quite low frequencies, but in the free field, ILDs become useful only if the source is close to the head. By contrast, in a room environment, low-frequency ILDs are often important even for large source distances. Experiments in very different rooms agree that the linear crossover between ILD and ITD dominance is 550 ($\pm$100) Hz. New, alternative calculations lead to similar crossovers. The frequency of 1000 Hz is especially strange. In the free field, it produces an image at the back of the listener, unlike the image for all other frequencies below 8000 Hz. Room measurements show that binaural coherence (optimum correlation) has a minimum exactly at 1000 Hz.

Contributed Papers

3:40

4pAA8. Deaf-centered architectural acoustics via toolpath-tuned robotic FDM: Tunable diffusion from printed morphology. Po-Chun Chou (Univ. of Michigan, 2213 CRAM Pl., APT 03, Ann Arbor, MI 48105, pochunc@umich.edu) and Qingru Yang (Univ. of Michigan, Ann Arbor, MI)

This study presents a tunable architectural acoustic wall fabricated by large-format robotic fused deposition modeling. Using PETG with a 2-mm nozzle, we vary layer height (0.6–1.5 mm), extrusion rate, surface subdivision density, and recession depth (35–400 mm) across a field of 2480 recessed modules. Macro-scale indentation gradients steer mid-frequency reflection paths, while micro-scale toolpath texture increases high-frequency scattering. A height-dependent tuning concentrates scattering near the typical vocal–ear level ($\sim$1.55 m) to address conversational bands. Finite-element room-acoustic simulations in a $5 \times 5 \times 2.4$ m3 virtual room evaluate impulse-response structure and decay behavior, including suppression of specular peaks, improved temporal uniformity of energy decay, and reduced flutter-echo indicators. Results show that higher subdivision and finer layers redistribute early reflections into temporally dispersed, spatially diffuse fields, mitigating mid-frequency spectral coloration from late-stage reflections without relying on added absorption. These changes reduce reverberant smearing and listening effort, supporting deaf and hard-of-hearing users in segregating speech from background noise and contributing to more communication-friendly conversational environments.

3:55

4pAA9. The implementation of dipole loudspeakers into an active acoustics system. Mihai-Vlad Baran (Sound Recording, McGill, 1199 Rue Bishops, 807, Montreal, QC H3G 0A7, Canada, vlad.mihai.baran@gmail.com), Kathleen Ying-Ying Zhang (Music Res., McGill Univ., Montreal, QC, Canada), Aybar Aydin (Sound Recording Area, McGill Univ., Montreal, QC, Canada), Richard King, and Wieslaw Woszczyk (Music Res., McGill Univ., Montreal, QC, Canada)

This paper details the application of dipole speakers as part of a whole-room virtual acoustics system. Installed in the McGill Immersive Media Lab (IMLAB) in 2022, the Virtual Acoustics Technology (VAT) system has so far consisted of up to 6 microphones and 15 omnidirectional speakers, which radiate toward the lab's walls and reflect back into the space of the room, accumulating reflections in the listening environment. In the system's largest physical update thus far, 16 planar loudspeakers with bi-directional responses have been installed in order to actively diffuse sound along the room's perimeter and extend the aural environment beyond the space of the room. We document the process of this installation and present use cases and applications in musical recording scenarios.

4:10

4pAA10. Acoustically modeling the transition from unamplified to sound-reinforced Jazz performance in the 1930s. Casey Hanauer (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hanauc@rpi.edu) and Jonas Braasch (School of Architecture/Future of Computing Inst., Rensselaer Polytechnic Inst., Troy, NY)

This study investigates how the introduction of early sound-reinforcement systems altered the acoustic conditions of a major American performance venue during the emergence of amplified jazz in the 1930s. Radio City Music Hall serves as a historical case study due to its architectural continuity and its successive upgrades from fully acoustic performance conditions to public-address systems. Archival newspaper accounts and period documentation were reviewed to characterize audience and critical perceptions of the hall's acoustic environment across technological eras. A recently established dataset of anechoic jazz recordings is used as a source for architectural auralizations. A geometric model of the venue was developed and evaluated using pachyderm to quantify changes in room-acoustic parameters, including clarity, early-to-late energy ratios, spatial impression, and predicted listener envelopment under both unamplified and reinforced conditions. Preliminary results indicate that early PA systems substantially modified the spatial distribution of energy and reduced reliance on the hall's natural acoustic support, while the preserved architectural features maintained a consistent visual and cultural identity. These findings highlight the interplay between technological reinforcement and architectural acoustics in historically significant venues. [Work supported by American Association of University Women Selected Professions Fellowship.]

Session 4pBAa

Biomedical Acoustics: Shocking Considerations in Therapeutic Ultrasound II

Kenneth B. Bader, Cochair
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Tatiana Khokhlova, Cochair
Dept. of Medicine, Univ. of Washington, 1013 NE 40th St., CIMU, Portage Bay Bldg., Seattle, WA 98125

Jonathan R. Sukovich, Cochair
Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Ann Arbor, MI 48109

Contributed Papers

1:00

4pBAa1. Characterization of pre-focal cavitation during histotripsy and its influence on focal cavitation dynamics. Leah R. Holmes (Biomedical Eng., Virginia Tech, 325 Stanger St., Blacksburg, VA 24061, rhleah@vt.edu), Adam D. Maxwell, and Eli Vlaisavljevich (Biomedical Eng., Virginia Tech, Blacksburg, VA)

Histotripsy is a focused ultrasound therapy that employs controlled cavitation to mechanically liquefy tissue. Treatment safety and efficacy depend on confining cavitation activity to the focal region; however, cavitation may occur pre-focally in the coupling medium under certain acoustic and environmental conditions, including shallow treatment depths, medium inhomogeneities, and irregular tissue–water interfaces. This study investigates the acoustic pressure threshold and characteristics of pre-focal cavitation at tissue-phantom/water interfaces and examines its influence on focal cavitation dynamics. Experiments were conducted in tissue-mimicking phantom gels at treatment depths of 3.5, 5, and 15 mm in deionized water with low (10%) and high (80%) dissolved oxygen content. Thirty-one peak negative pressures from 10 to 72 MPa were applied with a 700-kHz focused transducer using pulse repetition frequencies of 0.5 and 500 Hz. Cavitation activity was assessed using passive cavitation detection, high-speed optical imaging, and B-mode ultrasound to identify cavitation onset and characterize spatiotemporal dynamics. Initial observations indicate the pressure threshold for pre-focal cavitation is lower than the corresponding focal cavitation threshold. Ongoing experiments aim to further quantify threshold behavior and evaluate how pre-focal cavitation alters focal ablation rate and spatial precision. These findings will inform monitoring and mitigation strategies to improve histotripsy treatments.

1:15

4pBAa2. Ultrasound-guided delivery of fatty acid nitroalkene (CP8)-loaded microbubbles for treating triple-negative breast cancer. Muhammad Wahab Amjad (Medicine, Univ. of Pittsburgh, 3550 Terrace St., 959 Scaife Hall, Pittsburgh, PA 15213, MUA56@pitt.edu), Soheb Anwar Mohammed, Xucai Chen, Marco Fazzari, Carola Neumann, Bruce Freeman, and John J. Pacella (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Triple-negative breast cancer (TNBC) is an aggressive subtype lacking estrogen, progesterone, and HER2 receptors, limiting effective targeted treatment options. Current therapy relies largely on systemic chemotherapy, which is associated with significant toxicity and modest clinical benefit. Nitrooleic acid (CP8), a nitrated fatty acid, exhibits strong antiproliferative and antitumor activity in aggressive cancers, including TNBC. To enhance targeted delivery, CP8 was encapsulated into lipid-based microbubbles (MBs) and delivered using ultrasound-targeted microbubble cavitation (UTMC). CP8-loaded MBs were formulated and characterized for stability and drug loading. *In vitro* studies in MDA-MB-231 TNBC cells demonstrated significantly greater antiproliferative efficacy with UTMC + CP8-loaded MBs compared to free CP8. In a TNBC xenograft mouse model, UTMC-mediated delivery produced a marked reduction in tumor growth relative to untreated controls and free CP8. Quantitative LC-MS analysis confirmed a 35-fold increase in intratumoral CP8 concentration compared to adjacent skeletal muscle. These results demonstrate that UTMC-enabled CP8 microbubble delivery enhances tumor-specific drug accumulation and therapeutic efficacy while minimizing off-target exposure, supporting its potential as a targeted TNBC treatment strategy.

1:30

4pBAa3. Ultrasound-targeted microbubble cavitation for the treatment of myocardial microvascular obstruction: A porcine study. Soheb Anwar Mohammed (Medicine, Univ. of Pittsburgh, Dept. of Medicine, 3350 Scaife Hall, Pittsburgh, PA 15213, som57@pitt.edu), Muhammad Wahab Amjad, Xucai Chen, Flordeliza S. Villanueva, and John J. Pacella (Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

Microvascular obstruction (MVO) following a stenting procedure is an underlying problem associated with worse clinical outcomes. We are developing a novel treatment, sonoreperfusion (SRP), to restore microvascular patency. Ultrasound-targeted microbubble cavitation (UTMC) of Definity® microbubbles (MBs) disrupts obstructive microthrombi in the coronary microvessels. We used a Wisconsin Miniature Swine™ to test the efficacy of Definity® MBs on MVO. We hypothesized that clinically used ultrasound contrast MBs could be theranostic agents for both imaging and treating obstructed microvessels. A stable MVO was created, as confirmed by no angiographic flow and contrast echocardiography in the mid-LAD. Following 10 min of stable MVO, Definity® MBs were infused for contrast-enhanced ultrasound (CEUS) imaging, followed by 20 min of UTMC treatment or no treatment. CEUS cine loops with burst replenishment, which were obtained at baseline (BL), MVO, and 48 h post-MVO/UTMC, were analyzed using MATLAB. ®Angiography confirmed LAD patency during BL and at 48 h post-UTMC. UTMC treatment significantly increased blood volume in LAD-perfused regions compared with the untreated control. Similarly, with the UTMC treatment, we have found improved left ventricular function. UTMC improved myocardial structural and functional parameters. Studies to explore the underlying physical and molecular mechanisms are underway.

1:45

4pBAa4. Non-viral targeted inducible nitric oxide synthase (iNOS) delivery to the brain for modulating sterile inflammation induced by ultrasound-mediated BBB opening. Aref Ghaderi (Radiology, Children's Hospital of Philadelphia, 3615 Civic Ctr. Blvd., Philadelphia, PA 19104, ghaderim@chop.edu), Arindam Pal, Yash J. Narsian, Leonidas Bleris (Bioengineering, The Univ. of Texas at Dallas, Richardson, TX), Zhenpeng Qin (Mech. Eng., The Univ. of Texas at Dallas, Richardson, TX), and Shashank Sirsi (Bioengineering, UT Dallas, Richardson, TX)

Focused ultrasound (FUS) combined with circulating microbubbles (MBs) enables noninvasive, spatially targeted opening of the blood–brain barrier (BBB) for drug and gene delivery. However, accumulating evidence indicates that even mild BBB opening parameters can induce sterile neuroinflammation, characterized by microglial activation and astrocytosis, which could pose a significant barrier to clinical translation. In this study, we explored whether localized NO production in the brain, derived by inducible nitric oxide synthase (iNOS) transfection, could modulate inflammatory responses associated with FUS-mediated BBB opening. iNOS plasmid DNA was delivered to the mouse hippocampus using cationic MBs and low-pressure FUS. BBB opening alone elicited a robust sterile inflammatory response without overt tissue damage. High-dose iNOS transfection further exacerbated inflammation and induced apoptosis, indicating dose-dependent neurotoxicity. In contrast, low-dose iNOS delivery resulted in minimal to no microglial accumulation, reduced astrocyte activation, and no detectable apoptosis or structural injury, despite successful opening of the BBB. These findings suggest that modest, localized NO production can suppress ultrasound-induced sterile inflammation in a dose-dependent manner. This work identifies a previously unrecognized anti-inflammatory role for low-level iNOS expression and suggests a straightforward strategy to improve the safety and therapeutic window of FUS-mediated BBB opening for neurotherapeutic applications.

2:00

4pBAa5. Acoustic emissions-guided focused ultrasound for liquid biopsy in infiltrative gliomas: A multicentre, open-label, phase 1/2 trial. Pavlos Anastasiadis (Neurosurgery, Univ. of Maryland School of Medicine, 670 W Baltimore St., Baltimore, MD 21201, panast@som.umaryland.edu), Ahmad Ozair, Chixiang Chen, and Graeme F. Woodworth (Neurosurgery, Univ. of Maryland School of Medicine, Baltimore, MD)

To characterize acoustic emissions (AE)-based closed-loop control and dosing for transcranial microbubble-enhanced focused ultrasound (MB-FUS) blood-brain barrier opening (BBBO) performed in the infiltrative margins of high-grade gliomas. We analyzed MB-FUS BBBO treatments performed using a 1024-element hemispheric phased-array system (Exablate Neuro Model 4000 Type II). AE signals were captured via integrated passive cavitation detectors. Sonications used a 1%–10% duty cycle, 5 ms pulse length, and 1 Hz pulse repetition frequency per subsonication. BBBO was quantified as a percentage of new T1-weighted contrast enhancement (new T1c) within each target on post-treatment MRIs. Sono-liquid biopsy profiling was performed with paired blood draws obtained immediately before and within 1 h after MB-FUS, enabling analysis of post-to-pre changes in total plasma cfDNA and fragmentomic features. cfDNA levels were analyzed in 23 participants, and cfDNA fragmentomics in 19 patients. Post-to-pre ratios of cfDNA and fragment length ratio (FLR) were significantly associated with both progression-free and overall survival (cfDNA: HR 0.45 and 0.71; FLR: HR 1.62 and 3.40; all $p < 0.01$). Longitudinally, increasing post-to-pre biomarker trajectories were associated with early progression, whereas decreasing trajectories were associated with long-term survival; trajectory concordance was 76% for cfDNA (PFS/OS) and 90% for FLR with PFS.

THURSDAY AFTERNOON, 14 MAY 2026 FRANKLIN HALL 9, 2:30 P.M. TO 4:35 P.M.

Session 4pBAb

Biomedical Acoustics: Wearable Ultrasound

Katherine G. Brown, Chair

Bioengineering, Univ. of Texas at Dallas, 800 W. Campbell Rd., Richardson, TX 75080

4p THU. PM

Invited Papers

2:30

4pBAb1. Piezoelectric micromachined ultrasound transducers for next-generation wearable ultrasound. Yao Yu (Texas Instruments, 4555 Great America Pkwy, Santa Clara, CA 95054, y-yu5@ti.com), Yande Peng (Texas Instruments, Santa Clara, CA), bichoy Bahr (Texas Instruments, Dallas, TX), and Yogesh Ramadass (Texas Instruments, Santa Clara, CA)

Piezoelectric micromachined ultrasound transducer (PMUT) is a promising technology for compact, low-cost, and high-performance wearable ultrasound systems. Fabricated using silicon micromachining, PMUTs operate in a flexible, "drum-like" mode offering advantages over conventional bulk transducers—including smaller size, lower cost, 2-D array compatibility, and broader bandwidth. This talk details recent advancements in PMUT development at Texas Instruments aimed at realizing next-generation wearable ultrasound applications. Utilizing high-piezoelectric coefficient materials, our PMUT devices demonstrate comparable pulse-echo sensitivity to traditional bulk transducers. The thin-film nature of PMUTs enables large bandwidth operation without acoustic matching layers (due to the "drum-like" motion), exceeding bulk transducers with matching layers. Furthermore, this performance is achieved with tiny individual

PMUT elements connected in parallel, allowing for the integration of thousands of addressable elements within a die size of just a few millimeters and facilitating diverse array configurations. These innovations will enable PMUT technology to significantly advance the capabilities of wearable ultrasound imaging.

2:50

4pBAb2. Towards a wearable ultrasound bladder monitoring system. Katherine G. Brown (Bioengineering, Univ. of Texas at Dallas, 800 W. Campbell Rd., Richardson, TX 75080, katherine.brown@utdallas.edu), Rouzbeh Imenabadi (Bioengineering, Univ. of Texas at Dallas, Richardson, TX), Gregory Thoreson (Dept. of Surgery, Baylor Univ. Medical Ctr., Dallas, TX), and Dinesh Bhatia (Elec. and Comput. Eng., Univ. of Texas at Dallas, Richardson, TX)

Ultrasound is used to monitor the volume of the urinary bladder in clinical settings. However, conventional point-of-care ultrasound (POCUS) units require a skilled operator and repeated visits for patient monitoring. Wearable ultrasound for bladder monitoring is emerging but limited in its ability to give volume readings with high accuracy. Challenges to achieving a highly accurate wearable ultrasound monitor for the clinical setting include the reduction of overall size and weight, the reduction of power, and management of an inherently high raw data rate present in B-mode ultrasound imaging. The objective of this work is to achieve high volume accuracy in an autonomous ultrasound imaging device. This work presents a 128-channel prototype ultrasound system (50 x 90 mm2) for urinary bladder monitoring which achieves high accuracy, and high efficiency. It implements a sparse receiver and uses deep learning to reconstruct the missing received data lines to achieve low power and a small footprint. The raw data is streamed wirelessly to a smart device for image reconstruction, segmentation, and volume measurement using deep networks. The wearable achieves an accuracy of bladder volume measurement similar to POCUS but with continuous autonomous monitoring after initial placement on the patient. The size, accuracy, and efficiency of this autonomous ultrasound imaging device are a promising direction for a fully wearable solution.

3:10

4pBAb3. Wearable ultrasound technology for deep tissue monitoring and beyond. Muyang Lin (Dept. of Anesthesiology, Perioperative and Pain Medicine, Stanford Univ., 8720 Costa Verde Blvd. APT1408, San Diego, CA 92122, m1lin@stanford.edu) and Sheng Xu (Nanoengineering, UC San Diego, La Jolla, CA)

Medical ultrasound is a powerful diagnostic and evaluation tool in hospitals and clinics due to its safety and versatility. Traditional ultrasound suffers from operator dependency and does not allow long-term usage. Recent advances in wearable ultrasound technologies have demonstrated the potential for hands-free data acquisition, but technical barriers remain as these probes require wire connections, can lose track of moving targets, and create data-interpretation challenges. Here we report a fully integrated autonomous wearable ultrasonic-system-on-patch (USoP). A miniaturized flexible control circuit is designed to interface with an ultrasound transducer array for signal pre-conditioning and wireless data communication. Machine learning is used to track moving tissue targets and assist the data interpretation. We demonstrate that the USoP allows continuous tracking of physiological signals from tissues as deep as 164 mm. On mobile subjects, the USoP can continuously monitor physiological signals, including central blood pressure, heart rate, and cardiac output, for as long as 12 h. This result enables continuous autonomous surveillance of deep tissue signals toward the internet-of-medical-things.

3:30

4pBAb4. Ultrasound you can wear: Power-aware architectures and edge intelligence for continuous physiological sensing and human–machine interaction (invited). Andrea Cossettini (ETH Zurich, Andreasstrasse 5, Zurich, ZH 8050, Switzerland, cosandre@iis.ee.ethz.ch)

Ultrasound is a powerful modality for non-invasive sensing of deep physiological structures, yet its translation from handheld and cart-based systems to truly wearable devices introduces fundamental constraints in power consumption, data throughput, raw-data access, and system integration. In this talk, I will review our group's work on wearable, ultra-low-power ultrasound architectures, showing how power-aware system and acquisition strategies, combined with on-device signal processing, enable continuous, multi-day wearable monitoring while preserving access to raw acoustic data. The talk first examines WULPUS, a truly wearable ultra-low-power A-mode ultrasound platform consuming less than 30 mW, and summarizes its use for multi-day monitoring of muscle dynamics. WULPUS-based armbands are shown to enable robust hand-wrist kinematic tracking, while lightweight on-device signal processing preserves physiologically relevant acoustic features and drastically reduces wireless bandwidth requirements. The discussion then turns to TinyProbe, a 32-channel wearable ultrasound platform integrating transmit beamforming, 30 MSPS sampling, and Wi-Fi-based raw-data streaming, which enables multichannel imaging and high-frame-rate arterial monitoring. Together, these systems highlight key system-level design choices and tradeoffs, and outline open challenges and opportunities for continuous, body-worn ultrasound sensing and human–machine interaction.

Contributed Papers

3:50

4pBAb5. Automated muscle fascia tracking with wearable ultrasound for real-time muscle coordination assessment. Erica L. King (Bioengineering, George Mason Univ., 4400 Univ. Dr., Fairfax, VA 22030, eking20@gmu.edu), Connor Delaney, Morgan Lamarre, Amad Qureshi, Siddhartha Sikdar, Qi Wei, and Parag V. Chitnis (Bioengineering, George Mason Univ., Fairfax, VA)

Musculoskeletal ultrasound (MSK-US) provides real-time muscle imaging, but bulky, single-site probes limit its use in dynamic settings. Wearable ultrasound (WUS) enables dynamic imaging, creating opportunities for real-time monitoring in rehabilitation. This study evaluated five fascia-tracking algorithms, Maximum Pixel Intensity (MPI), Muscle Boundary Tracking Algorithm (MBTA), Principal Component Analysis (PCA), Composite-Factorization PCA (CF-PCA), and U-Net segmentation, against expert manual traces. Data included 572 M-mode images collected during quadriceps activations and squats using a multi-site WUS system with transducers on the vastus lateralis (VL) and vastus medialis oblique (VMO). U-Net segmentation achieved the lowest mean absolute error, while PCA-based methods provided the highest correlation with expert traces. Based on these

findings, the same WUS system collected data from 20 participants performing bilateral countermovement jumps (CMJs). M-mode images were processed using CF-PCA, and correlation analysis was used to assess muscle coordination in healthy individuals ($n = 11$) and those with a prior knee injury ($n = 9$). CF-PCA produced time traces that tracked tissue movement and correlated with force–time curves. Injured participants exhibited reduced VL–VMO correlation ($p = 0.039$) and differing activation times ($p = 0.013$), indicating persistent coordination deficits. WUS with automated processing provides actionable metrics for rehabilitation feedback and supports future real-time implementation. [Work sponsored by the Government under Other Transactions Number W81XWH-15-9-0001 and HT9425-24-9-0019.]

4:05

4pBAb6. Simultaneous real-time multichannel M-mode and Doppler imaging in a miniature wearable ultrasound device. Abhishek Aher (Bioengineering, George Mason Univ., 4400 Univ. Dr., Fairfax, VA 22030, aaher@gmu.edu), Ahmed Bashatah, Parag V. Chitnis, and Siddhartha Sikdar (Bioengineering, George Mason Univ., Fairfax, VA)

Wearable ultrasound systems face challenges for simultaneous structural and motion-sensitive imaging due to constraints on size, power, and data throughput. This work presents a miniature wearable ultrasound imaging system integrating simultaneous M-mode and Doppler measurements using time-delay spectrometry (TDS). TDS employs a low-voltage, wideband chirp to establish a deterministic relationship between reflector depth and frequency enabling separation of overlapping echoes in both time and frequency domains, enabling real-time estimation of depth-resolved tissue displacement and velocity from a single transmission sequence. The system architecture combines a single-element transducer, low-power (5V) excitation, and synchronized signal processing to extract M-mode and Doppler-derived tissue velocity profiles at multiple sites. Experiments using string phantoms and musculoskeletal motion demonstrated stable M-mode imaging and repeatable depth-resolved tissue velocity estimates. String phantom experiments demonstrated a strong linear relationship between Doppler frequency shift and true string velocity (Pearson $r = 0.999$, $p < 10^{-13}$). Our wearable battery-powered system supports simultaneous M-mode and tissue velocity imaging at four sites at a frame rate of 400 Hz with an average current draw of 200mA. This work represents the first demonstration of a wearable ultrasound system enabling real-time tissue Doppler and M-mode measurements.

4:20

4pBAb7. Printable and flexible ultrasound transducers using laser-induced porous graphene and piezopolymer. Ehsan Ansari (Bioengineering, George Mason Univ., 3051 Hickory Grove Court, Fairfax, VA 22031, eansari@gmu.edu), Clayton A. Baker, Dulcce A. Valenzuela, Ahmed Bashatah, Spencer Hagen (Bioengineering, George Mason Univ., Fairfax, VA), Pilgyu Kang (Mech. Eng., George Mason Univ., Fairfax, VA), Shirin Movaghgharnezhad, and Parag V. Chitnis (Bioengineering, George Mason Univ., Fairfax, VA)

Ultrasound technology is crucial in diagnostic imaging, making it widely used in medical applications. However, traditional ultrasound transducers face limitations in flexibility and ease of fabrication, leading to the exploration of thin-film and flexible piezoelectric materials. Here, we present a novel approach that combines laser graphitization with 3D printing to integrate flexible laser-induced porous graphene (LIG) with poly(vinylidene fluoride-trifluoroethylene) (PVDF-TrFE), resulting in the development of flexible LIG/PVDF-TrFE ultrasound patches. The thickness of PVDF-TrFE is adjusted to tune the central frequency of the ultrasound transducer, allowing customization within a range of 10–28 MHz. The LIG-based ultrasound transducer demonstrates a high signal amplitude of 6.72 V and a signal-to-noise ratio (SNR) of 433, along with a 6 dB bandwidth of 8.86 MHz (37%). The LIG-based transducer exhibits higher acoustic performance compared to the smooth silver-based transducer. A high-quality two-dimensional ultrasound image, including a B-mode image of a cyst phantom, demonstrates the transducer's imaging capabilities. The patterning of LIG-based electrodes facilitates the desired sensor configuration, demonstrating the suitability of our novel technique for producing flexible transducer arrays without dicing and cutting. The materials cost of our LIG/PVDF-TrFE transducer is under $5 per unit, making it a low-cost solution for ultrasound patches.

Session 4pBAc

Biomedical Acoustics: General Topics in Biomedical Acoustics: Therapy, Microbubbles, and Beyond

Yun Jing, Chair
Acoust., Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802

Contributed Papers

3:45

4pBAc1. Ultrasound control of cell fate determination of human induced pluripotent stem cell-derived neural progenitors *in vivo* via heat shock promoters. Kama Bell (Graduate Program in Acoust., Penn State, P.O. Box 188, Pine Grove Mills, PA 16868, krb5915@psu.edu), Chad Smies (Biology, Penn State, University Park, PA), Alessandro R. Howells, Tahir Haideri (Biomedical Eng., Penn State, University Park, PA), Janine Kwapis (Biology, Penn State, University Park, PA), Thomas Neuberger (Huck Institutes of the Life Sci., Penn State, University Park, PA), Xiaojun Lian (Biomedical Eng., Penn State, University Park, PA), and Yun Jing (Graduate Program in Acoust., Penn State, State College, PA)

Inducible systems are essential toolsets in regenerative medicine, enabling precise spatiotemporal control of gene expression in response to specific stimuli. An emerging approach utilizes the natural cellular response to heat stress, where heat shock promoters (HSPs) drive endogenous gene expression to negate heat shock. Temperature-dependent exogenous gene expression can therefore be induced by HSPs in genetically modified cells via mild hyperthermia. Focused ultrasound (FUS) offers thermal energy delivery non-invasively with spatiotemporal precision. In this research, a Cre/LoxP recombination system was employed downstream of HSPs in human induced pluripotent stem cell-derived neural progenitors, enabling permanent expression of a gene of interest after a brief heat shock. HSP activation, leakage kinetics, and cell viability after thermal dose were characterized *in vitro* for tunability and translation to an *in vivo*, thermal dose was assessed for up to a total of 1 h at 42°C incubation, or 15 cumulative equivalent minutes at 43° C. *In vitro*, modified cells were stereotaxically injected into immunocompromised mice, and magnetic resonance imaging-guided FUS hyperthermia was used to guide heat shock. This system poses a versatile mechanism for translational controlled gene activation and regulation in modified stem cells for regenerative medicine.

4:00

4pBAc2. Pressure responses of different lipid-shelled microbubbles: A perspective on pressure dependent stability and subharmonic response. Mehmet Yapar (Mech. and Aerosp. Eng., George Washington Univ., Sci. & Eng. Hall, 800 22nd St. NW, Washington, DC 20052, myapar23@gwu.edu), Saikat Halder (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC), John Eisenbrey, Flemming Forsberg (Radiology, Thomas Jefferson Univ., Philadelphia, PA), Wim van Hoeve (Solstice Pharmaceuticals, Enschede, Netherlands), and Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC)

Microbubbles are widely used as ultrasound contrast agents because of their high echogenicity and nonlinear oscillatory behaviors. Under sufficiently strong acoustic excitations, they generate multiple harmonic components, including a subharmonic response. The subharmonic exhibits a strong correlation with ambient pressure, whereas the fundamental and second harmonics show little to no sensitivity. In this study, we investigate the characteristics and pressure-dependent behaviors of several microbubble formulations. These include commercially available polydisperse contrast agents, such as Definity, Sonazoid, and SonoVue, a new monodisperse agent, and polydisperse microbubbles produced in-house containing different gases (SF_6, C_3F_8, and C_4F_{10}). Each microbubble type was characterized by attenuation and size measurements. We developed a new optical microscopy imaging setup to observe the stability of these microbubbles under static overpressure. Finally, we measured the ambient pressure sensitivity of subharmonic responses from each microbubble type. The findings offer new insights into the stability and pressure-dependent behaviors of different classes of microbubbles, which are critical for further development of subharmonic-aided pressure estimation (SHAPE).

4:15

4pBAc3. Interaction mechanics of ultrasound cavitation on biological matrices. Shorup chanda (Biomedical Eng., Univ. of Wisconsin–Madison, Madison, WI), Aarushi Bhargava (Biomedical Eng., Univ. of Wisconsin–Madison, 1550 Eng. Dr., Fitchburg, WI 53706, aarushi.bhargava@wisc.edu), and Claire Nesemann (Biomedical Eng., Univ. of Wisconsin–Madison, Madison, WI)

Collagen and fibrin form fibrous networks that provide structural integrity and an extracellular matrix for cells across different tumor and blood clot environments. These biological matrices possess unique and complex mechanical properties, distinct from most biological tissues. Stable cavitation, bubbles undergoing periodic oscillations under ultrasound, has been extensively studied for various applications, such as drug delivery in soft tissue models. However, there is limited information on the governing physics of the interaction of these bubbles with fibrin and collagen matrices. This understanding is essential for optimizing ultrasound-mediated drug delivery in environments characterized by dense biological matrices. Our ongoing study aims to explore the bubble–matrix interactions and the evolving mechanics of these networks. Using high-speed and confocal imaging combined with atomic force microscopy, we quantify the bubble behavior in collagen and fibrin networks of different porosities and stiffnesses, and correlate it with the network's mechanical response. Results indicate that cyclic forcing from the radial oscillation of bubbles induces damage to the fiber mesh with each cycle. This damage accumulates over time, leading to mesh softening and rupture, with enhanced drug penetration. We further examine the correlation of this induced damage with the fiber network structure and insonation parameters.

4:30

4pBAc4. Comparing focused ultrasound to dry needling in two *ex vivo* tendinopathy models. Surabhi J. Menon (Graduate Program in Acoust., Penn State Univ., Penn State Univ., University Park, PA 16802, sjm7872@psu.edu), Grace M. Wood (Graduate Program in Acoust., Penn State, University Park, PA), Meghan E. Vidt (Biomedical Eng., Penn State Univ., State College, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Tendinopathy, or tendon injury, is a musculoskeletal condition characterized by collagen fiber degeneration. Dry needling (DN) is a standard conservative treatment for tendinopathy; however, clinical success rates are mixed. Prior work suggests that focused ultrasound (FUS) disrupts tendon fibers and may be used as an alternative therapy to promote healing. Here, we compare the mechanics and histology after DN and FUS application in two *ex vivo* tendinopathy models: collagenase-injection or mechanical

overload. Injured bovine tendons (n= 3/group) were treated with DN (50 fenestrations/2 min) or FUS [2.45 MHz; 10-ms dual-amplitude bursts (5 ms at p+ =52MPa/p− =18MPa; followed by 5 ms at one-third amplitude) at 1 Hz]. Ultimate tensile stress (UTS), elastic modulus (E), and stiffness (K) were assessed. After mechanical overload, DN-treated tendons measured UTS = 26.8 ± 4.6-MPa, E = 117.9 ± 16.6MPa, and K = 100.83 ± 81.86N/mm, whereas FUS-treated tendons measured 21.6 ± 13.74MPa, 69.3 ± 12.4MPa, and 49.5 ± 17.9N/mm, respectively. For collagenase injection models, DN-treated tendons measured UTS = 30.9 ± 2.9MPa, E = 114.8 ± 10.7MPa, and K = 42.8 ± 7.2N/mm, whereas FUS-treated tendons measured 17.6 ± 4.2MPa, 61.1 ± 26.1MPa, and 67.6 ± 12.3N/mm, respectively. Mann–Whitney U tests yielded no difference between treatment groups for the mechanical outcomes, suggesting little difference following FUS or DN across models. In future work, sample sizes for the treatment groups will be increased to draw robust conclusions. [Work supported by NIH-R01EB032860.]

4:45

4pBAc5. Sensitivity of the shell parameter estimation method for lipid-coated microbubbles. Saikat Halder (Mech. and Aerosp. Eng., George Washington Univ., 800 22nd St. NW Ste. 3000, Sci. and Eng. Hall, Washington, DC 20052, saikat.halder@gwmail.gwu.edu), Mehmet Yapar, and Kausik Sarkar (Mech. and Aerosp. Eng, George Washington Univ., Washington, DC)

Stability and acoustic responses of microbubbles depend on their encapsulating lipid shells. The shell is characterized by different models involving interfacial elasticities and viscosities, determined using microbubble size distribution and attenuation measurements of polydisperse microbubbles. We have found that the results are much sensitive to the size distribution than to the small changes in the attenuation. Here, we further investigate the robustness and sensitivity of the estimation method based on the exponential elasticity model (EEM) by manipulating the size distribution. We varied the bin width (0.01–1 μm), the size distribution fit (Normal/Gaussian, Lognormal, Weibull, and Cumulative), the microbubble boundary (varying by half a pixel), and the total concentration (varying by 50%). This study demonstrates the necessity of accurate size distribution measurements for reliable estimation of shell parameters.

5:00

4pBAc6. Ultrasound-targeted microbubble cavitation induces piezo1-mediated Ca2+ influx and transient gene expression changes. Anurag N. Paranjape (Medicine/Cardiology, Univ. of Pittsburgh, 3550 Terrace St., Pittsburgh, PA 15213, anuragnparanjape@gmail.com), Shuai Yuan, Adam C. Straub (Medicine/Dept. of Pharmacology and Chemical Biology, Univ. of Pittsburgh, Pittsburgh, PA), Xucai Chen, and Flordeliza S. Villanueva (Medicine/Cardiology, Univ. of Pittsburgh, Pittsburgh, PA)

Endothelial barriers tightly regulate vascular permeability, reducing delivery of cell-impermeant therapeutics to diseased tissues. Ultrasound-targeted microbubble cavitation (**UTMC**) offers a non-invasive way to temporarily loosen this barrier, but the endothelial signaling events that drive this response are not well understood. Using human coronary artery endothelial cells, we show that UTMC (f = 1 MHz; PNP = 250 kPa; pulse length = 10 μs; pulse interval = 10 ms; treatment duration = 10 s) rapidly activates a mechanotransduction pathway initiated by piezo1-mediated Ca^{2+} influx, subsequently activating endothelial nitric oxide synthase (**eNOS**), leading to increased nitric oxide (**NO**) production and widespread protein S-nitrosylation. The S-nitrosylation includes reversible modification of VE-cadherin, a core component of endothelial adherens junctions that link adjacent endothelial cells. Together, these changes cause inter-endothelial gaps and a brief increase in paracellular permeability without permanently compromising endothelial integrity. Importantly, junctional structure and gene expression changes return to baseline within 24 h, as shown by bulk RNA sequencing, indicating barrier recovery. Overall, our results describe a piezo1-Ca^{2+}-eNOS-NO signaling axis involved in UTMC-induced endothelial barrier remodeling and define molecular events associated with transient increases in vascular permeability. These data yield insights into bioeffects of UTMC which are important to implementing clinical translation.

5:15

4pBAc7. Egg white-based blood mimicking fluid for therapeutic ultrasound device evaluation. Yunbo Liu (FDA, 10903 New Hampshire Ave., Silver Spring, MD 20993, yunbo.liu@fda.hhs.gov) and Subha Maruvada (U.S. Food and Drug Administration, Silver Spring, MD)

An egg white-based blood mimicking fluid (BMF) was developed and characterized as a blood coagulation surrogate for the acoustical and thermal evaluation of therapeutic ultrasound, especially high-intensity thermal ablation and histotripsy devices. Physical properties, including coagulation temperature, frequency-dependent attenuation, sound speed, thermal conductivity, and thermal diffusivity, were measured as a function of temperature (20–95 °C) The fluid viscosity was quantified at room and body temperature. With the addition of Nylon particles in the solution, the backscattering coefficient of the BMF was quantified before and after a complete thermal coagulation. For a 30s thermal exposure, the egg white-based BMF (3 mm thickness) started to denature at 65 °C and coagulate into an elastic gel at 85 °C. The coagulation temperature can be lowered by adding a small amount of acid solution to the BMF. The temperature-dependent ultrasound attenuation and other physical parameters were found to be similar to the reported values of human blood. These properties make this egg white-based blood mimicking fluid a useful tool for pre-clinical bench testing of therapeutic ultrasound devices.

Session 4pCA

Computational Acoustics: General Topics in Computational Acoustics

Stephanie G. Konarski, Cochair
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Contributed Papers

1:30

4pCA1. Acoustic source models for human vital signals. Joshua Langlois (Graduate Program in Acoust., The Penn State Univ., 556 White Course Dr., State College, PA 16801, jgl5306@psu.edu) and David C. Swanson (Penn State ARL, State College, PA)

The human body produces vibrations that are caused by breathing and the heart beating. Understanding these signals can provide new methods for human health monitoring. Previous research has measured these signals using acoustic cardiograms, however no effort has been made to mathematically model these vibrations as acoustic sources. We present multiple acoustic models, varying in complexity, that can be used to understand the acoustic fields created by these vibrations. The models generate artificial acoustic signatures replicating observations for chest wall motion due to breathing, the chest wall movement due to the heart beating, and the vertical body motion due to blood pumping up and down the aorta. These models can provide simulation data for use in human health monitoring development work, where data are very limited.

1:45

4pCA2. Choice of acoustic model in Nigerian English vowel analysis. Oluwasegun Amoniyan (Linguist, Univ. of Pittsburgh, 2816 Cathedral of Learning, Pittsburgh, PA 15261, oma55@pitt.edu)

Nigerian English (NigE) is a widely spoken English variety, yet its phonetic description has often relied on acoustic models developed for Standard British English (SBE). While these studies have offered insights into the NigE vowel system, they raise methodological concerns about whether model choice influences its description. This study examines whether the acoustic model influences vowel description in NigE. The study analyzes passage readings from 23 NigE speakers sampled from the Kaggle corpus, and their speech was force-aligned using the Montreal Forced Aligner with two pretrained acoustic models: NigE and SBE. 1770 vowels were extracted and compared across models using four measures: vowel annotation and extraction, boundary alignment, vowel duration, and midpoint formant estimation. All analyses were conducted in R 4.5.0. Results show differences in vowel extraction across English models: the NigE model extracted 1050. In contrast, SBE extracted 720 vowels, and boundary alignment mismatches are more frequent for vowels that are not available in the SBE acoustic model. The mismatch contributes to the differences in duration estimates and to the dispersion in the geometry of the F1 and F2 vowel spaces at the midpoint analysis. These findings reveal that acoustic model choice influences phonetic interpretation and support the use of a variety-specific acoustic model for NigE vowels.

2:00

4pCA3. From scalar confidence to structured uncertainty: Modeling child speech intelligibility from acoustic model evidence. Vishal Shrivastava (Commun. Sci. and Disord., Northwestern Univ., 1024 Noyes St., Evanston, IL 60201, shrivastava_vishal@outlook.com) and Marisha Speights (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Accurate assessment of children's speech intelligibility is critical for identifying speech sound disorders. Clinical practice relies on human transcription and perceptual ratings that are costly, inconsistent, and difficult to scale. Existing ASR-based intelligibility metrics offer scalability but face two fundamental issues: (i) training-data mismatch, as adult-centric models produce unstable representations for developmentally variable child speech, and (ii) decoding artifacts, e.g., softmax sharpening, that yield spuriously confident scores and mask acoustic ambiguity. We propose a Speech Intelligibility Probability (SIP) framework that conceptualizes intelligibility as acoustically induced model uncertainty rather than as word error or decoder confidence. Using the Speech Exemplar and Evaluation Database, we extracted token-level posterior trajectories from Whisper Large-v2 under controlled decoding and analyzed them using a multidimensional uncertainty profile capturing distributional dispersion (Tsallis entropy) and temporal instability (coherence), thereby characterizing acoustic-driven uncertainty beyond post-decoding confidence scores tied to committed hypotheses. We compared SIP against Hustad *et al.* (2021) developmental growth curves. It aligned with human perceptual difficulty: typically developing speech clustered near mid-range expectations, while disordered speech concentrated in the bottom decile. This correspondence indicates that internal model uncertainty serves as a robust proxy for speech intelligibility.

2:15

4pCA4. Temporal localization of morphologically-conditioned stop categories in Korean: A computational approach. Chloe D. Kwon (Dept. of Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, dk837@cornell.edu) and Fengyue Lisa Zhao (Dept. of Linguist, Cornell Univ., Ithaca, NY)

The distinction between Korean plain and tense consonants in the word-medial position is subtle because compound tensification allows underlyingly plain stops to surface as tense at compound boundaries. To determine when and where category-related information is present in the acoustic signal, we used a deep neural network-based classification. We trained bidirectional LSTM classifiers on MFCCs from sentences containing words across morphological and phonological contexts, using two types of input manipulations relative to stop onset: (1) varying temporal window size extending leftward or rightward, and (2) shifting the center of the fixed-size temporal window to different positions. Models classified stops as plain or tense. We evaluated performance using classification accuracy as well as confusion rates, calculated as the proportion of plain stops misclassified as tense and vice versa. Results show that category-related information emerges both

earlier and later than predicted by standard theoretical assumptions, with stronger effects later in time, indicating carryover from stop closure into the following vowel. Importantly, confusion rates are higher in compounds than in simplex nouns, suggesting reduced acoustic distinctiveness under compound tensification. These findings demonstrate that this method can localize category-related information in time and capture the effects of phonological processes on morphologically conditioned category realization.

2:30

4pCA5. Time reversal focusing of single-frequency, continuous-wave sound in rooms: The impact of the loudspeaker array layout on the spatial quality of the focusing. Darian B. Comsa (Mathematics, Brigham Young Univ., BYU Dept. of Mathematics, 275 TMCB, Provo, UT 84602, dcomsa@student.byu.edu), Bryce A. Lundstrom, and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) is a two-step process of recording an impulse response and then playing it backwards in the same environment, which focuses the sound in space and can reconstruct a desired temporal signal at a target location. Using TR to focus transient signals has been demonstrated many times in air, liquids, and solids, employing a variety of frequency ranges. However, the application of TR to spatially focus a single-frequency continuous wave (CW) is not yet fully understood. A modal summation approach is used to simulate CW TR focusing in a room. The layout of the loudspeaker sources that creates the CW TR focusing is explored, comparing a line array to a randomly spaced array. The resulting wavefields throughout the room are compared to determine which layout is more likely to produce a higher quality focus in space, i.e., which layout produces less prominent side lobes elsewhere in space and thus provides a more efficient delivery of energy to the target location.

2:45–3:00 Break

3:00

4pCA6. Exploring phi-bits in coupled acoustic waveguides: A classical platform for quantum-inspired information processing. Abrar N. Faiaz (Mech. Eng., Wayne State Univ., DEPT OF Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, ABRAR.FAIAZ@WAYNE.EDU), Akinsanmi S Ige (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ), Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., Detroit, MI), Jake Balla (Univ. of Arizona, Tucson, AZ), M Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI), Pierre A. Deymier (Mater. Sci. and Eng., Univ. of Arizona, Tucson, AZ), Keith Runge, and Joshua A. Levine (Univ. of Arizona, Tucson, AZ)

Classical systems that emulate quantum behaviors provide an effective means of exploring information processing without relying on fragile quantum hardware. In this work, we introduce a validated theoretical framework for phi-bits—nonlinear, phase-dependent modes serving as classical analogues of qubits—in a system consisting of three coupled finite-length acoustic waveguides. Using a discrete mass-spring model, we reproduce the experimentally observed continuous phase evolution of low-order combination modes under dual-frequency excitation. The model integrates nonlinearities at the end springs, damping effects, and boundary conditions, capturing the evolution of phi-bit phases as a function of the driving frequency. For spectrally isolated, high-SNR modes, the phase responses predicted by linear combinations of the driver phases are in qualitative agreement with experimental observations. Additionally, we investigate the role of higher-order nonlinearities, which extend the range of phi-bit behaviors but can disturb the linear relationship unless spring coefficients are recalibrated via nondimensional scaling. This framework provides a set of design criteria for selecting stable phi-bits, tuning sweep protocols, and adapting models to different geometries and materials. Our results demonstrate that classical nonlinear lattices can replicate key aspects of quantum information flow, offering a versatile and tunable platform for phase-encoded logic and quantum-inspired acoustic computation.

3:15

4pCA7. Hilbert space framework for geometric phases and topological dynamics in mechanical lattices. Kazi Tahsin Mahmood (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Ste. #2100, Detroit, MI 48201, hk2799@wayne.edu) and M Arif Hasan (Mech. Eng., Wayne State Univ., Detroit, MI)

Mechanical lattices can exhibit topological behavior rooted in the geometric structure of their vibrational eigenstates. In this work, we introduce a concise Hilbert-space framework that maps elastic lattice dynamics onto the Bloch sphere, enabling a direct geometric interpretation of Berry and Zak phases using experimentally accessible amplitude–phase measurements. Focusing on one-dimensional mass–spring chains, we express intra-cell motion as a superposition of orthogonal eigenmodes, whose complex modal coefficients naturally form a classical two-level state. Within this formulation, inversion symmetry is shown to enforce quantized Zak phases of 0 or π in diatomic lattices, while symmetry breaking leads to a continuous geometric phase despite leaving the band spectrum unchanged. The approach generalizes to triatomic lattices and temporally modulated systems, where coupling between carrier and sideband harmonics gives rise to open-path geometric phases analogous to non-adiabatic Berry evolution. Treating stiffness modulation as a controllable parameter further reveals how modal trajectories trace qubit-like rotations on the Bloch sphere and emulate gate-level operations in an entirely classical platform. Together, this algebraic–geometric perspective provides a unified description of how symmetry, modulation, and topology shape elastic wave dynamics and points toward programmable topological functionalities operating under ambient conditions.

3:30

4pCA8. A quantum approach for vortex shedding induced vibrations. Kenneth Czuprynski (Appl. Res. Lab., The Penn State Univ., 5130 ARL Main Lab, University Park, PA 16802, kdc168@psu.edu) and Andrew S. Wixom (Appl. Res. Lab., The Penn State Univ., State College, PA)

This work explores the development of a quantum algorithm for computing the vibrations due to vortex shedding. The classical "wake oscillator" models for this problem use a nonlinear oscillator (e.g., a van der Pol oscillator) to represent the forcing function of the shed vorticity applied to a linear structural model. The vibrations of the coupled model would then traditionally be simulated using time integration or some other nonlinear solution technique. This work investigates the use of Carleman linearization as a means of embedding the nonlinear behavior into a form suitable for quantum linear solvers. We examine the accuracy of the linearization and discuss the scaling of the system size alongside practical issues related to its integration into available quantum linear solver algorithms.

3:45

4pCA9. Quantum machine learning approaches to acoustic signal classification. Sanjay Majumder (Style tree, 801 Paularino Ave., Apt A201, Costa Mesa, CA 92626, sanjay.majumder1991@gmail.com)

Incorporating recent developments in quantum computing enables new machine learning methods beyond traditional classical methods. This research uses quantum machine learning (QML) to classify acoustic signals—a field that is characterized by large dimensionality in its time frequency representations, and significant non-linear relationships. It examines hybrid classical-quantum architectures where acoustic features (i.e., spectrograms, mel-frequency cepstral coefficients etc.) will be embedded into quantum states through variational quantum circuits. In addition to assessing how well quantum kernels and parameterized quantum models can capture higher-order correlations in the audio data that exceeds what classical models can accomplish, it also compares these models' performance with established classical benchmarks in terms of accuracy, robustness to noise, and computational efficiency, utilizing representative acoustic data sets (e.g., environmental sounds and bio-acoustic signals). The analysis has been performed while taking into account constraints associated with quantum computing platforms (i.e., quantum noise, quantum circuit depth, number of qubits available), and overall, the study's findings have clearly demonstrated both the potential and current limits of applying QML to acoustic analysis, and in doing so identified areas where application of quantum enhanced models may provide a quantifiable advantage for future acoustic classification tasks.

Session 4pED

Education in Acoustics: Sound Communication in Today's Mediascape

Andrew A. Piacsek, Cochair
Phys., Central Washington Univ., 400 E. Univ. Way, Dept. of Phys., Ellensburg, WA 98926-7422

Kerri D. Seger, Cochair
Integral Consulting Inc., 719 2nd Ave., Seattle, WA 98104

Chair's Introduction—1:00

Invited Papers

1:05

4pED1. From measurement to meaning: Communicating rocket acoustics to the media. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu)

Work conducted at Brigham Young University in rocket-related acoustics has received media attention, including coverage by *The New York Times*, *The Washington Post*, CNN, the BBC, and space-focused outlets such as NASASpaceFlight.com. These interactions have highlighted both the opportunities and challenges inherent in communicating technical acoustics research to broad public audiences. This talk draws on several media engagements to identify lessons that are broadly applicable to acousticians engaging with journalists. Key observations include: (1) scientific accuracy and media clarity are not opposing goals, but they operate under different constraints, particularly time, attention, and narrative structure; (2) uncertainty and scope limitations are among the most difficult concepts to convey, yet are essential to responsible communication; and (3) research findings are often interpreted through a policy lens, requiring additional care in how results and limitations are discussed. Together, these lessons offer practical guidance for communicating acoustics research in ways that preserve technical integrity while promoting public interest and improving understanding.

1:25

4pED2. Lessons learned from enaging with media. Hannah A. Daniel (American Inst. of Phys., 1 Phys. Ellipse Dr., Riverdale, MD 20737, hdaniel@aip.org) and Wendy Beatty (American Inst. of Phys., College Park, MD)

As intermediaries between scientists and the media, we have our fair share of experience helping researchers gain visibility for their work. This talk will cover the scope of our work promoting ASA's conferences. We will also present our lessons learned from years of experience working with researchers and journalists. What makes a good science story? What multimedia assets do journalists want to see? How can you pitch your own work to media outlets? These topics and more will be discussed during our talk.

1:45

4pED3. The layman's perspective as an effective way to improve science communication in Argentina. Fernando M. del Solar Dorrego (Patagon Acoust., 445 Waupelani Dr., Apt B10, State College, PA 16801, fsolar@gmail.com)

The word "acoustics" evokes a series of mental associations when mentioned in a casual conversation. In my experience, non-technical people in Argentina tend to associate acoustics with the problem of noise pollution in cities and with concert hall acoustics. Any further analyses of the areas of application of the science of acoustics on behalf of the layman seem to stop there. When communicating science to the public, journalists and media experts seek to relate the results of a scientific discovery to the way they will affect everyday life, or to marketable concepts, such as space travel or climate change. Fighting against this paradigm and trying to impose complex mathematical or physical concepts in a news story is prone to failure. In this paper I will discuss a series of strategies that were used in Argentina to effectively communicate and disseminate the science of acoustics, based on the premise of gripping the attention of the public with concepts they can relate to.

2:05

4pED4. Social media sociolinguistics and phonetics. Nicole Holliday (Linguist, UC Berkeley, 277 Waverly Dr., Pasadena, CA 91105, nicole.holliday@berkeley.edu)

I'm Nicole Holliday and I do science communication on sociolinguistics and phonetics, mainly on Tiktok but also in traditional media outlets. As part of the panel for the Education in Acoustics and Public Relations Committee joint session, I will contribute to the panel knowledge on topics like crafting a catchy social media grab, ideal posting schedules for the target audience, and how to incorporate technical jargon-y phrases into short-form video content for a general audience.

2:25

4pED5. Story, stakes and action: How to better communicate your work. Bishop Sand (The Washington Post, 1301 K St. NW, Washington, DC 20071, bishop.sand@washpost.com)

This presentation will focus on advice for effectively communicating when you (the scientist) are in control of the presentation (e.g., developing a story structure, setting stakes, condensing your ideas into a few sentences, and clear, active writing). I'll offer other advice when you are not in control of producing the story: when you are a source (e.g., best practices in email communication, during an interview, and some things you can ask for from journalists). I'll also suggest various ways to hone your communication skills and produce your own media. The presentation will involve a brief biographical overview of myself, Bishop Sand, through various careers and stories about restaurant noise and pickleball sound. I'll also motivate an argument for the value of effective communication to the public and outline some similarities between journalism and science.

2:45

4pED6. Learning improvisational comedy to improve science communication skills. Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska – Lincoln, PKI 100C, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

Fundamentals of improvisational comedy provide a good foundation for communicating and presenting science to colleagues and laypersons. The Durham School of Architectural Engineering and Construction at the University of Nebraska–Lincoln has offered one credit hour "Improvisation for Complete Engineers" classes to support science communication skills for our undergraduate and graduate students, beginning in January 2022. In this presentation, we review the core lessons commonly taught in improvisational comedy, describe how those lessons lead to improved communication skills, demonstrate sample improvisational comedy exercises used in those classes, and provide guidance on how others can implement similar courses or integrate the materials into existing curricula.

3:05–3:15 Break

3:15-4:15 Panel Discussion

THURSDAY AFTERNOON, 14 MAY 2026 FRANKLIN HALL A, 1:30 P.M. TO 2:30 P.M.

Session 4pMU

Musical Acoustics: Musical Acoustics Poster Session

Andrew C. Morrison, Chair
Natural Sci., Joliet Junior College, 1215 Houbolt Dr., Joliet, IL 60431

All papers will be on display and all authors will be at their posters from 1:30 p.m. to 2:30 p.m.

Contributed Papers

4pMU1. Characterizing timbre of musical instruments using waveforms, overtones, and Fourier analysis. Sydney R. Markham (Jefferson Community College, 430 Lachenauer Dr., Watertown, NY 13601, sydney.r.markham@icloud.com)

Mathematical concepts such as superposition and Fourier analysis can be difficult to visualize. This project applies these concepts by recreating and comparing musical instrument waveforms. Recordings of a violin, clarinet, piano, and flute were analyzed using Python and Librosa. Short-time Fourier transforms were used to generate spectrograms, then frequency peaks were identified as overtones and used to reconstruct waveforms. By comparing the waveforms to a sine wave and detected overtones to those of an ideal standing wave, timbre differences were analyzed. The violin had gradual falloffs on its spectrogram peaks, a complex waveform, and deviations from expected overtone frequencies with higher harmonics. These were consistent with the stick-slip cycle of bowing, resulting in the violin's bright sound. The flute and clarinet had steeper and fewer spectrogram peaks. However, the flute's waveform was more similar to a sine wave. The clarinet's odd harmonics caused interference resembling a square wave and confirmed its closed-pipe geometry. Both instruments had little deviation from expected overtones. The piano was qualitatively analyzed due to its percussive nature and simultaneous notes. Overall, this work demonstrates how Fourier analysis can be used to model acoustical systems and deviations. [This work was an independent student project.]

4pMU2. User prompting and engagement in generative music platforms. Xinyang Wu (The Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Hong Kong, xwuch@connect.ust.hk)

Prompting has become a new kind of musical instruction in generative music platforms: users write brief descriptions that function like program notes, production briefs, or imagined performance directions. This study surveys how people describe desired music through prompts and how those descriptions relate to audience response, using platform signals such as plays and likes. We categorize prompt language into musically meaningful domains—genre and style lineage, mood and affect, instrumentation and texture, references to place/time, and production or listening-context cues—and examine which vocabularies recur across users. We then explore associations between these prompt categories and engagement, asking what kinds of musical narratives and descriptors are most legible to listeners and most effective at shaping shareable outcomes. Crucially, we use prompt–engagement mismatches to surface creative and technical blind spots: prompt types that reliably underperform can indicate where current models struggle, while consistently successful prompt patterns suggest strengths and emergent aesthetic defaults. The survey highlights opportunities for prompt literacy, cultural analysis, and more transparent human–model co-creation.

4pMU3. Infant-directed instrumental output of music. Lilian Nguyenphuoc (Dept. of Speech and Hearing Sci./Inst. for Learning and Brain Sci., Univ. of Washington, 1715 NE Colombia Rd., Seattle, WA 98195, lilianqn@uw.edu), Lindsay Hippe, Ella De Falco, and Tian C. Zhao (Dept. of Speech and Hearing Sci./Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Exposing infants to music has been shown to enhance babies' neural processing of speech. As such, it is important to explore which particular qualities of music benefit development of speech processing. Past approaches primarily utilized singing caregivers to identify characteristics of infant-directed (ID) song production that differ from adult-directed (AD) song production. This paper investigates if there are analogous differences between ID and AD caregiver music production using an accessible form of purely instrumental music instead: clapping. Analysis will be conducted on a dataset of audio recordings of primary caregivers clapping rhythms of popular songs to their infants and to an adult loved one. As infants are sensitive to acoustics of beat and meter, which can be influenced by parental music background, caregiver scores from a musical perception skill test will also be analyzed. Using various computational approaches, we expect to find acoustic qualities in rhythmic music production that are able to distinguish ID from AD instrumental music production, and that these qualities may correlate to musical perception skill. If such differences exist and correlate to differences in our follow-up MEG study, this may lead to novel implications for future studies of both vocal and instrumental music research in infants.

THURSDAY AFTERNOON, 14 MAY 2026 FRANKLIN HALL 10, 1:00 P.M. TO 2:45 P.M.

Session 4pPAa

Physical Acoustics, Biomedical Acoustics and Structural Acoustics and Vibration: Ultrasonic Characterization Across Scales and Applications II

Andrea P. Arguelles, Cochair
Eng. Sci. and Mech., Penn State Univ., 212 Earth-Eng. Sci. Bldg., University Park, PA 16802

Olivia J. Cook, Cochair
Eng. Sci. and Mech., Penn State Univ., 212 Earth Eng. Sci., University Park, PA 16801

Haley N. Jones, Cochair
Mater. Sci. and Eng., Penn State Univ., N-225 Millenium Sci. Complex, State College, PA 16802

Contributed Papers

1:00

4pPAa1. Nondestructive evaluation of meso-structural and material properties by quantifying the effect of cooling conditions on 3-D printed PLA. Partha Pratim Pandit (Mech. Eng., Purdue Univ., 585 Purdue Mall, West Lafayette, IN 47907, pandit18@purdue.edu), Anna Keim, Harshith Kumar Adepu, Monique McClain, and Luz D. Sotelo (Mech. Eng., Purdue Univ., West Lafayette, IN)

Among additive manufacturing (AM) techniques, fused filament fabrication (FFF) is the most widely adopted. Parameter optimization strategies are commonly implemented to provide some control over print quality, while optical and thermal measurements aim to identify defects in the printed parts. However, challenges regarding both quality and repeatability, and FFF parts are notorious for their variability across prints and printers in terms of both defect content and material properties. Thus, there is still a gap in understanding of the relationship between printing operating conditions and part quality. The purpose of this study is to nondestructively quantify the effect of cooling via forced convection on the material properties and geometrical accuracy of FFF samples. Samples with varying cooling fan activation layers were produced using polylactic acid (PLA) filament. Ultrasonic scanning, X-ray computed tomography (X-ray CT), and optical profilometry, were leveraged to quantify the change in acoustic properties,

porosity content, and deformation due to the different cooling conditions. Spatial maps of acoustic properties, porosity, and deformation qualitatively showed property variation trends throughout the build that were associated to the cooling conditions. The relationships between porosity, deformation, and acoustic properties were also quantitatively assessed to identify dominant factors on the acoustic property trends.

1:15

4pPAa2. Ultrasonic characterization of poromechanical behavior in water-sorbing nanoporous glass. Ashoka Karunarathne (Dept. of Phys. and Astronomy, The Univ. of MS, 145 Hill Dr., P.O. Box 1848, Univ., MS 38677, atholang@olemiss.edu), Alexei Khalizov, and Gennady Gor (Dept. of Chemical and Mater. Eng., New Jersey Inst. of Technol., Newark, NJ)

Nanoporous materials with pore size ∼1 to 100 nm, are widely used as absorbents due to their high surface area and tunable structural properties. Fluid confinement in nanopores strongly affects the poromechanical behavior of saturated porous media due to significant solid–fluid intermolecular interactions. This behavior includes changes in the stiffness of both solid matrix and confined fluid, as well as adsorption-induced deformation of the material. Currently, the interplay between fluid sorption and poromechanical responses has not yet been systematically investigated. To address this knowledge gap, we performed simultaneous measurements of water sorption, ultrasonic wave propagation, and adsorption-induced deformation on nanoporous Vycor glass. The observed variations in both longitudinal and shear wave speeds clearly reflect key features of the sorption isotherm, including monolayer adsorption, capillary condensation, and condensate evaporation. Elastic moduli derived from the wave speeds reveal the change of material stiffness throughout the water sorption process. Interpreting these trends of material stiffness requires a combined analysis of adsorption-induced deformation and water sorption data. Furthermore, the measured ultrasonic attenuation reflects the occurrence of capillary condensation and evaporation. Overall, this study demonstrates the utilization of ultrasonic characterization for probing fluid sorption in nanoporous materials and understanding their associated poromechanical behavior.

1:30

4pPAa3. Real-time ultrasonic monitoring of cold sintering evolution. Jian Song (Eng. Sci. and Mech., Penn State Univ., 656 White Course Dr., 212 Earth-Eng. Sci. Bldg, University Park, PA 16802, jbs7472@psu.edu), Silas Wieland, Kevin Quaye, and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Cold sintering densification evolves rapidly under coupled thermal–mechanical conditions, motivating *in situ* measurements that track consolidation as it occurs. We present an ultrasonic monitoring platform that integrates press-based sensing to quantify densification during the cold sintering process. A rigid, bottom-mounted pulse-echo configuration couples a longitudinal transducer to the underside of the press base plate via a custom designed mounting plate, enabling repeatable measurements through the die stack without disrupting loading. LabVIEW synchronization allows real-time extraction of time-of-flight and attenuation metrics. The objectives are to (i) identify sensor placements and geometries that minimize temperature and pressure exposure while preserving informative propagation paths, (ii) validate arrival-time interpretation using room-temperature trials and *ex-situ*/interrupt-stage specimens, and (iii) determine the densification range that yields reliable ultrasonic signatures for subsequent in-process studies. The resulting dataset enables quantitative correlations between geometric densification, thermal evolution, and acoustic parameters, supporting systematic investigation of consolidation kinetics and identification of process windows associated with optimal mechanical and electrical properties. This capability provides a foundation for interpreting microstructure-driven changes during CSP and for guiding process adjustments aimed at improved uniformity and performance.

1:45

4pPAa4. Beyond point measurements: Mapping elastic moduli in cold sintered ceramics using oblique-incidence immersion ultrasound. Silas Wieland (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci. Bldg., University Park, PA 16802, srw5795@psu.edu), Haley N. Jones (Mater. Sci. and Eng., Penn State Univ., State College, PA), Susan Trolier-McKinstry (Mater. Sci. and Eng., Penn State Univ., University Park, PA), and Andrea P. Argüelles (Eng. Sci. and Mech., Penn State Univ., State College, PA)

The cold sintering process (CSP) produces ceramic microstructures containing residual porosity and a compliant intergranular phase. Conventional quantitative characterization via immersion ultrasound forms an under-constrained inverse problem, as longitudinal wave speed alone cannot differentiate compliance effects arising from pores versus soft grain boundaries. To decouple their mechanical contributions, a custom oblique-incidence immersion C-scan fixture was developed to perform spatially resolved shear wave velocity mapping via mode conversion. Operating at 20MHz, the system enables two-dimensional mapping of longitudinal and shear wave velocities generated through raster scanning at set angles of incidence. Measurements in the long-wavelength limit allow the microstructure to be treated as an effective medium such that local elastic moduli can be estimated from the measured velocity maps. Control samples showed a monotonic increase in both moduli with density, confirming the system's sensitivity to local stiffness variations. Comparison between as-sintered samples and heat-treated controls (where boundaries are recrystallized) demonstrates separable contributions of porosity and intergranular compliance. Spatially resolved shear-wave immersion measurements provide information inaccessible to point-measurement ultrasonic methods, enabling improved elastic property characterization in multiphase systems.

2:00

4pPAa5. Ultrasonic *in situ* monitoring of multilayer ceramic capacitors for degradation and damage assessment. Haley N. Jones (Mater. Sci. and Eng., Penn State Univ., N-225 Millenium Sci. Complex, State College, PA 16802, hnj5051@psu.edu), Silas Wieland, Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA), and Susan Trolier-McKinstry (Mater. Sci. and Eng., Penn State Univ., University Park, PA)

Multilayer ceramic capacitors (MLCCs) are critical components in electronic circuits and understanding MLCC degradation is essential for improving circuit reliability. MLCCs can fail catastrophically through dielectric breakdown, and they can fail gradually via decreased insulation resistance over time. A widespread method of measuring the evolving degradation state of MLCCs is leakage current monitoring under DC bias, but this method lacks spatial resolution to identify failure locations and complimentary techniques require destructive sample preparation. This study addresses these limitations by utilizing high frequency ultrasound (100 MHz) for failure detection in commercial MLCCs simultaneously subjected to 300 V DC bias over 3 h. Quantitative analysis of wave speed and attenuation revealed time-dependent changes correlated with applied voltage. Wave speed and attenuation increased after 1 h into high voltage application, followed by a drastic decrease in attenuation occurring after 2.5 h. Internal heating during testing may have induced domain reorientation, causing the material to stiffen and thus increase wave speed. Domain structure changes could increase ultrasonic scattering and attenuation, and the drastic attenuation decrease observed suggests domain wall elimination from surpassing the Curie temperature. Overall, this work highlights ultrasonic capability of *in situ* detection of temporal changes in MLCCs under operational electric stress.

2:15

4pPAa6. Effect of particle concentration on the formation of striations in aqueous suspensions. Nicholas M. Rivet (Mech. Eng./Pharmaceutical Sci., Western New England Univ., 1215 wilbraham Rd., Springfield, MA 01119, nicholas.rivet@wne.edu), Matthew Dougherty (Pharmaceutical and Administrative Sci., Western New England Univ., Springfield, MA), Ali Elhage (Mech. Eng., Western New England Univ., Springfield, MA), and Bart Lipkens (Pharmaceutical and Administrative Sci., Western New England Univ., Springfield, MA)

Planar ultrasonic standing waves are widely used to manipulate particles for concentration, washing, or fractionation. The acoustic radiation force moves particles to pressure nodal or anti-nodal planes based on acoustic contrast factor and forms striations or gratings of particles at half-wavelength spacing. The particle number density profile along the acoustic axis is a function of the particle's effective potential energy to the thermal energy of the suspended particle (Kwiatkowski, Marston, 1998). We report on the effect of particle concentration on the formation of the striations and acoustic standing wave properties. The experimental setup is a flow-through system in a rectangular duct with two transducers, opposite each other, located in the duct walls. One transducer is used as a transmitter and the other as a probe. The particle suspension occupies either the entire volume, or only a smaller central region of the volume surrounded by clear buffer fluid. Polystyrene (10μm) and polyethylene (30μm) particle suspensions of various concentrations flow through the acoustic standing wave excited at nominal frequencies of 1 and 3MHz. The effects of frequency, particle size, concentration, flow velocity, and transducer power on formation of striation, resonance frequency shift, and quality factor of the standing wave are evaluated.

2:30

4pPAa7. Distortion in wave speed measurements when using focused ultrasonic transducers. Aleta Lyzinski (Eng. Sci. and Mech., Penn State Univ., 5524 Indian Ridge Rd., Doylestown, PA 18902, aal5683@psu.edu), Silas Wieland, and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Accurate determination of longitudinal wave velocity is fundamental to material characterization and high-resolution acoustic imaging techniques. In immersion testing, spherically focused immersion transducers are used to improve spatial resolution; however, their beam convergence introduces geometric diffraction effects that affect apparent time-of-flight. These artifacts result in systematic, thickness-dependent velocity overestimations observed up to 200 m/s when compared to planar transducer measurements and literature values. A critical challenge in correcting these errors is the discrepancy between manufacturer specification nominal focal geometries and the true beam propagation. This study builds upon a series of pulse-echo immersion experiments performed on fused silica samples with thicknesses ranging from 1/16 to 1/2 inches, using transducers with center frequencies primarily of 10, 15, and 20 MHz, and corresponding F-numbers spanning approximately 2 to 8. Measured wave speed error was found to increase with decreasing F-number when comparing focused responses against unfocused transducers. Beam field propagation was modeled using the semi-analytical simulation platform CIVA (CEA-LIST), which required iterative optimization of transducer inputs to account for uncharacterized lens properties. These findings quantify diffraction-related wave velocity error and form a framework for the development of an analytical diffraction correction factor, improving the accuracy of quantitative high-resolution ultrasonic imaging.

Session 4pPAb

Physical Acoustics: Infrasound II

Philip S. Blom, Cochair
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David N. Green, Cochair
AWE Blacknest, AWE Blacknest, Brimpton, RG7 4RS, United Kingdom

Kyler Goddard, Cochair
Air Force, 10989 South Patrick Dr., Patrick AFB, FL 32925

Invited Papers

1:00

4pPAb1. Infrasound directivity from equivalent acoustic and ground-surface reflected sources. Alexandra Iezzi (Cascades Volcano Observatory, U.S. Geological Survey, 1300 SE Cardinal Court, Vancouver, WA 98683, aiezzi@usgs.gov), Robin Matoza (Dept. of Earth Sci. and Earth Res. Inst., Univ. of California Santa Barbara, Santa Barbara, CA), Emma Opper (Dept. of Appl. Mathematics, Univ. of Colorado Boulder, Boulder, CO), and Keehoon Kim (Geophysical Monitoring Program, Lawrence Livermore National Lab., Livermore, CA)

Quantifying infrasound source parameters such as mass flow rate of a volcanic eruption or yield of an anthropogenic explosion using infrasound relies upon assumptions of the acoustic source type. Sources may produce omnidirectional or directional wavefields, and propagation effects such as interactions with topography can also induce wavefield directionality. Sampling of directional wavefields is often limited (particularly in the vertical direction), which may inhibit our ability to infer the underlying acoustic source and therefore estimate source parameters that may be vital for research and monitoring. Here we review equivalent acoustic sources, with a focus on the scale and wavelengths for local volcano infrasound field studies, and highlight the equivalent bipole source that can be induced by ground reflection of an elevated monopole source. We find that both increasing the source height above the ground and increasing the source frequency leads to increased directivity in the resultant wavefield. Using numerical modeling of a simple monopole source, we illustrate that both horizontal and vertical directivity are introduced by interactions with the topography of a volcanic crater. Information on infrasound directivity explored in this study may be used to help guide future infrasound deployments intended to estimate source parameters or quantify wavefield directivity.

1:20

4pPAb2. Seismo-acoustic signatures of crashing ocean waves (surf). Robin S. Matoza (Dept. of Earth Sci., Univ. of California, Santa Barbara, CA 93106, matoza@geol.ucsb.edu), Jeremy Francoeur (Dept. of Earth Sci., Univ. of California, Santa Barbara, CA), Hugo Ortiz (Dept. of Earth, Atmospheric and Planetary Sci., Massachusetts Inst. of Technol., Santa Barbara, CA), and Rodrigo de Negri (CNRS, IRD, OPGC, Laboratoire Magmas et Volcans, Université Clermont Auvergne (UCA), Aubière, France)

Surf infrasound has been recorded for decades on a variety of global seismo-acoustic networks and is a well-known noise source primarily in the band 0.5–5 Hz. We conducted a series of field experiments designed to study seismo-acoustic surf signals in Santa Barbara, California. Our data included one continuously running infrasound sensor (COPR, September 2022 to July 2023) and several short-duration (∼1–6 days) field campaigns involving multiple infrasound arrays, co-located seismometers, and time-synchronized video recordings (*Francoeur et al.* 2025). Our results demonstrate that seismo-acoustic signals from surf are composed of repetitive transient events, which can be detected and characterized using template matching. We record the repetitive surf infrasound waveforms at COPR throughout varied oceanographic and atmospheric conditions over the course of the ∼11 month deployment; the amplitudes of detected infrasound signals correlate with offshore significant wave height and local wind speed. Here we present an overview of the observations (*Francoeur et al.* 2025) and compare strategies for source localization of the surf infrasound transients. Francoeur, J.W., R.S. Matoza, H.D. Ortiz, R. De Negri (2025), Identification of transient seismo-acoustic signals from crashing ocean waves: Template matching and location of discrete surf events, *Geophys. J. Int.*, **243**, 2, ggaf317, doi:10.1093/gji/ggaf317.

1:40

4pPAb3. Design of coiled acoustic channels for compact directional infrasound sensing. Sarah Albert (R&D, Sandia National Labs., Albuquerque, NM), Cameron A. McCormick (Simulation Modeling Sci., Sandia National Labs., Albuquerque, NM), Jerry W. Rouse (R&D, Sandia National Labs., Albuquerque, NM), Benjamin C. Treweek (Computational Solid Mech. and Structural Dynam., Sandia National Labs., P.O. Box 5800, Albuquerque, NM 87185, btrewee@sandia.gov), and Benjamin C. White (R&D, Sandia National Labs., Albuquerque, NM)

Determining direction of arrival (DOA) for infrasound is important for monitoring both anthropogenic and natural phenomena such as nuclear explosions, volcanic eruptions, large-caliber gunfire, and general urban activity. Existing infrasound sensor arrays have been successful at determining DOA for global- and regional-scale events, but due to their large spatial footprint and long deployment time, such arrays are impractical for local-scale monitoring in environments with diverse terrain or space limitations. Recent work by Rouse *et al*. has shown theoretical success using coiled acoustic channels to construct a sub-wavelength DOA sensor with a characteristic length of only a few meters. In this work, we use Sierra/SD, a massively parallel HPC application developed for finite element method simulations in structural dynamics and acoustics, to investigate designs that reduce the spatial footprint further to a characteristic length of one meter or less. We examine a variety of possible design modifications and discuss their practical advantages and limitations, and we present the scattered acoustic response for various design combinations. Finally, we identify a candidate for experimental testing using a modular, multi-layer design. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

1:55

4pPAb4. Experimental methods in compact directional infrasound sensing. Sarah Albert (R&D, Sandia National Labs., Albuquerque, NM), Daniel C. Bowman (Pacific Northwest National Lab., Richland, WA), Cameron A. McCormick (Simulation Modeling Sci., Sandia National Labs., Albuquerque, NM), Jerry W. Rouse (Analytical Structural Dynam., Sandia National Labs., Sandia National Labs., Albuquerque, NM 87185, jwrouse@sandia.gov), Benjamin C. Treweek (Computational Solid Mech. & Structural Dynam., Sandia National Labs., Albuquerque, NM), and Benjamin C. White (Sandia National Labs., Albuquerque, NM)

Recent work by Rouse *et al*. presented a design for infrasound directional-of-arrival determination that significantly reduces the spatial footprint by using coiled acoustic channels. The design consists of four coiled channels arranged in a two-by-two array. The direction of arrival is determined from the relative amplitude of the pressure at the bottom of each channel. In this work, we present experimental results obtained from full-scale prototypes of the original design and also recently developed advanced designs. Data are shown from prototypes deployed at the Energetic Materials Research and Testing Center (EMRTC) and Kirtland Air Force Base (KAFB), and successes and challenges are discussed for each prototype. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

2:10

4pPAb5. Frequency-dependent specular reflection and topographic scattering during the OSIRIS-REx reentry captured using balloon-borne microbarometers. Elizabeth Silber (Sandia National Labs., 1515 Eubank Blvd. NE, Albuquerque, NM 87123, esilber@uwo.ca), Daniel C. Bowman (Pacific Northwest National Lab., Richland, WA), Jonathan Lees (The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Keehoon Kim (Geophysical Monitoring Program, Lawrence Livermore National Lab., Livermore, CA), and Siddharth Krishnamoorthy (Jet Propulsion Lab. California Inst. of Technol., Pasadena, CA)

Infrasound may reflect off the Earth's surface one or more times on its path from source to receiver. Theory and simulations suggest that low frequency waves experience minimal absorption, while higher frequency waves suffer attenuation due to ground porosity. Studies also indicate that topography can induce scattering. However, direct acoustic observations of these effects at high incidence angles are lacking. We used multiple balloon-borne microbarometers to capture the direct acoustic wave and ground echo during the hypersonic reentry of the OSIRIS-REx space capsule. We found that the acoustic reflection coefficient was approximately unity below ∼10 Hz but decreased as frequency squared in the high infrasound/low audio range. Reflections from flat ground also returned an impulsive arrival, but reflections from mountains produced a diffuse wave train we attribute to topographic scattering. These findings support the assumption of near-perfect reflectors in the low infrasound range but indicate that higher frequency sounds may be strongly attenuated depending on the state of the ground surface. Furthermore, interactions with topography may greatly diminish the range to which low frequency sounds can travel. These results can be used to improve acoustic propagation simulations. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525. Cleared for release.]

2:25

4pPAb6. Stochastic infrasonic propagation including buoyancy wave impacts. Philip S. Blom (Los Alamos National Lab., P.O. Box 1663, M/S F665, Los Alamos, NM 87545, pblom@lanl.gov) and Jordan W. Bishop (Los Alamos National Lab., Los Alamos, NM)

Infrasound propagating through the atmosphere is refracted by temperature and wind gradients; however, some atmospheric processes occur at subgrid scales and are not captured in temporally averaged atmospheric models. One such process is buoyancy- or gravity-induced waves that propagate upward through the atmosphere with frequencies covering the range from 0.1 to 10 mHz. Such waves perturb the atmospheric wind structure and impact infrasonic propagation paths that extend through the middle and upper atmosphere. In the stratosphere, fine structure produces scattering and dispersive effects. In the meso- and thermosphere, decreasing density increases the amplitude and vertical wavelength of the buoyancy waves and larger perturbations are expected with gravity wave breaking in the thermosphere that can have a significant impact on propagation path geometry. A simplistic framework for generating gravity wave realizations to perturb the atmosphere in stochastic infrasonic propagation studies will be presented along with example predicted propagation impacts using finite-frequency and ray tracing analyses.

2:40–3:00 Break

3:00

4pPAb7. Quantifying the influence of local topography on infrasound wind noise levels. Ceu Jesus (Univ. of MS, Univ., Oxford, MS 38677, mnetode@go.olemiss.edu), Roger M. Waxler, Lance Yarborough (Univ. of MS, Univ., MS), Carrick Talmadge (Univ. of MS, Oxford, MS), and Claus Hetzer (Univ. of MS, Univ., MS)

Wind-induced turbulent pressure fluctuations, known as "wind noise," are the dominant noise source in infrasound measurements, limiting detection capability. Wind noise levels are strongly influenced by local topography, yet their relationship remains poorly quantified. This study develops a quantitative framework linking measurable topographic features to wind noise characteristics for improved site selection. Infrasound data collected at the University of Mississippi's Biological Field Station (UMBFS) were analyzed in 0.1–1 Hz and 1–10 Hz frequency bands. Root mean square (RMS) pressure time series were statistically characterized using Kernel density estimation (KDE) to assess probability density functions (PDFs) and corresponding cumulative distribution functions (CDFs). Topographic variables—including ground elevation, vegetation density, and canopy height—were derived from airborne and terrestrial LiDAR. Results have been demonstrating distinct noise distributions between sites with different topographic characteristics separated by as short as 20-m. These findings support the development of a predictive algorithm that uses topographic inputs to estimate probabilities of wind noise levels remaining below specified

thresholds, providing a quantitative basis for improved infrasound site selection and local wind noise mapping.

3:15

4pPAb8. Experimental evidence of flow separation observed in measurements of near-surface velocity and surface pressures of a spherical windscreen in atmospheric turbulence. Gordon M. Ochi (U.S. Army ERDC, 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil), Carl R. Hart (U.S. Army Engineer Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH), and Benjamin F. Racelis (U.S. Army ERDC, Champaign, IL)

Current models for the wind noise reduction (WNR) of microphone windscreens in atmospheric turbulence generally assume that surface pressures average out over the surface of a windscreen. At infrasonic frequencies the surface pressure fluctuations on the upstream side are approximately the opposite of the downstream side, producing a cancellation effect and yielding the WNR. Under this assumption, both the mean surface pressures and near-surface turbulent velocity must be homogeneous. The WNR should approach that of an unscreened microphone. In this study, we examine measurements of the near-surface turbulent velocity and surface pressures, which suggest that both are inhomogeneous. Near-surface fluctuating velocity is measured through the use of tri-axial, cross-flow constant temperature anemometry probes positioned at several radii and at 30 deg angle increments longitudinally about the screen. Surface pressures were measured also in 30 deg increments longitudinally about the screen through probe microphones embedded along the surface. In addition to highlighting the inhomogeneous nature of these phenomena, the data show that flow separation is a likely source of inhomogeneity.

3:30

4pPAb9. Data-efficient infrasound characterization using convolutional neural networks. Sarah Albert (R&D, Sandia National Labs., Albuquerque, NM), Cameron A. McCormick (Simulation Modeling Sci., Sandia National Labs., 1611 Innovation Pkway SE, Albuquerque, NM 87123, camccor@sandia.gov), Jerry W. Rouse (Albuquerque, NM), and Benjamin C. Treweek (Computational Solid Mech. and Structural Dynam., Sandia National Labs., Albuquerque, NM)

In this presentation, we will describe the application of convolutional neural networks (CNNs) for the detection and characterization of infrasound sources, utilizing signals from a coiled-space infrasound sensor. We will begin by describing this novel sensor, the motivation behind its design, and its advantages. We will then describe our machine learning approach. The approach begins by training the CNN on an arbitrarily large dataset of synthetic signals generated from numerical models, allowing the network to learn underlying patterns. Subsequently, we fine-tune the CNN using a significantly smaller dataset derived from experimentally measured responses, which greatly improves accuracy on real test data. We will present the performance of this approach on a theoretical infrasound application, discuss practical limits, and explore the minimal amount of measured data required to maintain sufficient accuracy. Our findings demonstrate the efficacy of even simple CNN architectures in infrasound source characterization while addressing challenges related to data scarcity. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

3:45

4pPAb10. Reinforcement learning-based sequential model selection for robust multimodal yield estimation. Christophe Millet (CEA, CEA, DAM, DIF, Bruyères-le-Châtel 91297, France, christophe.millet@cea.fr)

Estimating the yield of explosive events from acoustic observations remains a challenging inverse problem due to strong dependencies on poorly constrained environmental parameters. In practice, yield estimation relies on heterogeneous numerical models and observables whose domains of validity only partially overlap, and which are often used independently across sensing modalities, such as infrasonic and seismic data. This frequently leads to inconsistent estimates and a limited exploitation of complementary information. In this work, we reformulate yield estimation as a robustness-driven inference problem, where the objective is to minimize sensitivity to dominant sources of uncertainty rather than to combine independent inversions. We propose a sequential decision framework in which a reinforcement learning (RL) policy learns how to adaptively select propagation models, sensing modalities, and quantities of interest, reflecting different physical approximations, before estimating source yield. RL training is enabled using pre-trained surrogate models, including neural operators and stochastic perturbation models, allowing efficient exploration of inference strategies under controlled uncertainty. At inference time, a language-model interface instantiates the learned RL policy using real-world data, simulation tools, and expert constraints, ensuring interpretability of the decision process. The approach is illustrated through the reanalysis of explosive events, highlighting its potential to deliver a single yield estimate.

4:00

4pPAb11. Considerations in determining timing and position of acoustic events during Starship launches. Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., N357 ESC, Provo, UT 84602, grant_hart@byu.edu), Kent L. Gee, Makayle S. Kellison, and Noah L. Pulsipher (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

During the 2024–2025 SpaceX Starship flight tests at Starbase, Texas, Brigham Young University (BYU) deployed approximately 20 acoustic recording stations to monitor launch and reentry phenomena. Beyond normal launch and landing noise, the network captured distinct impulsive events, including the ignition overpressure (IOP) from launch startup and sonic booms from the super-heavy booster and hot-staging ring return to the atmosphere. These events (as well as the launch noise) have spectral peaks in the high-infrasound region of the spectrum. Using the GPS-enabled clocks in our stations, the data are synchronized with sub-millisecond precision. We then performed spatio-temporal localization of these sources. Variations in signal detectability across the network between different launches highlight significant meteorological influences. We were also able to detect a measurable sound-speed differential between over-water (Gulf) and over-land propagation paths. Under some simple meteorological conditions surface winds significantly influence results in a predictable way, but at other times their influence is unclear.

4:15

4pPAb12. Magnitude and phase estimation techniques for very-low frequency reciprocity calibration. Thomas B. Gabrielson (Penn State Univ., State College, PA) and Chad M. Smith (Penn State Univ., 225 Sci. Park Rd., State College, PA 16803, cms561@psu.edu)

Low-uncertainty calibration of infrasound sensors by reciprocity can be challenging, particularly below one hertz. Long-stroke loudspeakers can provide higher acoustical volume velocity than other drivers; however, reciprocity requires that the reciprocal transducer—the loudspeaker—be approximately linear, which limits the usable amplitude in transmit mode. Furthermore, the low coupling factor of a loudspeaker means that, in receive mode, the output voltage may be quite small. These factors conspire to limit the signal-to-noise ratio (SNR) at sensor outputs and the associated uncertainty can dominate over measurement-apparatus uncertainties. A critical part of reciprocity analysis is formation of complex ratios—ratios of various sensor outputs to the reciprocal transducer's drive current. One of the strengths of practical implementations of reciprocity is that the SNR of the drive-current channel can be very high. This suggests synchronized processing for estimating complex amplitudes; that is, using the high-SNR channel to find the accurate signal frequency that is then used as reference for the lower-SNR channels. This presentation is based on demonstration of synchronous analysis for large-chamber reciprocity from 0.02 to 1.00 Hz. The agreement between a frequency-domain process and a time-domain process suggests that either of these methods may yield lower uncertainty than conventional methods.

Session 4pPAc

Physical Acoustics: Physical Acoustics Poster Session

Andrea P. Arguelles, Chair
Eng. Sci. and Mech., Penn State Univ., 212 Earth-Engr. Sci. Bldg., University Park, PA 16802

Philip S. Blom, Chair
Earth and Environ. Sci., Los Alamos National Lab., P.O. Box 1663, M/S F665, Los Alamos, NM 87545

David N. Green, Chair
AWE Blacknest, AWE Blacknest, Brimpton, RG7 4RS, United Kingdom

All posters will be on display and all authors will be at their posters from 3:30 p.m. to 5:30 p.m.

Contributed Papers

4pPAc1. Seismic and infrasonic signatures of icy environments. Subhatra Sivam (Earth Sci., UC Santa Barbara, 2042 Webb Hall, Univ. of California, Santa Barbara, CA 93106-9630, ssivam@ucsb.edu) and Robin Matoza (Earth Sci., UC Santa Barbara, Santa Barbara, CA)

Glacial dynamic processes are sources of seismic and infrasonic signals in polar regions. This study will process seismoacoustic data from the Arctic region to detect and localize coherent, near-sensor signals. These detections will then be compared with satellite imagery to investigate their cryospheric origin. Integrating multiple data types is expected to reveal seismoacoustic sources associated with various ice-related phenomena, exploring the potential for acoustic remote sensing to track and understand cryospheric changes.

4pPAc2. Radiosonde-to-Space (R2S) atmospheric specifications: Bridging observations and models for infrasound propagation. Loring Schaible (Sandia National Labs., Albuquerque, NM) and Elizabeth Silber (Sandia National Labs., 1515 Eubank Blvd. NE, Albuquerque, NM 87123, esilber@uwo.ca)

The propagation of infrasonic ($\sim$0.01–20 Hz) pressure oscillations in the atmosphere is highly asymmetric and time-dependent. Therefore, modeling is key to predicting when and where a signal may arrive as well as interpreting observations. The usefulness of modeling results depends on how accurately the sound velocity model used reflects the true conditions of the atmosphere through which the signal travels. The Naval Research Laboratory's Ground-to-Space (G2S) model is a commonly used source for specifications, combining low-atmosphere observations and high-altitude climatological data into a single, altitude-varying sound velocity model for any location and time. While G2S availability has greatly improved propagation modeling efforts, there is still a demonstrated need for more accurate specifications. Here, we describe a method for calculating what we call Radiosonde-to-Space (R2S) specifications, G2S profiles augmented with direct observations of the lower atmosphere collected by radiosondes. We then compare over 6000 pairs of G2S-R2S specifications for Albuquerque, NM and the modeling results therefrom. While most G2S-R2S pairs are very similar, G2S profiles do not always reflect the conditions observed by radiosondes, leading to substantial differences in where arrivals are predicted. This work demonstrates the importance of using timely radiosonde observations when available. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

4pPAc3. DL-G2S: A deep learning Ground-to-Space uncertainty model for infrasound propagation. Sarah Albert (R&D, Sandia National Labs., Albuquerque, NM), Elizabeth Silber (Sandia National Labs., 1515 Eubank Blvd. NE, Albuquerque, NM 87123, esilber@uwo.ca), Julia Sakamoto, and Minjae Cho (Sandia National Labs., Albuquerque, NM)

Infrasound, low frequency sound less than 20 Hz, is generated by a variety of natural and anthropogenic sources (chemical and nuclear explosions, volcanic eruptions, earthquakes, etc.). Propagation modeling allows us to understand the path an infrasound wave takes through the atmosphere, and is important for both predicting and understanding signal arrivals. Infrasound propagation is driven by effective sound speed, which is a function of wind and temperature. Therefore, atmospheric specification profiles and corresponding uncertainty are essential for accurate infrasound propagation modeling. Current atmospheric uncertainty models rely on large datasets and are computationally expensive. A deep learning solution promises portability and speed. Here we present a new deep learning model, a conditional generative adversarial network, for atmospheric specification uncertainty analysis for infrasound propagation modeling. We show comparisons to the current state-of-the-art, quantify increase in accuracy and speed, and show applications for explosion monitoring. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

4pPAc4. Shock wave formation in the thermosphere by an earthgrazing fireball. Elizabeth Silber (Sandia National Labs., 1515 Eubank Blvd. NE, Albuquerque, NM 87123, esilber@uwo.ca), Denis Vida (Western Univ., London, ON, Canada), Miro Giannone (Sandia National Labs., Dallas, TX), Jamie Shepherd (Easter Auchraw Cottage, Lochearnhead, United Kingdom), Sarah Albert (R&D, Sandia National Labs., Albuquerque, NM), Daniel C. Bowman (Sandia National Labs., Richland, WA), Tammy Do, Margaret Campbell-Brown (Western Univ., London, ON, Canada), Peter Jenniskens (SETI, San Francisco, CA), and Reynold Silber (Sentinel SCG LLC, Rio rancho, NM)

A rare class of meteoroids known as earthgrazers enter the atmosphere at shallow angles, with some returning to space after a brief hypersonic flight through the upper atmosphere. We present the detection and analysis of infrasound from a rare horizon-to-horizon earthgrazer event observed over northern Europe on September 22, 2020. The fireball generated ballistic shockwaves which were detected by three stations of the Royal Netherlands Meteorological Institute (KNMI) network. Despite the high-altitude trajectory, the pressure wave reached the ground at low frequencies detectable by infrasonic instruments. The infrasound signal exhibited a high trace velocity, indicative of near-vertical arrival angles, with most energy

concentrated $< \sim 4$ Hz. The modeled and observational data revealed that infrasound emanated from three distinct parts of the fireball trail. This finding marks a significant milestone, representing the first documented evidence of capturing ballistic shocks from multiple distinct parts of the trail of a high-altitude fireball using infrasound. Another notable aspect of this fireball was that it was detected despite its very high altitude (>90 km). This event reinforces the potential of infrasound as a tool for monitoring and detecting unconventional high-altitude sources, such as fireballs. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

4pPAc5. Dense seismoacoustic observations of the 2025 Alaska fireball. Logan Scamfer (Sandia National Labs., Albuquerque, NM), Elizabeth Silber (Sandia National Labs., 1515 Eubank Blvd. NE, Albuquerque, NM 87123, esilber@uwo.ca), Marc Fries (NASA, Houston, TX), Denis Vida (Western Univ., London, ON, Canada), Damir Segon (Astronomical Society Istra Pula, Pula, Croatia), Peter Jenniskens (SETI, San Francisco, CA), Yasuhiro Nishikawa (Osaka Kyoiku Univ., Osaka, Japan), Vedant Sawal (Sandia National Labs., Albuquerque, NM), and Trevor Rector (Univ. of Alaska Anchorage, Anchorage, AK)

Fireballs are luminous phenomena formed when entering Earth's atmosphere at hypersonic velocities, generating shock waves that rapidly decay into acoustic waves detectable by infrasound and seismic sensors. On April 24, 2025, a bright daytime fireball traversed the Cook Inlet in Southcentral Alaska. This event was recorded by >50 individual seismic and infrasound sensors, four infrasound arrays, and multiple cameras. Additionally, doppler radar imagery identified a possible meteorite fall area around Whitefish Lake. We analyze this fireball's geophysical signatures to characterize trajectory, energy, velocity, and the infrasound source function. Acoustic signatures allowed detailed examination of the fireball's spatiotemporal evolution. Multiple array detections enabled back-projection analysis to identify likely fragmentation points. Ballistic acoustic arrivals further refined trajectory parameters, including entry location and azimuth. Infrasound propagation modeling at inferred fragmentation sites provided theoretical verification against observed signals, supporting trajectory reconstructions. The density of regional seismoacoustic sensors and comprehensive observational coverage make this event uniquely valuable for advancing our understanding of fireball dynamics and acoustic propagation. This study emphasizes the critical role of seismoacoustic monitoring in effectively detecting and analyzing unpredictable, high-energy atmospheric phenomena. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

4pPAc6. Resonance behavior of a bubble near a boundary: Application to elastography at the microscale. MICOL Thomas (Inserm, LabTAU, 43 rue Raoul Servant, Lyon 69007, France, thomas.micol@univ-lyon1.fr), Alexander Doinikov (Inserm, LabTAU, Lyon, France), Cyril Mauger (LMFA, Lyon, France), and Claude Inserra (Inserm, LabTAU, Lyon, France)

Gas microbubbles subjected to ultrasonic fields act as resonant systems with high quality factors used in numerous engineering and medical applications, ranging from surface cleaning to therapeutic drug delivery. Understanding the acoustic properties of these bubbles, specifically their oscillation dynamics, is crucial to master their interaction with the surrounding medium. In this context, we present an analytical model describing the frequency response of a gas microbubble oscillating near a spherical inclusion of arbitrary size and mechanical nature (rigid, fluid, or viscoelastic) immersed in a viscous compressible fluid. The theoretical framework considers both radial and nonspherical oscillations in the linear regime. It predicts how resonance frequencies and oscillation amplitudes are altered by the bubble size, material properties, and the distance to the nearby sphere. As a key application, we demonstrate that scanning the frequency response of a bubble near a viscoelastic object, such as an erythrocyte-like particle mimicking a biological cell, offers a pathway to recover its mechanical properties through inverse modeling. These results open new possibilities for high-resolution elastography at the microscale using stable cavitation.

4pPAc7. A mechanistic insight into sonochemical nitrogen fixation. Xuelei Pan (Eng. Sci., Univ. of Oxford, Warwick, United Kingdom), Davide Presso (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Qian Liu (Hoffmann Inst. of Adv. Mater., Shenzhen Polytechnic Univ., Shenzhen, China), Lucia Mullings, Mohamad Salimi, Yi Qin, Pankaj Sinhmar (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and James Kwan (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, james.kwan@eng.ox.ac.uk)

Acoustic cavitation concentrates ultrasonic energy into transient microreactors capable of driving chemistry far from equilibrium. Here, we investigate catalyst-free nitrogen fixation driven by inertial cavitation and quantify the effects of acoustic parameters on radical generation and product selectivity. Using a combination of experimental and theoretical methods, we show that pulsed focused ultrasound (e.g., 820 kHz, 6.1% duty cycle), gas–vapour bubbles nucleate, expand, and collapse nearly adiabatically, producing hotspot temperatures >5000 K with ultrafast heating/cooling ($\sim 10^{12}$ K/s) and a $\sim$3 ns thermal pulse. These extremes enable N≡N activation during the spike, followed by rapid quenching that stabilizes products before reverse-reactions can occur, thereby avoiding any stable equilibrium. By varying feed-gas composition and cavitation dynamics, we switch between oxidative NOx chemistry (nitrite/nitrate) and reductive NH4ã ¤ formation. Cavitation nuclei (talc) reduce the cavitation threshold and improve collapse reproducibility, while noble-gas doping modifies collapse temperature and shifts nitrate-to-nitrite ratios. Frequency and acoustic pressure act along independent axes—collapse intensity versus bubble population density—providing a practical framework for sonochemical reactor optimization. Finally, 3D mapping of the acoustic pressure field enables energy accounting and yields an optimized oxidative energy cost of 3.75 MJ/mol fixed nitrogen.

4pPAc8. Pulse-envelope tuning of ultrasound to boost sonochemical yield: A case study with hydrogen production. Lucia Mullings (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom, lucia.mullings@balliol.ox.ac.uk) and James Kwan (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Pulsed ultrasound is commonly used in sonochemical reactors to manage heating and reduce energy usage. Yet the role of pulse-envelope shape in governing cavitation at high ultrasonic frequencies remains underexplored. We investigate how temporal shaping of acoustic pulses alters cavitation activity and energy-normalized hydrogen production. As an initial study, square-modulated pulses were compared with amplitude-modulated up-ramped pulses of equal duration in argon-saturated water at 527 kHz (4.5% duty cycle; 50-cycle pulses). Cavitation dynamics were assessed using broadband passive acoustic measurements (broadband emissions as a proxy for inertial cavitation), alongside hydrophone pressure characterization of the driving field. For the same drive voltage, the up-ramped excitation pulse produced $\sim$1.3-times higher peak acoustic pressure while maintaining a similar spatial pressure distribution. Despite this, broadband-noise analysis indicated a higher apparent cavitation threshold for the up-ramped waveform (with and without a nucleation agent). Under comparable excitation conditions, the up-ramped pulses yielded 2-times greater hydrogen production per unit electrical energy. These results highlight pulse-envelope shaping as a practical lever for controlling cavitation onset and improving sonochemical efficiency beyond conventional on/off pulsing.

4pPAc9. Theoretical and numerical modelling for acoustic characterization of sub-millimetric amplitude surface waves in fluids. Zhengwu Wang (Phys. and Astronomy, Univ. of MS, 145 Hill Dr., National Ctr. for Physical Acoust., Univ., MS 38677, zwang7@go.olemiss.edu), Zheguang Zou, Harrison Roth (Dept. of Phys. and Astronomy, Univ. of MS, Univ., MS), and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, Univ., MS)

Acoustic measurement of free-surface wave amplitudes has emerged in recent years as an effective and cost-efficient alternative to optical methods. The general approach is to recover the surface elevation from Doppler-induced sidebands in the received frequency spectrum. However, existing

idealized theoretical frameworks impose strict constraints on acoustic transducer and receiver configurations, thus limiting experimental accessibility and accuracy. In this study, we present a two-dimensional model that acoustically determines the amplitudes of two oppositely propagating water surface waves. The model remains accurate for general transducer configurations and does not require a prescribed transducer–receiver angle. We further validate the model with integral-based numerical calculations of the receiver pressure spectrum, which contain the Doppler sidebands generated by two prescribed counter-propagating surface waves, demonstrating accurate recovery of their amplitude ratio. This model also enables systematic studies of transducer directivity effects and supports a broader range of experimental configurations. [Work supported by the National Science Foundation under Grant No. 2306106.]

4pPAc10. Ultrasonic measurement of multi-directional, submillimetric amplitude capillary-gravity waves. Harrison Roth (Phys. and Astronomy, Univ. of MS, 1069 National Ctr. for Physical Acoust., 145 Hill Dr., Univ., MS 38677, hmroth@go.olemiss.edu), Zhengwu Wang (Phys. and Astronomy, Univ. of MS, Univ., MS), and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, Univ., MS)

Measurements of surface waves in fluids have been studied extensively to meet practical application needs, but optical techniques, such as Laser Doppler Vibrometry and Schlieren imaging, typically require expensive instrumentation and impose strict constraints on experimental setup and equipment design. Acoustic methods can measure surface waves in fluids by demodulating sidebands that arise when an incident airborne ultrasonic wave is modulated by a propagating surface wave, allowing for the recovery of the surface wave frequency and amplitude. Whereas experimental studies of wave diffraction at rough surfaces have largely been confined to large-scale ocean-acoustic contexts between the seabed and sea surface. Here, we present a tabletop-scale acoustic experiment that enables precise measurements of millimetric and submillimetric capillary–gravity waves in controlled laboratory environments. The present technique allows for the determination of amplitudes and directions of two oppositely propagating capillary-gravity waves for arbitrary transceiver configurations. This low-cost acoustic method provides high-precision determination of surface-wave amplitude and implementation in controlled wave-obstacle interactions in tabletop experiments. [Work supported by the National Science Foundation under Grant No. 2306106.]

4pPAc11. Quantifying drought stress in trees using ultrasound. Neal Ahuja (Bala Cynwyd, PA), Lucas McGarvey (Eng., Swarthmore College, Swarthmore College, 500 College Ave., Swarthmore, PA 19081, lmcgarv1@swarthmore.edu), Jake J. Grossman (Biology and Environ. Studies, St. Olaf College, Northfield, MN), and E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA)

Drought can cause vascular embolism in trees and other woody plants, which depend on maintaining liquid-filled vessels in their xylem to transport water and nutrients from roots to leaves through trunk and branches. In optical embolism vulnerability analysis, biologists assess the level of drought stress by removing a branch and observing optical changes in the vessels as the branch dries out and air bubbles form. However, the possibility of a non-invasive instantaneous estimate of drought stress led us to employ 20 MHz pulsed ultrasound to interrogate the cross-section of $\sim$1 cm-diameter, branches for the presence of bubbles. The battery-powered Clarius L20HD3 provides a B-mode greyscale image that we have processed to quantify the percentage of pixels above a threshold as indicators of the greater reflection of ultrasound from gas-filled vessels over liquid-filled vessels. Our results corroborate findings from the optical method, and could serve as the basis for non-invasive assessment of drought tolerance in trees and other woody plants.

4pPAc12. Characterising scattered transient infrasonic wavefields using both array and single-sensor measures. Jack-Andrew Smith (School of GeoSciences, Univ. of Edinburgh, Edinburgh, United Kingdom), David N. Green (AWE Blacknest, AWE Blacknest, Brimpton RG7 4RS, United Kingdom, dgreen@blacknest.gov.uk), and Stuart E. Nippress (AWE Blacknest, Reading, United Kingdom)

Infrasonic scattering by fine-scale atmospheric sound speed structure impacts both the spatial and temporal wavefield characteristics recorded at ground-based sensor arrays. Array measures, including the structure of the sample covariance matrix, contain information regarding the directional energy distribution of the received wavefield. Single-sensor measures, including multi-scale entropy and spectral amplitude correlation, provide complementary information regarding the propagation regimes. We apply these techniques to recordings at array IS37, Norway, from repeated explosions in Hukkakero, Finland (at a distance of 321km). Results indicate that the energy of the received wavefield in slowness space is approximated by a skewed Gaussian distribution, for which the beamwidth increases with time. Entropy and spectral correlation measures identify a mixture of ballistic (weakly-scattered) and diffusive (strongly-scattered) arrivals within the infrasonic waveforms. We consider the implications of these results for how infrasound recordings may be used to probe the sound speed structure in the middle atmosphere.

Session 4pSA

Structural Acoustics and Vibration: Lessons Learned: Mistakes, Missteps, and Miscalculations in Structural Acoustics and Vibration

Colby W. Cushing, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Peter Kerrian, Cochair
ATA Eng. Inc., 13290 Evening Creek Dr. S, San Diego, CA 92128

Invited Papers

1:00

4pSA1. To err, or not to err, that is the quest eon. Alexey Titovich (Carderock Div., Naval Undersea Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700, alexey.titovich@navy.mil)

This talk will explore the delicate interplay of human curiosity, imperfection, fear, and perseverance within the demanding fields of acoustics and engineering. While we are often told that mistakes are vital for innovation and growth, they can also lead to catastrophic consequences. This presentation will identify common pitfalls in our field and, more importantly, introduce a simple yet highly effective methodology for systematically minimizing their occurrence. Finally, we will address a pressing and highly relevant question: can engineering accuracy survive in the age of AI-generated chaos, and what does the future hold for our profession?

1:20

4pSA2. The folly of thinking, 'This will be a quick, simple experiment'. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

Albert Einstein once said, "If we knew what we were doing, it would not be called research, would it?" I frequently use the equipment in my research lab to set up acoustics demonstrations for classroom use or to collect experimental evidence to support homework problems or lecture examples. Too many times, I have made the mistake of assuming "this will just be a simple, quick measurement," only to realize hours (or days) later that the data I was trying to collect was not turning out anything as expected. In this talk I will describe several "simple" acoustics experiments for which the results did not work nearly as well as I thought they would, or for which the results were surprisingly different than what was expected. Included in the examples will be an attempt to measure the input acoustic impedance of a Helmholtz resonator; the application of a constrained layer damping material which, instead of adding mass and damping to lower the frequency of a pickleball paddle raised the frequency instead; and attempts to measure the reflection of longitudinal waves in a thin rod.

1:40

4pSA3. The impact of bubbles on multiple inclusion unit cell testing in water. Amanda Hanford (Penn State Univ., P.O. Box 30, State College, PA 16802, ald227@psu.edu)

As part of an in-depth study into the characterization of effective material parameters of metamaterial solid inclusion unit cells with water as the background fluid, multi-inclusion unit cells were tested using non-destructive acoustic excitation techniques in in-water environment. For example, including a foam material that is less dense than the background fluid, results in an anisotropic density tensor with one component greater than and one component less than the value of the background fluid. The results of a parametric study determined that for a multi-inclusion unit cell, the effective material parameters can be controlled by the dimensions of the rigid inclusion as well as the material parameters and dimensions of the foam inclusions. Testing techniques were used to extract the material parameters of different grades of foam to identify the ideal grade for use in a multi-inclusion unit cell. Almost no sound energy was transmitted through the multi-inclusion metamaterial, contrary to expectation, because of the presence of air bubbles, both on the surface of the foam as well as potentially in between the inclusions. Additionally, the foam did not withstand hydrostatic pressures in the water tank and testing results for the multi-inclusion unit cells are inconclusive.

2:00

4pSA4. Challenges in quantifying the structural impacts of rocket noise. Makayle S. Kellison (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, makayle@byu.edu), Noah L. Pulsipher, Kent L. Gee, and Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The high-amplitude noise generated during rocket launches can have significant impacts on surrounding structures, wildlife, and nearby communities. While previous work has focused on understanding the physical mechanisms driving rocket noise generation, there has been a recent interest in the structural impacts, motivated by reports of windows shattering and foundations cracking during launch. Quantifying these impacts has introduced an entirely new set of challenges for the rocket noise community. Vibration measurements (both successful and failed) and lessons learned are presented for Falcon 9, Atlas V, New Glenn, Space Launch System, and Starship. Data were collected on a diverse set of structures, including homes, large warehouses, and a lighthouse, at distances ranging from 500 m to 15 km from the launchpad. Peak accelerations and displacements are discussed as well as the acoustic transmission loss through each type of structure. Results are placed in the context of other transportation noise sources, including trains and commercial aircraft. [Work supported, in part, by the National Science Foundation Graduate Research Fellowship.]

2:20–2:35 Break

2:35

4pSA5. Lessons learned in wind tunnel testing of phononic subsurfaces. Abigail T. Juhl (AFRL, 2941 Hobson Way, Bldg. 654 Rm. 347, Wright Patterson Air Force Base, OH 45433, abigail.juhl.1@us.af.mil), Vincent Chen (AV Inc./ AFRL, Beavercreek, OH), Carson Willey (AV Inc./ AFRL, Wright-Patterson AFB, OH), Caleb Barnes, Alberto Medina (AFRL, Wright Patterson AFB, OH), Greta Colford (Dept. of Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Burak Tuna, Rajan Kumar (Florida Ctr. for Adv. Aero-Propulsion, Tallahassee, FL), Theodoros Michelis, and Marios Kotsonis (Delft Univ. of Technol., Delft, Netherlands)

In 2015, phononic subsurfaces (PSubs) were introduced as a novel and promising methodology to achieve passive flow control by engineering the elastodynamic response of a phononic crystal to a forcing by a flow instability with the ultimate goal of delaying the transition to turbulence. The study of PSubs required an overlap of expertise in disparate disciplines including structural dynamics and fluid dynamics, requiring consideration of aerodynamicists and material scientists when characterizing the fluid structure interface at a wind tunnel. This project served as proof that mastering of a new field is an iterative process where mistakes serve as essential data points required for deeper understanding. This talk will step through lessons learned in the testing of PSubs during wind tunnel testing including updating initial computational fluid dynamics models, optimizing on-site measurement of PSub resonances, methods to improve correlating data inputs and outputs, changing experimental setups to meet unexpected findings, and even cleaning methods for unforeseen experimental discrepancies.

2:55

4pSA6. Listening to impacts on plates; Learning from mistakes. Allison M. King (Naval Architecture and Marine Eng., Univ. of Michigan, 2370 GG Brown, Ann Arbor, MI 48109, kingalli@umich.edu)

Matched field processing (MFP) offers a powerful framework for source localization, but practical performance is often limited by experimental, modeling, and implementation challenges. This talk synthesizes lessons learned across three MFP studies focused on impact localization on aluminum plates: infinite plate, finite plate, and a plate suspended at an air–water interface. These studies combine finite-element simulations and laboratory experiments on a 91.4 cm diameter aluminum plate with 6.4 mm thickness. Experimental data were collected using linear arrays of up to 14 sensors with 5 cm spacing, including microphones positioned 9.2 cm above the plate and hydrophones located 13.5 cm below it. Results across multiple impact locations, frequency bands, and grid resolutions indicate that successful MFP depends on integration of experiment design, signal processing, accurate modeling, and a variety of implementation details. The experimental setup strongly influences how well finite-element simulations match reality, dictating localization performance. The sensor array configuration and signal processing decisions, including preprocessing, filtering, and normalization, all reshape MFP ambiguity surfaces. The computational modeling effort is critical. Trade-offs between physical fidelity and computational cost emerge quickly. And finally, MFP implementation choices, such as grid construction and frequency band selection, directly affect ambiguity surface resolution and localization outcomes.

Contributed Papers

3:15

4pSA7. Measurement system effects in the characterization of COTS piezoelectric elements. Nathan Geib (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, geibnath@gmail.com), Brooklynn Petty, and Christina Naify (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Commercial off-the-shelf (COTS) piezoelectric elements are widely used in acoustical systems, but reported material properties often vary due to both part-to-part differences and measurement system effects. This work examines sources of variability in the characterization of COTS piezoelectric parts with simple geometry, with emphasis on separating true part variation from measurement system effects. Ten nominally identical parts were characterized using repeated measurements of geometry, mass, density, electrical impedance, and piezoelectric charge coefficient, d33. Measurements were performed by two operators with minimal procedural guidance to assess repeatability and reproducibility. Direct d33 measurements using a Berlincourt-type meter were compared with effective d33 values extracted from impedance measurements. Analysis of variance (ANOVA)–based Gage R&R methods were used to evaluate the relative contributions of part variation, operator effects, calibration, and test setup conditions. Results indicate that geometric, mass, and impedance measurements are largely insensitive to operator and setup variations. In contrast, direct d33 measurements exhibit substantial operator dependence and measurement-to-measurement variability comparable to part-to-part differences. Impedance-derived d33 values show strong correlation with direct measurements but include a systematic offset that requires further investigation.

3:30

4pSA8. Modes of America: Computational acoustics and the sound of the Liberty Bell. Jon Young (Structural Acoust., Penn State Univ., 243 Musser Ln., Bellefonte, PA 16823, jyoung.engr@gmail.com), Sean Collier (Graduate Program in Acoust., The Penn State Univ., University Park, PA), Aaron Stearns (Mech. Eng., Penn State Univ., State College, PA), and Eric Brown (Structural Acoust., Penn State Univ., State College, PA)

Have you ever wondered what the Liberty Bell sounded like ringing in the Pennsylvania State House in the year 1800? Computational acoustics might have the answer. Numerical solutions, such as those obtained from finite and boundary element methods, can be used to predict the noise radiated by a vibrating structure with accuracy. Several methods are discussed for different frequency ranges, with the modal reduction approach being the primary method used here for its efficiency. The pros and cons of using modes to represent vibratory responses is discussed, as well as how to couple these modes to a surrounding acoustic fluid. Additionally, different diagnostic and analysis tools will be used to identify dominant acoustic radiation patterns of the bell. These tools are then used to predict the sound of the Liberty Bell before and after it cracked. Lessons learned regarding model accuracy compared to experimental data, and difficulties in predicting the sound of the bell are discussed.

3:45

4pSA9. Structural-acoustic measurements with unexpectedly large damping. John B. Fahnline (ARL / Penn State, P.O. Box 30, State College, PA 16804-0030, jbf103@arl.psu.edu) and Aaron Stearns (Mech. Eng., Penn State Univ., State College, PA)

Correctly representing damping in structural-acoustic systems is difficult, especially if the goal is to directly compare experimental measurements and numerical simulations. Several practical examples are given with measured damping losses much larger than expected. First, a heavy bronze plate in air suspended from a crane using eyebolts is considered. Frictional damping due to the eyebolt threads resulted in large damping and was eventually eliminated by gluing the bolts in place. A second example is the acoustic field excited in air by a small speaker inside a heavy wooden box with a hole resting on the ground. Large damping levels were observed at the internal acoustic resonances and it is theorized that the pressure field becomes large enough at resonance to cause significant vibrations in the box and frictional damping. A third example is an aluminum plate resting on the air-water interface in a reverberation tank. The plate was suspended from a crane with fishing line to try to minimize damping as much possible. Measurements in air were used to specify modal loss factors for the structural finite element model of the plate alone. Suspending the plate at the air-water interface gives much higher damping than a corresponding structural-acoustic model.

THURSDAY AFTERNOON, 14 MAY 2026 FRANKLIN HALL A, 1:30 P.M. TO 5:00 P.M.

Session 4pSCa

Speech Communication: Cognition, Social Factors & Higher-Level Influences (Poster Session)

Cynthia G. Clopper, Chair
Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210

All posters will be on display from 1:30 p.m. to 5:00 p.m. Authors of odd-numbers papers will be at their posters from 1:30 p.m. to 3:15 p.m.and authors of even-numbered papers will be at their posters from 3:15 p.m. to 5:00 p.m.

Contributed Papers

4pSCa1. Contributions of phonological short-term memory and production variability to speech sound learning. Alexandra M. Kapadia (Linguist, Univ. of Chicago, 1115 E 58th St., Rm. 203, Chicago, IL 60637, alexandrakapadia@uchicago.edu), Shelby Bruun, Charlie Nagle (Dept. of Spanish & Portuguese, The Univ. of Texas at Austin, Austin, TX), and Melissa Baese-Berk (Linguist, Univ. of Chicago, Chicago, IL)

Language learners with similar language backgrounds (L1) who have received the same training still show significant variability in their trajectories of learning to perceive and produce novel speech sounds. Exactly which abilities inform individual differences in learners' sensitivity to novel phonetic cues and capacity to reproduce them remains unclear. Previous work has suggested that learners with stronger phonological short-term memory have an advantage in their identification of unfamiliar speech sounds and sensitivity to novel phonetic cues, and individual differences in motor control have been linked with articulatory skills and flexibility, abilities which may support learning novel speech sounds. In this study (part of a multi-week training paradigm), native English speakers were trained on an unfamiliar stop contrast from either Thai or Korean. As in English, both sets of contrasts are cued by voice onset time, with the Korean contrast additionally cued by fundamental frequency. Before beginning training, subjects completed individual differences tasks to assess phonological short-term memory, L1 production variability, and phonetic imitation ability. This analysis will present baseline discrimination and production of the target contrast in relation to performance on individual differences tasks, illustrating the role of these skills in contributing to the acquisition of novel speech sounds.

4pSCa2. Comparing human and nonhuman judgments of perceived similarity of gender diverse talkers. Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., 1579 S Renwick Blvd., Bloomington, IN 47401, mjhenry5519@gmail.com) and Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Listeners conduct fine-grained analyses of gender from speech, and their perceptual organization of voices is presumed to be grounded in structured acoustic-phonetic information. Here, we compare human perceived similarity to automated self-supervised speech representations from a deep-learning model (i.e., HuBERT), which provides pairwise distance scores that

increase with divergence in acoustic representations. Comparing human similarity judgments with HuBERT-derived distances tests the extent to which listeners' perceptual organization is driven by acoustic representations versus socially- or task-driven constraints. Stimuli consisted of one sentence produced by 20 talkers representing five gender identities (cisgender man, cisgender woman, transgender man, transgender woman, nonbinary). In a free classification task, listeners (29 cisgender and 29 gender diverse) first grouped talkers by general similarity and then by perceived gender identity. Across all listeners and tasks, higher perceived similarity was associated with smaller HuBERT distances. This relationship was stronger in the unconstrained than the constrained task. Reliable alignment between human and nonhuman similarity metrics suggests that holistic acoustic distance partially drives how listeners classify talkers. However, human listeners also likely draw on knowledge of social constructs that automated similarity algorithms fail to capture, especially when the social category is invoked by task instructions.

4pSCa3. Abstract withdrawn.

4pSCa4. "Tobby", "swabwee", and "strawberry": Speech productions by toddlers in a linguistically-diverse community. Emily Shroads (Psych., Univ. of Toronto Mississauga, CCT 4110, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, emily.shroads@mail.utoronto.ca), Sophia MacKeigan, Priscilla Fung, and Elizabeth K. Johnson (Psych., Univ. of Toronto Mississauga, Mississauga, ON, Canada)

While there is growing awareness of the need to account for bilingualism in clinical assessments, there is a dearth of work on speech development adequately considering how input diversity, including bilingualism and accent variation, might affect toddlers' productions. We recorded 2- and 3-year-olds with varied profiles of language input: (1) monolingual single-variety (Canadian English), (2) monolingual multiple-variety (additional English dialects/accents), and (3) bilingual. Target words were picture-elicited and spontaneously-produced. In ongoing complementary studies, we are investigating whether input diversity helps account for extensive variation across children in early productive patterns and in their degree of intelligibility to adult listeners. Thus far, phonetic transcription demonstrates speech sound distortions, substitutions, and omissions common to young children, and interestingly, several reflections of patterns uncommonly-reported in more linguistically-homogeneous samples (e.g., backing, e.g. [k^h] for /t/; frication, [v] for /b/; substantial distortions of vowels, [o] for /ɑ/). In an open-set transcription task presenting this corpus to Canadian English-speaking adults, we evaluate the hypothesis that deviations from canonical forms produced by monolingual single-variety toddlers are more predictable and interpretable to listeners than those by toddlers with varied input. Results will yield novel insights into speech development in diverse populations, with both theoretical and clinical implications.

4pSCa5. Exploring listener attributions of talker gender in three-way forced choice tasks and gender diverse voices. Line Lloy (Commun. Sci. and Disord., New York Univ., 261 23rd St., Unit 2, Brooklyn, NY 11215, mal9792@nyu.edu) and Molly Babel (Linguist, Univ. of BC, Vancouver, BC, Canada)

Listeners implicitly and automatically attribute genders to the voices they hear. While in the past it has been shown that these attributions often match with talkers' gender identities, older work has often been limited to binary cisgender male and female speakers, with binary options for listener attributions of talker identity. Recent work has begun to explore how different listeners attribute gender diverse (transgender and non-binary) individuals, and how there may be variability within how listeners attribute gendered features such as femininity and masculinity to voices within a given gender category. The current work analyses results from two studies which include gender diverse talkers and listeners, and explores the impacts of two design choices: (1) the inclusion of a three-way forced choice categorization task, with the labels "man", "woman" and "non-binary", and (2) the impact of other voices and the order in which they are presented. Altogether, the results of these studies have social, clinical, and theoretical implications for gender diverse individuals, for whom these decisions have important impacts on their well-being.

4pSCa6. Interdental tongue protrusion in California English. Keith Johnson (UC Berkeley, Univ. of California, Berkeley, Dept. of Linguist, Berkeley, CA 94720, keithjohnson@berkeley.edu) and Alexandra Pfiffner (UC Berkeley, Berkeley, CA)

It has been claimed that the pronunciation of the interdental fricatives [θ] and [ð] has tongue protrusion in California English while British English has no protrusion. This paper examines tongue protrusion in movies of hour-long sociolinguistic interviews conducted in Oakland, California. We isolated video frames from the fricative midpoint and judged whether the tongue was visibly protruding between the teeth in 2160 tokens produced by 26 talkers. Tabulating the counts of visible tongue protrusion by talker we find that talkers reliably differed from each other ($X^2(25) = 409$, $p < 0.01$). Four talkers had no tongue protrusion in interdentals, and four had over 30% of tokens with tongue protrusion (one with over 65%). Tabulating the counts by word (considering the top 25 most frequent words) we find that words reliably differed from each other to a degree ($X^2(24) = 57.6$, $p < 0.01$). Words with [θ] in a cluster with /r/ like "three" and "through" were much less likely to have tongue protrusion. Acoustic analysis is in progress and the paper will report the results of our attempt to find any reliable acoustic evidence of tongue protrusion in interdental fricatives.

4pSCa7. Time-continuous rating of emotional valence reveals flexible temporal weighting of acoustic and lexico-semantic features. Jeffrey Xing (Psych., Univ. of California San Diego, 5215 Fiore Terrace, A106, San Diego, CA 92122, j8xing@ucsd.edu), Kate Urrutia (Psych., Univ. of California San Diego, La Jolla, CA), Benjamin Lang (Linguist, Univ. of California, San Diego, La Jolla, CA), Lauren Ostrowski (Neurosciences Graduate Program, Medical Scientist Training Program, Univ. of California San Diego, La Jolla, CA), and Tim Gentner (Psych., Univ. of California San Diego, La Jolla, CA)

Emotion perception in speech supports speech comprehension and depends on listeners' integration of acoustic and lexico-semantic cues. The computational principles underlying this integration remain unclear. Here, we model the perceived speaker emotion as an adaptive internal state that updates continuously during speech perception, revealing temporally evolving weights on acoustic and lexico-semantic features. We presented 90 natural conversational speech excerpts (30–90 s each) from the Buckeye Corpus (Pitt *et al*. 2007) to 173 listeners, who rated speaker emotional valence continuously over time from most positive to most negative. Participants also rated excerpts without lexico-semantic cues (via low-pass filtering) and without acoustic cues (via reading time-locked transcripts). We find that time-resolved emotional valence ratings for some excerpts are more affected

by removing lexico-semantic cues, and others are more affected by removing acoustic cues, implying that listeners flexibly utilize different features to infer speaker emotion at any given moment. We then explicitly map the temporal relationship between acoustic/lexico-semantic features and emotional valence ratings by fitting time-lagged random forest regressors across a range of lag values, identifying the relative weighting between features that drive perceived speaker emotion and resolving the timing of their integration during speech perception.

4pSCa8. Top down lexical effect on dichotic fusion of vowels in CVC words. Michael Bennie (Univ. of Florida, 5950 sw 151 ct, Gainesville, FL 33193, michaelbennie@ufl.edu), Michelle Molis, and Lina A. Reiss (Oregon Health and Sci. Univ., Portland, OR)

Dichotic vowel fusion occurs when two different vowels presented to opposite ears are heard as a single vowel, that is often a spectral average of the two presented vowels. The effect of lexical frequency on fusion frequency has not been studied previously. In the current study, we tested lexical influences on dichotic vowel fusion into /ɛ/ using pairs of CVC words with the vowels /ɪ/ and /æ/ split across ears. Lexical frequency was manipulated by choosing four consonant contexts: (1) the presented words (containing /ɪ/ and /æ/) were common and the candidate word (containing /ɛ/) was rare; (2) the presented words were rare and the candidate word was common; (3) all three words were common; (4) all three words were rare. Nine normal-hearing adults were tested. Upon dichotic word presentation, participants could choose from the three word options (containing /ɪ/, /ɛ/, or /æ/) and three counterpart vowel-only options to measure partial fusion percepts. Participants were allowed to choose up to three responses. Repeated measures ANOVA showed a significant effect of lexical condition on fusion rate into a single /ɛ/ percept, with higher fusion when all three words were all low lexical frequency ($F(3, 27) = 5.45$, $p < .01$). Three words were sometimes selected, indicating perceptual reconstruction beyond simple spectrum averaging. These results suggest a top-down lexical contribution to dichotic vowel fusion in addition to acoustic proximity. [Work supported by the ASA SUREIA program (Hearing Health Foundation) and NIH R01 DC022777.]

4pSCa9. Language or accent? Effects of beliefs on spontaneous phonetic imitation. Zeyu Feng (West Lafayette, IN), Joan Downey (Purdue Univ., West Lafayette, IN), Yuhyeon Seo (Psychol. Sci., SUNY Oswego, Oswego, NY), Alexis N. Zhou (Linguist, Purdue Univ., West Lafayette, IN), and Olga Dmitrieva (Purdue Univ., 640 Oval Dr. Stanley Coulter Hall, West Lafayette, IN 47907, odmitrie@purdue.edu)

The present study examines spontaneous phonetic accommodation toward speech presented as an unfamiliar language versus L2-accented speech during shadowing. Fifty-four native speakers of American English were assigned to either a Russian or a Russian-accented English condition. Participants completed an English reading pre-test, shadowed Russian-English near-homophones (e.g., Russian [kit] 'whale' as a near-homophone of English *kit*) produced by a native Russian speaker and presented as Russian words or as Russian-accented English words, depending on the group assignment, and then completed an English post-test reading task. Vowel formants and VOT were extracted. Formant measurements were normalized using the Bark Difference Metric; VOT analyses are ongoing. Results revealed significant convergence toward Russian vowels /o/ and /a/ during shadowing, with stronger effects when stimuli were presented as Russian rather than accented English. Convergence toward Russian /i/ and shorter voiceless VOT was observed only in the Russian condition. Convergence effects did generalize to post-test English productions. These findings suggest that phonetic imitation is not fully automatic: acoustically identical targets elicited weaker convergence when framed as accented L1 speech. Convergence was short-lived and segment-specific but emerged rapidly without explicit imitation instructions or social interaction, highlighting its potential role as an initial mechanism in sound change.

4pSCa10. Developmental changes in spontaneous descriptions of talkers from speech. Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, tbent@iu.edu), Rachael F. Holt, Ryan Hale (Speech and Hearing Sci., Ohio State Univ., Columbus, OH), Malachi Henry (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Anna Claire Ingram (Speech and Hearing Sci., Ohio State Univ., Columbus, OH), Rowan Kilgore (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Brianna Montgomery, Piper Reusch, Michaela Sershen (Speech and Hearing Sci., Ohio State Univ., Columbus, OH), Tzu Pei Tsai, and Amelia Xanders (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Speech is a central way individuals convey their identities. Listeners' sensitivity to some dimensions of identity through speech begins in infancy, while sensitivity to others develops into adolescence. However, most studies investigating how listeners link phonetic and socio-indexical information focus on single identity dimensions. Here, we evaluate how spontaneous descriptions of talkers change over development from 5 years of age to adulthood. Participants were presented with simple sentences and described the person talking. Responses were divided into categories: psychological, stimulus, sexual orientation and gender identity, physical, social, and other. Psychological and stimulus terms increased with age while gender descriptors decreased. The focus on social characteristics appeared stable from the early school age years to adulthood, although developmental differences were observed in the subcategories with more interpersonal term use for younger children and more focus on region of origin in adulthood. The proportion of physical appearance descriptors was stable in childhood with a decrease in adulthood. Children build representations of speakers from their voices that get more detailed with development. This change likely reflects cognitive development, including theory of mind, and exposure to larger social networks. [Work supported by National Science Foundation #SMA-2146474; Ohio State University Center for Cognitive and Brain Sciences.]

4pSCa11. Language context effects on English voice onset time during Stroop interference in sequential English–Spanish bilinguals. Kathleen Siren, Tepanta Fossett (Speech-Language-Hearing Sci., Loyola Univ. Maryland, Baltimore, MD), and Molly Suplee (Speech-Language-Hearing Sci., Loyola Univ. Maryland, 4501 N. Charles St., Baltimore, MD 21210, mdsuplee@loyola.edu)

Voice onset time (VOT) distinguishes English and Spanish voiceless stop consonants, with Spanish voiceless stop consonants typically exhibiting substantially shorter VOT than those in English. This study examined whether language context during a cognitive interference task modulates bilingual speakers' voiceless stop consonant production as measured by VOT. In the classic color-word Stroop task, participants are instructed to name the ink color while inhibiting the tendency to read the printed word (for example, the word BLUE printed in red ink requires the response "red.") In this investigation, eleven sequential English–Spanish bilingual speakers completed two computerized Stroop tasks. In one Stroop task, color words were presented in English and participants responded in English. In a second Stroop task, color words were presented in Spanish, though participants still responded in English. The response "purple" occurred eight times in each condition. Voice onset time for /p/ in purple was measured for each participant and averaged within condition. It was hypothesized that Spanish-language stimulus presentation would yield shorter VOT in English speakers' productions of "purple," reflecting increased activation of Spanish phonetic patterns despite an English output requirement. Findings inform models of bilingual speech production, language co-activation, and phonetic implementation under attentional load.

4pSCa12. A case study of Gramle performance: Insights into visible speech, pattern learning, and teaching acoustic cues. Stefan A. Frisch (Rehabilitation Sci., Appalachian State Univ., Leon Levine Hall, 1179 State Farm Rd., ASU Box 32165, Boone, NC 28608-2165, frischsa@appstate.edu)

Gramle (https://nascl.rc.nau.edu/gramle/) is a spectrogram reading game modeled after the spelling-based game Wordle. The player is presented with a spectrogram of a 5 sound word and a phonetic keyboard to enter and submit a guess of the word's transcription. Feedback on the guess is provided

as to whether each sound in the guess is in the word or in the word and in the guessed position. The player has up to 5 guesses to identify the transcription. The author, a professor of speech science, played Gramle most days for three years and recorded the number of tries and the correct transcription of the puzzle word. An analysis of roughly 900 Gramle attempts found the cues in visible speech combined with the restrictions of the game led to solving the puzzles relatively quickly (mean guesses 2.27, median 2, and just under 30% solved with the first guess). Improvement in performance over time suggests that idiosyncrasies of the speaker's acoustic cues were learned. Analysis of the speech sounds in the puzzles finds the most difficult sounds to identify were non-strident fricatives while strident fricatives, voiceless stops, rhoticity, and near corner vowels contributed to solving puzzles quickly.

4pSCa13. Spectral contrast and lexical knowledge influence speech perception independently. John Kingston (Linguist Dept., Univ. of Massachusetts, 650 N. Pleasant St. ILC 434, Amherst, MA 01003, jkingston000@gmail.com)

A central question addressed by many empirical studies of speech perception is: how far down does the listener's linguistic knowledge go in influencing their behavior? This study crosses manipulation of spectral contrast between a preceding sentence and the vowel in a following syllable with three properties of the listener's lexical knowledge: which vowel makes a word in that syllable, how frequent that word is, and how dense its lexical neighborhood is. All manipulations influenced listeners' categorization of the vowel, but spectral contrast did so independently from all three lexical properties. Their influences were also independent of each other. These findings support a model of speech perception in which an earlier, alternatively lower, but still measurable stage is only influenced by the listener's auditory experience, free of any influences from their linguistic knowledge. A number of surprising interactions with response time were also obtained.

4pSCa14. Effects of social information on fricative discrimination in an oddball detection task. Kyler B. Laycock (Linguist, Ohio State Univ., 1308 Carbone Dr., Columbus, OH 43224, laycock.21@osu.edu) and Cynthia G. Clopper (Linguist, Ohio State Univ., Columbus, OH)

Listeners integrate social and linguistic information in speech perception. In sociophonetic literature, it has been demonstrated that cues to gender systematically influence sibilant identification, shifting /s/-/ʃ/ boundaries even with identical acoustic signals. This study examined whether explicit instructions enable listeners to ignore social information and maintain attention to phonetic variation. Using an oddball detection paradigm, participants heard fricative-initial minimal pairs from an /s/-/ʃ/ continuum while viewing faces and hearing voices independently indexing gender. In each trial, either the onset fricative, voice gender, or face gender changed. Participants were instructed to respond only to lexical changes (fricative differences) and ignore face or voice changes. Analyses assessed perceptual sensitivity and response bias across oddball types and face-voice congruity. When face and voice were constant, fricative discrimination was impacted by face gender. More of the continuum was perceived as /s/ when paired with a male face, and /ʃ/ with a female face. In gender change trials, false alarms increased when gender cues biased ambiguous fricatives in either direction. Discrimination accuracy remained high regardless of face-voice congruity, though increased response times for incongruent pairings suggest residual processing costs. Results indicate that social information influences perception even under explicit instructions to ignore it.

4pSCa15. Popular metrics find phantom convergence: A case for controls in laboratory shadowing. Ivy Hauser (Linguist, Univ. of Texas Arlington, 701 Planetarium Pl., Box 19559 — 132 Hammond Hall, Arlington, TX 76019, ivy.hauser@uta.edu), Marie Bissell (Univ. of Texas at Arlington, Arlington, TX), and Melissa Baese-Berk (Linguist, Univ. of Chicago, Chicago, IL)

Laboratory shadowing is a methodology for investigating the perception-production link that examines how participants shift their speech production after exposure to a model talker. A common metric used in these studies has been difference-in-distance (DID), which calculates the difference between participant-model distance at baseline versus post-exposure (Babel, 2012). Recent work has shown that DID is susceptible to a starting distance bias (and other problems), such that speakers with higher baseline distance from the model are found to converge more (Cohen Priva and Sanker, 2019; MacLeod, 2021). These authors have proposed an alternative method, linear combination (LC), that accounts for starting distance. We compared DID and LC in a laboratory shadowing study with an exposure condition and a no exposure condition. In the baseline block, participants produced a word list including 80 English /s/-initial words. In the exposure block, they either listened to a model voice reading the word list or sat in silence for the same duration. In the post-exposure block, they produced the word list again. Participants in our no exposure condition were found to have significantly converged to model speech they had never heard under both DID and LC analyses. This "phantom" convergence is a problem for shadowing studies that only use these metrics. This can be mitigated by including a control condition with no exposure to model speech and comparing degree of convergence in the exposure (test) condition relative to no exposure (control).

4pSCa16. Quantifying exposure to regional dialect variation in American English: General and specific scores. Marie Bissell (Univ. of Texas at Arlington, Arlington, TX), Larisa Bryan (Penn State Univ., State College, PA), Declan Chandler-Holtz, Kyler B. Laycock, Kevin D. Lilley (Ohio State Univ., Columbus, OH), and Cynthia G. Clopper (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Lifetime exposure to dialect variation affects lexical processing of and perceptual adaptation to familiar and unfamiliar dialects. This dialect exposure is often assessed using participants' self-reported residential histories. These rich residential history data are then typically reduced to binary distinctions to compare local to non-local listeners, with respect to the specific stimulus dialect(s), and mobile to non-mobile listeners, with respect to general dialect exposure. To address more nuanced questions about the effects of lifetime dialect exposure on lexical processing, we have developed metrics for quantifying specific and general dialect exposure scores. We use participants' residential and travel histories, the places of origin of their close family and roommates, and their self-reported regional and dialect identities to generate specific dialect exposure scores for each of the major regional dialects of American English, plus international Englishes. We then use multidimensional scaling to reduce these specific dialect scores to a single general dialect exposure score. These scores predict performance in a range of lexical processing and perceptual adaptation tasks involving familiar and unfamiliar dialects, suggesting that the scores capture essential variation between participants in specific exposure to particular dialects as well as general exposure to regional dialect variation throughout the lifespan.

4pSCa17. What [ju] know about that?: Changing patterns of glide deletion in memphis African American English. Nikolai A. Schwarz-Acosta (Spanish and Portuguese, Univ. of California – Berkeley, 5317 Dwinelle Hall, Berkeley, CA 94720, niko.andres@berkeley.edu), Rhosean Asmah (Linguist, Univ. of California, Berkeley, Berkeley, CA), and Nicole Holliday (Linguist, Univ. of California, Berkeley, Pasadena, CA)

Glide deletion occurs in several English varieties: speakers delete /j/ before /u/, pronouncing words like tune [tun] rather than [tjun], a change from the historical pattern (Phillips, 1994; Pabst, 2022). Deletion may vary according to social and phonological factors, but the phenomenon remains underdescribed (Phillips, 1994; Pabst, 2022). Our study betters understanding of English variation by exploring glide deletion among speakers of African American English in Memphis, Tennessee. 29 Black AAE speakers (women = 23), ages 21–74, read a wordlist with /ju/ words. Recordings were aligned and segmented using the MFA, with subsequent vowel extraction and delta F normalization (Johnson, 2020). We measured F2 maximum-F2 minimum and F2 trajectory, then conducted linear mixed effects regression models and GAMMs with gender, neighborhood, and age (continuous) as independent variables. Model results indicate F2 max-F2 min is significantly lower for men than women ($p < 0.05$), indicating more monophthongal productions of historical /ju/. F2 trajectory differs by neighborhood, with East Memphis residents demonstrating less F2 movement than others ($p < 0.05$). Additionally, older speakers have greater F2 movement

than younger speakers. Altogether, results indicate a change in progress, with younger speakers, women, and non-East Memphis residents leading the change.

4pSCa18. Hey Alexa, are you glitching on me?: Investigating phonological adaptation and generalization across human and device guises. Nicholas Aoki (Lingustics, Univ. of California, Davis, 469 Kerr Hall, One Shields Ave., Davis, CA 95616, nbaoki@ucdavis.edu) and Georgia Zellou (Lingustics, Univ. of California, Davis, Davis, CA)

Listeners generalize phonological shifts across place of articulation. For instance, exposure to a shorter-than-average VOT in a /t/-biasing context (e.g., "fron?ier"), results in greater /t/ responses along a /d/-/t/ continuum and more /p/ responses along a /b/-/p/ continuum. However, will generalization still occur even if the speaker is a digital assistant? Since technology can "glitch," a shifted stimulus might be perceived as an "error," thus blocking generalization. The current study exposed L1-English listeners to either canonical /p/ ("panther"), shifted /p/ ("banther"), canonical /t/ ("tissue"), or shifted /t/ ("dissue"). The test phase then asked participants to categorize stimuli along a "buy-pie" continuum. Critically, everyone heard the same female synthetic voice across both phases, but saw one of two visual guises as a manipulation of apparent talker identity (a silhouette of a digital assistant or a human). There were two main findings. First, replicating prior work, phonological generalization was observed – listeners exposed to both types of shifted stimuli ("banther" or "dissue") provided an equally higher proportion of "pie" responses than listeners exposed to canonical stimuli. Second, contra our initial hypothesis, generalization was equivalent across human and device guises, suggesting that listeners possess similar linguistic expectations for humans and devices.

4pSCa19. Acoustic phonetics of /u/ in earlier Baltimore English. Margaret E. Renwick (Dept. of Cognit. Sci., Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218, mrenwick@jhu.edu), Aidan J. Malanoski (Linguist Program, CUNY Graduate Ctr., New York, NY), Yasmin Roach, Kayla Boateng, Shaily Mistry, and Mashal Nawabi (Johns Hopkins Univ., Baltimore, MD)

English spoken in Baltimore, Maryland is known today to exhibit a phonetically fronted /u/, similar to Philadelphia and the southeastern USA. Unlike in other cities, /u/-fronting in Baltimore extends to the Black community. What is the historic development of /u/-fronting in Baltimore, and how has the vowel changed in apparent time? We analyze F1 and F2 of /u/, including its nucleus position and dynamic trajectories, in spontaneous speech. We consider its characteristics in post-coronal contexts which coarticulatorily favor fronting, as well as non-post-coronal contexts. Speakers hail from predominantly White working-class Hampden, as well as historical Black neighborhoods East Baltimore and Old West Baltimore. Data are provided by the Baltimore Neighborhood Heritage Project, a collection of 232 oral history interviews with Baltimoreans born 1879–1951. Results indicate that post-coronal /u/-fronting is an earlier "Baltimore universal", while /ul/ is backed; but although Black speakers' /u/ remains backed after non-post-coronal consonants, some White speakers front /u/ in that context. We interpret these findings in light of later 20th Century patterns, for instance that /u/-fronting has continued to advance across phonological contexts for White speakers, while it is primarily restricted to post-coronal contexts for Black Baltimoreans.

4pSCa20. Phonetic convergence as a function of different distances. Jennifer Pardo (Psych., Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardoj@montclair.edu)

Variability in phonetic form is a core phenomenon for any account of speech perception or speech production. Speakers accommodate their utterances to a variety of addressees, often resulting in increased between-talker similarity, or phonetic convergence. The current project examines the extent that patterns of phonetic convergence are modulated by between-talker acoustic distance and within-talker variability in acoustic-phonetic form. One prominent account predicts that phonetic convergence should be greater among talkers who are already similar to each other, while others find that phonetic convergence is greater among dissimilar talkers. The present study examines a previously-reported dataset to model acoustic convergence without calculating the difference-in-distance measure. In these new analyses, greater between-talker distances were associated with greater acoustic convergence in some measures, but not in others, and only when accounting for post-exposure distances in the models. In analogous assessments, both pre- and post-exposure acoustic distances predicted patterns of phonetic convergence obtained in holistic AXB perceptual similarity tests. The relative prominence of between-talker versus within-talker variability on patterns of phonetic convergence has important implications for understanding relations between speech perception and production.

Session 4pSCb

Speech Communication: Bilingualism & L2 Learning (Poster Session)

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All posters will be on display from 1:30 p.m. to 5:00 p.m. Authors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:15 p.m. and authors of even-numbered posters will be on display from 3:15 to 5:00 p.m.

Contributed Papers

4pSCb1. Perception of Mandarin-accented /s/-/θ/ among L1-Mandarin learners of English. Jacob Bower (Los Angeles, CA), Yuting Gu (Univ. of California Irvine, Social Sci. Plaza B, Irvine, CA 92697, yuting.gu@uci.edu), Shawn N. Cummings (Univ. of California Irvine, Newport beach, CA), and Xin Xie (Univ. of California Irvine, Irvine, CA)

L1-English listeners often have difficulty distinguishing between /s/ and /θ/ in Mandarin-accented English. It remains unclear whether this difficulty arises because L1-Mandarin speakers fail to produce an acoustically discriminable contrast, or because they *do* contrast the sounds but in a way L1-English listeners are not sensitive to. The latter constitutes a case where communication fails due to a mismatch between the speaker and the listener, and where intelligibility is expected to improve once this mismatch is resolved, as predicted by the "interlanguage speech intelligibility benefit" (Bent & Bradlow, 2003). In this study, we assess L1-Mandarin listeners' perception of Mandarin-accented /s/ and /θ/. Participants complete a forced-choice task by identifying accented /s/ and /θ/ targets ("did you hear /s/ or /θ/?"), embedded in both words (e.g., anthem) and nonwords (e.g., ansropology). If L1-Mandarin listeners successfully discriminate between the sounds, it suggests that the /s/-/θ/ distinction is maintained in Mandarin-accented English. This finding would point to systematic mismatches in phonetic category representations across L1 and L2 accents, raising new questions about how listeners may accommodate this difficult contrast in perception. References: Bent, T., & Bradlow, A. R. (2003). The interlanguage speech intelligibility benefit. *The Journal of the Acoustical Society of America*, **114**(3), 1600–1610.

4pSCb2. Disentangling lexical decision criterion shifts from phonetic category changes in natural accent adaptation. Yuting Gu (Univ. of California, Irvine, Social Sci. Plaza B, Irvine, CA 92697, yuting.gu@uci.edu), Shawn N. Cummings (Univ. of California, Irvine, Newport beach, CA), Michelle Chao, and Xin Xie (Univ. of California, Irvine, Irvine, CA)

Listeners readily accommodate natural accent variation even when it deviates from their expectations. For example, American listeners endorse Mandarin-accented /θ/ (often produced as /s/, e.g., *anthem*→ *ansem*) after talker-specific exposure. Prior work debates the locus of perceptual flexibility: should this adaptation be attributable to lexical decision criterion shifts (H1), or updated category representations (H2)? Because previous studies typically employ a single task that is either biased towards lexical (e.g., word/nonword judgements) or prelexical (e.g., category identification) processing, it is difficult to make mechanistic conclusions unconfounded by task demands. To disentangle task demands from accent exposure effects, we query both lexical decision [LD] and category identification [CatID] of Mandarin-accented speech from the same listeners. Between subjects, we manipulate task order and contrast-biasing condition (/θ/-biased [e.g., *thief*] vs. /s/-biased [e.g., *sudden*]). Adaptation, signalled by a difference between biasing conditions, is expected in both task orders. Critically, if adaptation is driven purely by bottom-up changes in category representations (H2), no task order effect is expected. In contrast, if adaptation is mediated by a lexical decision criterion shift (H1), then we expect larger adaptation effects in categorization when it is preceded by LD tasks.

4pSCb3. Do automatic speech recognition systems learn like humans? A comparison of semantic predictability effects in second-language speech adaptation. Victoria Xiaohan Wen (Univ. of Chicago, 1000 S Michigan Ave., Chicago, IL 60605, wen7@uchicago.edu) and Melissa Baese-Berk (Linguist, Univ. of Chicago, Chicago, IL)

Human listeners adapt to second-language (L2) speech rapidly, benefiting from semantic predictability during comprehension. While studies in automatic speech recognition (ASR) models have focused on capability of handling diverse accents on phonetic level, it remains unclear whether the models exhibit human-like sensitivity to semantic context. The present study compares the semantic predictability effect in human listeners and a large-scale ASR system. Existing human data from Baese-Berk *et al.* (2021) include listeners trained on L2 English sentences with high-predictability (HP), low-predictability (LP), or anomalous (ANOM), or no training. Results showed semantic predictability effects (HP > LP > ANOM), training-related benefits over the untrained control group, and adaptation specificity on the sentence type listeners were trained on. To assess whether ASR models have similar adaptations, we evaluated Whisper (OpenAI), a transformer-based ASR system. We used in-context learning (ICL) to approximate human short-term learning and replicate the test phase. Preliminary analyses indicate that Whisper improves transcription following ICL exposure and shows semantic predictability effects comparable to humans (HP > LP > ANOM). Similar to human listeners, Whisper demonstrates adaptation specificity: performance is the highest when training and testing sentence types align, with the strongest matching effect emerging in anomalous test condition.

4pSCb4. The transfer of prior experience with pitch during tonal language learning. Carissa A. Diantoro (Linguist, Univ. of Oregon, 1451 Onyx St., Eugene, OR 97403, carissad@uoregon.edu) and Melissa Baese-Berk (Linguist, Univ. of Chicago, Chicago, IL)

Studies in second language (L2) acquisition suggest that when the target L2 contains features similar to the learner's first language (L1), it is generally easier to learn that L2 than if they share minimal similarities. For example, it is easier for native Mandarin speakers to learn a new tonal language than native English speakers because Mandarin uses pitch in a lexically contrastive way unlike English. The question is, does this advantage come from specific knowledge of tone contrasts or general pitch processing abilities? This study compares learner populations with different experiences with pitch: L2 tonal language learners, L2 learners of a nontonal language that uses pitch as a phonemic cue, and monolingual English speakers. Participants first did a nonlinguistic pitch processing task, then completed tone identification and categorization tasks in an unfamiliar tonal language before and after a tone perception training paradigm. Preliminary results with monolingual English speakers showed a large variation in tone

perception across individuals and a small correlation between performance on the nonlinguistic pitch processing and tone perception tasks. Further testing of the other groups will shed light onto how prior knowledge with pitch affects performance in tone perception tasks.

4pSCb5. Language learners' pronunciation preferences are modulated by the teacher's language background. Shannon Barrios (Linguist, Univ. of Utah, 255 S. Central Campus Dr., LNCO Bldg., Ste. 2300, Salt Lake City, UT 84112, s.barrios@utah.edu), Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT), and Alayo Tripp (Linguist, Univ. of Florida, Minneapolis, MN)

Beliefs about language users can impact the inferences language learners make about linguistic structure (e.g., Falkert, 2016; Ballard & Winke, 2017). Hayes-Harb *et al*. (2022) exposed participants to a pronunciation pattern associated with a talker identified as the "teacher" and another pattern with a "student" talker in a simulated language learning classroom. Later, participants indicated which of two utterances spoken by a new talker was "better", overwhelmingly preferring utterances that resembled the "teacher's" pronunciation. Barrios *et al*. (under review) replicated the preference for the "teacher's" pronunciation with a new accent, and further demonstrated that the degree of preference was reduced when a pre-exposure vignette characterized the teacher "unfavorably". Now, we examine "native speakerhood" attribution, measuring accent preference using the preference task (Exp1) and a new naming task (Exp2; data collection ongoing). Exp 1 shows a reduction in preference for the "teacher's" pronunciation when the "teacher" is characterized as a "nonnative" speaker, suggesting that learners' pronunciation preferences are modulated by beliefs about the teacher's language background. This research furthers our understanding of linguistic stereotyping and its impact on language teaching and learning, laying the foundation for experimental studies of the efficacy of interventions designed to dismantle accentism embedded in these contexts.

4pSCb6. Bilingual experience and auditory health literacy: Online comprehension of spoken medical instructions. Teneesha Young (Kingsborough Community College, 2001 Oriental Blvd., E329, Brooklyn, NY, teneesha.young36@login.cuny.edu), Anastasiia Myslyk (Sea Crest Nursing and Rehabilitation Ctr., Brooklyn, NY), Courtney Stathis, Ashwini Kadave (Kingsborough Community College, Brooklyn, NY), and Laura Spinu (Commun. and Performing Arts, City Univ. of New York – Kingsborough Community College, Brooklyn, NY)

Auditory health literacy (i.e., the ability to understand and act on spoken medical information) is critical for healthcare decision-making, particularly in telehealth and automated contexts. This study examines whether bilingual language experience is associated with improved auditory health literacy, focusing on comprehension and numeracy for spoken medical instructions. Participants complete an online task via Pavlovia in which they listen to instructions for medication dosages, schedules, and conditional constraints. After each recording, participants answer comprehension questions and complete numeracy judgments requiring accurate interpretation of quantities and intervals. These tasks demand speech perception, linguistic parsing, and short-term auditory memory. Language experience is assessed using questionnaires capturing age of acquisition, proficiency, and daily language use. Data collection is in progress, with a target sample of approximately 60 adult participants. We hypothesize that bilingual participants will demonstrate higher auditory health literacy with increased instructional complexity, reflected in greater comprehension accuracy and fewer dosing-related errors than monolinguals. Within the bilingual group, we predict that more frequent and balanced language use will be associated with stronger performance. Our project extends acoustically grounded research on bilingualism to applied health contexts and highlights the potential role of bilingual experience in supporting everyday medical communication.

4pSCb7. Orthographic effects on L2 speakers' imitation of voice onset time. Yuou Sheng (Dept. of Curriculum, Teaching, and Learning, Univ. of Toronto, 252 Bloor St. W, Toronto, ON M5S 1V6, Canada, yuou.sheng@mail.utoronto.ca) and Jessamyn Schertz (Dept. of Lang. Studies, Univ. of Toronto Mississauga, Mississauga, ON, Canada)

Orthographic information has been shown to inhibit phonetic imitation of non-canonical variants in second language (L2) speech, yet its role in within-category imitation remains unknown. At the same time, within-category imitation is phonetically selective: previous work has shown robust imitation of lengthened voice onset time (VOT), but weaker and more variable imitation toward shortened VOT. The present study explores the role of orthography in within-category VOT imitation in Mandarin-speaking learners of English. Participants completed an explicit imitation task involving 15 English words beginning with /p/, presented either with audio-only input or with simultaneous audio-plus-text input. Target stimuli were manipulated into three VOT conditions (canonical, shortened, lengthened), with VOT differing by 60 ms across conditions. Participants are expected to show greater imitation of lengthened than shortened VOT, replicating asymmetries reported in previous work. We expect the presence of orthography to inhibit imitation in both conditions, but for the orthographic effect to be particularly strong in the shortened VOT condition, due to the strong association between <p> and aspiration in English. These findings will build on our current understanding of the effects of orthography on imitation, as well as the extent to which imitation in the L2 differs from the L1.

4pSCb8. Blended or bordered? Vowel space patterns in Vietnamese–English bilingual speech. Anges Vu (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, angesvu@u.northwestern.edu) and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

This study compared Vietnamese (VIE) and English (ENG) vowel productions by adult Vietnamese "heritage" speakers (i.e., switched-dominance bilinguals with non-dominant/L1 VIE and dominant/L2 ENG). Both VIE and ENG feature large vowel inventories (10–11 vowels), of which six are shared. We asked whether the languages' shared vowels would reveal blending or separation within the F1 x F2 vowel space. Variation in VIE–ENG shared-vowel category distances revealed that while some sounds blended within the F1 x F2 vowel space, others formed separate categories for each language. Specifically, some vowels (e.g., /i/) were produced with little variability irrespective of language, whereas formant frequencies for other vowels (e.g., /u/) shifted significantly across languages. This variation in cross-linguistic segment mappings seems to be driven by the presence of non-shared vowels (e.g., VIE /ɯ/), such that the overall composition of the bilingual vowel space expands to accommodate both inventories. Overall, this finding suggests segment-specific L1–L2 mappings rather than a general inventory-wide principle. In contrast to the idea that L2 sounds must undergo "equivalence classification" before forming separate categories, we propose that heritage bilinguals may initially treat shared sounds from their L1 and L2 as categorically separate, only blending sounds into one category when deemed sufficiently "similar."

4pSCb9. Preliminary investigation of listening effort in bilingual speech recognition: Behavioral and neural evidence using the Children's English and Spanish speech recognition test. María Ángel, Abrahám Chávez (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, maryflah@illinois.edu)

This study examines bilingual speech recognition and listening effort using the Children's English and Spanish speech recognition (ChegSS) test. Although ChegSS is reliable for assessing word recognition across signal-to-noise ratios (SNRs) and proficiency levels, the cognitive and neural effort supporting bilingual speech processing remains less understood. This project incorporates behavioral and electroencephalography (EEG) measures to evaluate how language context affects listening effort, with ongoing data collection in bilingual and monolingual children and adults. Participants complete ChegSS under two fixed SNRs with English or Spanish targets presented in a two-talker background. Target and masker languages either matched or differed. Monolingual participants complete only English targets in English or Spanish maskers. Listening effort is assessed using accuracy,

response times, self-report, and wireless EEG. Vocabulary, proficiency, and attention measures are also collected. Preliminary data from bilingual adults (n = 12) showed reduced accuracy and increased effort under harder listening conditions. Accuracy was slightly higher in same language than cross language contexts, with marginally better performance for English targets. Response times increased with task difficulty. EEG analyses showed greater frontal alpha activity in same language conditions. These preliminary findings demonstrate sensitivity of ChegSS to language-context effects and motivate continued investigation of bilingual listening effort.

4pSCb10. Qualifying multilingualism: Connecting language background with speech production outcomes in Russian–English and Italian–German children. Evgeniya Maryutina (Speech-Language-Hearing Sci., CUNY Graduate Ctr., 365 Fifth Ave., New York, NY 10016, emaryutina@gradcenter.cuny.edu), Theresa Bloder (Catholic Univ. Eichstätt-Ingolstadt, Eichstätt, Austria), Genevieve Medina, Robin MacDonald, and Suzanne van der Feest (Speech-Language-Hearing Sci., CUNY Graduate Ctr., New York, NY)

Multilingualism exists on a wide spectrum, and assessing an individual's linguistic background and predicting language outcomes is challenging. To evaluate the predictive value of more precise language background questionnaires (LBQs) (e.g., Byers-Heinlein *et al.*, 2020) we investigate acoustic-phonetic outcomes in young simultaneous bilinguals and link these to their backgrounds assessed through our detailed LBQ (cf. Rinker *et al.*, 2017). We analyze elicited productions of (study 1) Russian vowels /i/ and /ɨ/ by 8- to 13-year-old Russian–English bilinguals, and (study 2) productions of voicing by 3- to 6-year-old German–Italian bilinguals. Both studies show differences in the realization of acoustic-phonetic cues of bilingual versus monolingual children. This may be indicative of motoric complexity involved in producing the formant-frequency height/movement of Russian /i/ or prevoicing; and/or can be interpreted as crosslinguistic sound-category interference from the majority/societal language (English; German) to the minority/heritage-language (Russian; Italian). Importantly, we find effects even in simultaneous bilinguals with daily heritage-language use and at least one native heritage-language parent. We argue that precise assessment of bilinguals' ongoing language use can serve as a predictor for acoustic-phonetic outcomes, and will discuss possibilities for further modernization and standardization of LBQs, specifically in terms of (multi)media use, motivation, and clinical settings.

4pSCb11. Production of Korean three-way laryngeal stop contrasts by late L2 learners. KYUNG EUN LEE (Speech-Language-Hearing Sci., The Graduate Ctr., City Univ. of New York, New York, NY) and Suzanne van der Feest (Speech-Language-Hearing Sci., CUNY Graduate Ctr., 365 Fifth Ave., New York, NY 10016, svanderfeest@gc.cuny.edu)

This study examines how English-dominant late L2 learners produce the Korean three-way stop voicing contrast (fortis, lenis, aspirated), cued by voice onset time (VOT) and onset fundamental frequency (F0). Participants were 15 Native Korean speakers (L1-NK) and 15 L2 learners of Korean literate in Hangul (L2-K). They completed a language background questionnaire and two production tasks conducted via Zoom: L1-NK completed sentence- and word-reading tasks; L2-K completed sentence-reading and imitation tasks. Stimuli included 27 monosyllabic target words (nine minimal triplets) balanced for place of articulation, plus fillers. A total of 2025 tokens (recorded by participants using iOS Voice Memos) were acoustically analyzed. Results show L2-K produced lenis stops with VOT values closer to fortis than aspirated stops, resembling production patterns seen in older Korean speakers. They showed minimal F0 differentiation, except for one heritage speaker whose productions aligned with L1-NK. Age of acquisition and length of exposure significantly predicted VOT accuracy, while length of residency did not. Early multilingual exposure appeared to shape variability. L2-K also showed improvement in lenis VOT during imitation, suggesting VOT is a more accessible cue than F0. Findings inform theories of L2 phonological representation and pedagogical approaches for teaching Korean voicing contrasts to L2 learners.

4pSCb12. Task and exposure effects on STRUT–PALM vowel production in Bangladeshi L2 English. Firoz Ahmed (Linguist, Univ. of Florida, Gainesville, Florida, Gainesville, FL 32603, firozahmed@ufl.edu) and Caroline Wiltshire (Linguist, Univ. of Florida, Gainesville, FL)

Bangladeshi L2 English learners are predicted to exhibit partial neutralization when pronouncing the English vowels /ʌ/ (STRUT, central, open-mid) and /ɑ/ (PALM, back, low), because their L1 Bangla has a single vowel nearby, central low /a/ (Payne and Maxwell, 2018; Rahman, 2018; Wiltshire, 2020). This study examines how speaking task and English exposure affect their STRUT–PALM contrast. Acoustic data were collected from two Bangladeshi L2 English speakers (ages 27–29) with differing English exposure (3 months versus over 2 years in the US). Two task types were analyzed: production of word-lists, and picture-based storytelling. Vowels from a total of 194 CVC tokens were analyzed. Vowel quality was quantified using F1–F2 measurements, Euclidean distance, Pillai scores, and linear discriminant analysis, and compared against Standard Southern British English (SSBE) and General American English (GA) reference norms. Results within each speaker's L2 vowel system showed STRUT–PALM with centralized realizations relative to English targets. STRUT consistently lies closer to Bangla /a/ than PALM, showing L1 transfer. Extended US English exposure shows minor STRUT fronting toward GA norms. Task effects are evident, with greater vowel separability in word-list production than in picture-based storytelling, demonstrating task-dependent modulation of L2 vowel contrasts.

4pSCb13. Hearing sensitivity and hearing-health awareness in English learner students. María Ángel (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign901 S Sixth St., Champaign, IL61820, mgangel2@illinois.edu) and Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Hearing health disparities among English as a second language (ESL) students are poorly understood, despite evidence that cultural, linguistic, and access-related factors influence hearing outcomes. Prior work identified unexpectedly higher low-frequency thresholds in Spanish-speaking ESL middle and high school students compared to monolingual peers, suggesting unrecognized hearing differences. The present study extended this work across a broader age range and multiple school districts to examine hearing sensitivity and hearing-health awareness. A total of 142 children (69 Spanish-speaking ESL; 73 monolingual English-speaking students) from three Illinois school districts completed tympanometry and pure-tone audiometry (250–8000 Hz) in quiet classroom settings; students eight years and older also completed a hearing-health knowledge and attitudes survey. ESL students demonstrated slightly poorer hearing sensitivity than monolingual peers, with consistent differences at low frequencies (250–500 Hz) across sites, averaging 5–6 dB, but remaining within normal clinical limits. Age was associated with threshold elevation, while ear and school site were not. Survey data indicated limited hearing-health awareness in both groups and did not explain threshold differences. Qualitative responses suggest linguistic and experiential differences in how hearing is conceptualized. Findings highlight small but consistent hearing-sensitivity differences and emphasize the need for culturally responsive school-based hearing-health initiatives.

4pSCb14. Bilingual flexibility in VOT imitiation. Erica Dagar (Linguist and TESOL, Univ. of Texas at Arlington, 701 S Nedderman Dr., Arlington, TX 76019, erica.dagar@uta.edu)

Language is dynamic and changes to reflect context. One of the ways this happens is through phonetic imitation, an unconscious cognitive process where speech shifts towards an interlocutor. There is conflicting evidence on whether imitation towards a phonological boundary occurs in voice onset time (VOT) imitation. This paper seeks to investigate if bilinguals of English and a true voicing language will imitate towards the phonological boundary due to their language exposure. The "monolingual" group participants (n = 10) consisted of English speakers. The "bilingual" group consisted of Spanish/English bilinguals (n = 10). The study using word-initial /p/ and /k/ words with a reduced VOT in English created by removing 40ms from the most stable part of aspiration. The procedure was a baseline

recording of the word list, a listening block where participants passively heard the model talker and a corresponding posttest. The VOT (ms) from the baseline and posttest were compared and the difference between the groups was analyzed. A Bayesian model with VOT as the dependent variable and experiment block and participant group as predictors found a statistically significant reduction in VOT from the baseline to posttest for the "bilingual" group only ($\beta = -0.90$, SE = 0.24, 95% CI [−1.37, −0.43]).

4pSCb15. Non-native clear speech: Investigating English vowel spaces in production and perception for native Mandarin speakers. Rebecca Gips (Linguist, Univ. of Pittsburgh, 4200 Fifth Ave., Pittsburgh, PA 15260, gobecca1029@gmail.com) and Melinda Fricke (Linguist, Univ. of Pittsburgh, Pittsburgh, PA)

In difficult communicative situations, speakers may produce "clear speech," which in English commonly includes expansion of the vowel space. However, while it is known that non-native speakers have a harder time implementing clear speech, it is not clear which acoustic characteristics are more difficult, or why this may be the case. The current study focused on vowel space expansion, investigating whether native Chinese speakers exhibit vowel space expansion in English when asked to produce clear speech. Preliminary results suggest that non-native speakers do not expand their vowel space to the extent that native speakers do. However, the question of why still remains. We used a perception task called the method of adjustment to measure participants' "ideal" perceptual spaces and compared these to their production spaces. While analyses are still underway, the extent of mismatch between each participant's perceptual space and their production space can help clarify the source of any difficulties in implementing vowel space expansion. If difficulties in expansion are primarily rooted in perception, then participants who show smaller clear speech adjustments should also show a large degree of perception-production mismatch; non-native speakers may "expand" their vowel spaces less because their ideal vowel targets are simply not the same as native speakers. If this relationship does not hold, however, it may suggest that difficulties in expanding the vowel space have their source primarily in motor planning or implementation.

4pSCb16. Unmarked versus marked onset clusters in L2 phonology: Evidence from perception and production. Alex Hong-Lun Yeung (SUNY Oswego, Wilber 234, Oswego, NY 13126, alex.yeung@oswego.edu)

English onset clusters differ in markedness based on sonority sequencing: clusters beginning with [s] + stop (e.g., [sk]) are considered marked, whereas stop + sonorant clusters (e.g., [pl]) are unmarked. Previous research on L2 onset clusters has often examined either perception or production in isolation, and L2 accuracy is typically evaluated in terms of overt repair strategies such as epenthesis or deletion. This study integrates perception and production and shifts the focus to acoustic timing in segmentally accurate productions, comparing English-L2 (e.g., Mandarin) with English-L1 speakers. Experiment 1 is a syllable-counting task assessing whether an additional syllable is perceived in English cluster-bearing words. Both Mandarin and English speakers show near-ceiling accuracy, indicating no perceptual illusion. Experiment 2 examines production of the same stimuli. Tokens were manually annotated, and onset duration was normalized by total word duration. English speakers show consistent timing across onset types, whereas Mandarin speakers, despite not producing an overt vowel, show systematic timing differences: [s] + stop clusters (mean = 0.59) and [s]+ sonorant clusters (mean = 0.56) are longer than other clusters (mean = 0.31) and simple onsets (mean = 0.32), based on the dataset analyzed to date. These findings suggest that markedness-driven structural representations influence gradient phonetic implementation in L2 speech even when perception and segmental accuracy are target-like.

THURSDAY AFTERNOON, 14 MAY 2026 SALON B, 1:30 P.M. TO 5:00 P.M.

Session 4pUW

Underwater Acoustics: Ocean Soundscapes and Underwater Noise

Vanessa M. ZoBell, Cochair
Biological Oceanogr., Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037

Samantha Juber, Cochair
Univ. of Washington, 1501 NE Boat St., #220, Seattle, WA 98195

4p THU. PM

Contributed Papers

1:30

4pUW1. Vertical-distribution of noise in the deep Philippine-Sea Basin during 2010–2011. Chandrasekhar Gaja (Ocean Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu, India), Tarun K Chandrayadula (Ocean Eng., Indian Inst. of Technol. Madras, Dept. of Ocean Eng., Indian Inst. of Technol., Madras, Chennai, Tamil Nadu 600036, India, tkchandr@iitm.ac.in), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

Acoustic systems to measure noise have either used a single hydrophone or vertical arrays that cover only a part of the ocean. The Philippine Sea 2010 experiment was however unique in recording noise across an entire column of water for depths 200 to 5400 m. The experiment that lasted for a year used 149 hydrophones recording sounds in the 10 to 488 Hz range, and arranged as five 1000 m subarrays. This paper presents statistics such as median noise-levels, and the frequency-wavenumber spectrum for all these depths. There were challenges in obtaining these statistics. The top subarray showed effects of strumming often saturating the low frequencies with high intensities. Fortunately, there were also comparatively quiet times when the strum did not dominate the spectrum. These occurred during the times the array was almost vertical, and not pulled down by currents. The analyses used these parts of the recordings to estimate the noise for the shallow depths, and contrast them with other subarrays. Apart from measurements of noise during the experiment, this paper also compares the noise-levels at

the deepest hydrophones to previous measurements at similar depths that happened in Philippine basin during a 10-day exercise in 1977 called Church Stroke II.

1:45

4pUW2. Investigating Northern Gulf soundscapes with glider volumetric array. Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, P.O. Box 43680, Dept. of Phys., UL Lafayette, Lafayette, LA 70504, nas@louisiana.edu)

Underwater gliders equipped with multi-channel passive acoustic monitoring systems, in addition to standard oceanographic sensors, are cost-efficient platforms for studying regional soundscapes. In 2022, 2023, and 2024 the University of Louisiana at Lafayette Slocum G3 glider Sedna, outfitted with a four-channel JASCO volumetric array, completed three fall-season autonomous surveys in the Mississippi Valley—DeSoto canyon region of the Northern Gulf. While the primary mission focused on evaluating the feasibility of using glider-based acoustic recordings to infer atmospheric wind forcing over the ocean surface, these deployments also yielded a rich, multi-use acoustic dataset of the regional soundscapes. This presentation demonstrates how a single glider mission can support a broad range of scientific inferences, including detection and localization of marine mammals, assessment of biological and anthropogenic soundscape contributors, and evaluation of geophysical survey impacts on ambient noise conditions. Beyond ecosystem monitoring, this advanced reconnaissance approach provides acoustic observations that may be integrated into operational ocean–atmosphere forecasting framework used by NOAA and the U.S. Navy, with the potential to improve air–sea interaction model parameterization and to enhance tropical cyclone intensity predictions. [Research supported in part by the Office of the Under Secretary of Defense for Research and Engineering, award# FA9550-21-1-0215.]

2:00

4pUW3. Abstract withdrawn.

2:15

4pUW4. Assessing the underwater impact of rocket noise in an estuarine environment. Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, thomas.jerome@arlut.utexas.edu), Colby W. Cushing, Robert T. Taylor, Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

With the expansion of space operations in south Texas, assessing the impact of noise is a growing environmental issue. Previous measurements of Starship Super Heavy launches from Starbase Texas resulted in unweighted in-air sound exposure levels in excess of 130 dB over 20 km from the launch site [Gee *et al.* JASA-EL **5**, 023602 (2025)]. Despite the extreme sound exposure expected from launches, little effort has been made to understand the impact of these intense acoustic events on nearby underwater ecosystems. This is of particular importance for operations at Starbase, which is located within the Boca Chica Wildlife Refuge with the Orbital Launch Pad on the shore of Boca Chica Bay on the Texas Gulf Coast. This presentation examines initial measurements of observable underwater sound levels in Boca Chica Bay from rocket noise events during the Spring of 2026. In addition to quantification of underwater sound exposure levels, potential behavioral impacts on local fauna are assessed via examination of the ambient soundscape in the estuary prior to and following events.

2:30

4pUW5. Preliminary modeling of tidal turbines in underwater environments. Mara Hirschmann (Phys. and Astronomy, Brigham Young Univ., N284 ESC, Provo, UT 84602, mhirschm@byu.edu) and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Energy is a pressing issue in modern society. Our need for energy drives both development and testing of new technology to meet industry standards and enhance the livelihood of people across the globe. A relatively new and effective manner of capturing energy are tidal turbines, which have the benefit that tides are a constant source of energy. Studying the effects of turbines, beyond simply their efficiency and effectiveness, teaches more about the overall benefits of tidal turbines. This research focuses on discovering more about the acoustic characteristics of tidal turbines and the resulting impact on surrounding underwater environments. Numerical methods utilizing the Ffowcs Williams-Hawkings equation simulate a tidal turbine in various scenarios, such as differing types of ground material, and their respective effects on the acoustic characteristics. This research lays the foundation for a better understanding of how tidal turbines will interact with the ocean environment, from an acoustic perspective. Through this improvement of our understanding, future implementations of tidal turbines can be done in a more environmentally responsible manner, enhancing their viability as truly renewable, reliable sources of energy. [Undergraduate research supported by Brigham Young University College of Computational, Physical, and Mathematical Sciences.]

2:45

4pUW6. Noise pollution in the Hudson river. Jiale Lin (LaGuardia Community College, Long Island City, NY), Justen Gallagher (LaGuardia Community College, Brooklyn, NY), Shenglan Yuan (LaGuardia Community College, Long Island City, NY), and Yun Ye (LaGuardia Community College, 31-10 Thomson Ave., Long Island City, NY 11101, yye@lagcc.cuny.edu)

Anthropogenic noise underwater has become a growing environmental concern in aquatic ecosystems, where boating traffic and other human activities overlap with aquatic habitats. This study examines sound levels recorded in the Hudson River at Pier 26 and Pier 97. Pier 26 is a high-traffic area, with nearby ferry routes, commercial shipping, and frequent recreational boating activity, while Pier 97 experiences comparatively lower commercial and recreational activity. We recorded underwater noise using an array of three hydrophones connected to a recorder. We then analyzed the data collected using customized MATLAB tools. Our analysis measured how loud the environment was over time and how often loud sound events occurred. Measured values were then compared with acoustic thresholds provided by the National Oceanic and Atmospheric (NOAA), for behavioral disturbance, temporary threshold shift (TTS), and onset of auditory injury. Findings indicate excessive exceedance of recommended thresholds at Pier 26, with major contributions from all three metrics examined: depth, current, and ship traffic.

3:00–3:15 Break

3:15

4pUW7. Ships tracking and their acoustic visibility zones on the shelf. Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), Denis Manulchev (Marine Geosciences, Univ. of Haifa, Haifa, Israel), Mark Shipton, and Roee Diamant (Marine Technologies, Univ. of Haifa, Haifa, Israel)

This paper examines anthropogenic acoustic noise, primarily of ship origin, recorded on the Mediterranean shelf near the port of Haifa. Acoustic measurements were conducted using four channel autonomous recorder, containing bottom hydrophones installed at a depth of 30 m. Three hydrophones were positioned at the vertices of a regular triangle and were equidistant 100 m from a central hydrophone. Based on the time delays of the acoustic wave front generated by passing ships arriving at various receiving elements, the relative geometry of the recorder placement was reconstructed. Solving the forward problem allowed us to reconstruct vessel trajectories using sparse automatic identification system data and the time difference of arrival method. In the next step, the acoustic visibility zones of passing ships were determined. For this goal, an acoustic wave energy loss function was constructed depending on the distance between the vessel and the receiving system, the direction of propagation relative to the coastline, and the vessel type (tankers, yachts, etc.). The relationship between the horizontal intensity distribution and the bathymetry of the region is demonstrated. [Work supported by ISF, grant 973/23.]

3:30

4pUW8. Characterization of hydroacoustic impacts of small-diameter timber and carbon fiber composite piledriving for use with regulatory assessments on the New England coastline. Adrian K. Doran (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, adrian.k.doran@usace.army.mil), Kyle G. Dunn (ERDC-CRREL, Hanover, NH), Heath Turner (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), T. Aran Mooney, Sierra Jarriel (Biology, Woods Hole Oceanographic Inst., Woods Hole, MA), and Naomi Handell (North Atlantic Div., U.S. Army Corps of Engineers, Brooklyn, NY)

The US Army Corps of Engineers is responsible for ensuring compliance with the Endangered Species Act during piledriving and other construction activities that generate hydroacoustic signals along the coastline of much of the United States. While a body of literature has been developed describing hydroacoustic sound pressure levels associated with large-scale activities (e.g., offshore wind turbine installation), very few data have been published that characterize the hydroacoustic impacts of small-diameter timber and carbon-fiber piles. Here we present data collected in a shallow harbor offshore Massachusetts during vibratory and impact piledriving of 10″ southern pine timber piles, 12″ southern pine timber piles, 12″ greenheart pine timber piles, and carbon fiber sheet piles. We present the distribution of 1-s = 10 dB between drives with consistent piles; in some cases, the variation was associated with observed changes in substrate conditions, and in others, long-period variations in broadband ambient noise levels biased individual measurements. Lastly, we report transmission loss observed during each piledriving condition measured between ranges of 1–65 m. < = 10 dB between drives with consistent piles; in some cases, the variation was associated with observed changes in substrate conditions, and in others, long-period variations in broadband ambient noise levels biased individual measurements. Lastly, we report transmission loss observed during each piledriving condition measured between ranges of 1–65 m.

3:45

4pUW9. Localized Maxwell-stress excitation of low-frequency modes of electrically conducting plates and shells in water. Christopher T. Powers (Phys. & Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, christopher.t.power@wsu.edu), Sterling M. Smith, Philip L. Marston (Phys. & Astronomy, Washington State Univ., Pullman, WA), Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Ahmad T. Abawi (HLS Res., San Diego, CA)

There is a specialized sonar system in development (by Smith and Marston) meant to locate targets based on exciting low-frequency (LF) vibrational modes in the target. To assist in the development of this method, an alternate method of exciting these LF modes was developed. Rather than driving the mode with modulated radiation pressure, the mode is driven electromagnetically by using a localized oscillatory magnetic field to excite eddy currents in the solid, electrically conducting target. These eddy currents cause oscillatory Maxwell stress on the target [Hefner and Marston, J. Acoust. Soc. Am. **106**, 3340–3347 (1999)]. Previously discussed was the use of an electromagnetic coil to generate the appropriate localized electromagnetic field on various generally cylindrically shaped targets both in a lake environment and in a lab tank environment. This work investigates the excitation of LF modes with further geometries of electrically conducting targets, such as plates and shells. Some target modes are even detectable in air. This investigation will look at both the frequency qualities of the modes as well as the shape of the modes, as recorded by scanning the relative coil position. [Work supported by the U.S. Office of Naval Research]

4:00

4pUW10. Near real-time passive acoustic monitoring and tracking of Pilot Whale pods with a coherent hydrophone array prior to visual confirmation. Nishanth Subramani (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., 409 DANA, Boston, MA 02115, subramani.ni@northeastern.edu), Arpita Ghosh, Saunak Samantray, Hrafn S. Sigurdarson, Max Rademacher, Matthew E. Schinault, and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Passive acoustic monitoring provides an effective means to detect and track marine activity at ranges beyond the limits of visual observation. A large-aperture coherent hydrophone array system, developed in-house at Northeastern University, was deployed from R/V Hugh R. Sharp in an experiment to investigate marine mammal presence and movement over instantaneous wide areas and in near real-time offshore Lewes, Delaware in September 2025. Array acoustic data acquired from a 96-element subaperture were sampled at 100 kHz for approximately 64 hours over five days, enabling monitoring of marine mammal vocal activity over extended time periods. Beamformed spectrogram images were generated in real-time at sea and applied to identify vocalization signals from multiple toothed whale and fish species. Distinct pilot whale tracks were observed in Baltimore and Wilmington Canyons, with vocalization detections occurring up to two hours prior to visual sightings. The array enabled signal-to-noise ratio enhancement and bearing estimation, enabling inference of pilot whale movement over multi-hour timescales, revealing spatial trends that preceded subsequent surface observations. Here we present the measured vocalization sequences and estimated spatial locations for a pod of at least 5 pilot whale individuals, with calls consisting of low-frequency whistles in the 1 to 2 kHz range, high-frequency whistles above 4 kHz, and broadband clicks.

4:15

4pUW11. Few shot bioacoustic event detection in freshwater soundscapes. Felicia M. Huffman (Comput. Sci., Michigan Technolog. Univ., P.O. Box 112, Hubbell, MI 49934, feliciafmh2@gmail.com), Evan Lucas (Michigan Technolog. Univ., Houghton, MI), and Rosalyn Putland (Ctr. for Environment Fisheries and Aquaculture Sci., Lowestoft, United Kingdom)

While marine soundscapes have been researched by many, freshwater soundscapes still hold many unknowns. These soundscapes are influenced by geophony, biophony, vessel traffic, and ice formation in colder climates. Prior work has indicated an increase in sound pressure with ice formation and vessel traffic, but researchers have noted it is difficult to extract and identify specific bioacoustic events which can be critical in understanding soniferous wildlife's response to environmental state and acoustic events. In this work, we contribute to these efforts by automating the process of bioacoustic event detection from limited examples, i.e., few-shot detection, using a minimal labeled dataset collected in Duluth, Minnesota as well as some unlabeled data collected in Houghton, Michigan. We establish detectability baselines using weather and traffic data aligned with our recordings, then apply machine learning methods to identify specific events based on the few labeled examples. This approach will allow us to work with biologists to study trends and gain a better understanding of the soundscape over time. By demonstrating the feasibility of using machine learning based methods in freshwater soundscape analysis and automating identification of unique bioacoustic noises, we help enable analysis of large passive acoustic monitoring for soniferous species.

4:30

4pUW12. Machine learning meets the ocean: Predicting ship noise for marine mammals protection. Soukaina Boujdi (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), 293 Rue Dubé, Rimouski, QC G5L 4W7, Canada, bous0383@uqar.ca), Loubna Benabbou (Departement de gestion, Université du Québec à Rimouski, Montreal, QC, Canada), and Pierre Cauchy (Institut des Sci. de la mer (ISMER), Université du Québec à Rimouski (UQAR), Rimouski, QC, Canada)

Underwater noise from ships is a growing pressure on marine mammals, affecting their ability to communicate, navigate, and interact. Existing noise prediction models have guided the field for decades but each has notable limitations. Ross relies on simplified log-linear relations derived from older datasets; RANDI uses empirical fits calibrated to limited ship classes; Wales–Heitmeyer reduces noise to a frequency-dependent relation; SONIC is based on tests conducted on a single reference vessel; and Wittekind requires numerous detailed input parameters that are often difficult to obtain. These constraints make these models less suited for modern vessels and dynamic ocean conditions. To address this gap, we develop a data-driven approach for predicting vessel noise using source levels collected through the Marine Acoustic Research Station (MARS). The dataset includes more than 2800 underwater recordings from 2021 to 2025 of ships transiting the St. Lawrence corridor in East Canada. We tested several ensemble methods, with CatBoost achieving a significatif performance and delivering the lowest MSE and RMSE. Feature-importance analysis shows that vessel speed and age are the strongest drivers of noise, followed by draft and length. These findings offer a path toward generalized models that can scale to larger datasets, notably AIS-based systems, empowering real-time noise prediction and mitigation. This approach supports the assessment of sensitive ecosystems and efforts to better manage ship traffic to protect marine mammals.

4:45

4pUW13. Monitoring underwater noise pollution in the Venetian lagoon, Italy. Alberto Carrera, Jacopo Boaga (Università di Padova, Padova, Italy), and Lapo Boschi (Università di Padova, via Gradenigo 6, Geosciences Dept., Padova 35131, Italy, lapo.boschi@unipd.it)

Underwater soundscape recordings in shallow, semi-enclosed environments such as the Venice Lagoon is challenging, due to variable bathymetry, soft mud substrates, strong tidal dynamics, and intense anthropogenic pressure from shipping, port operations, and tourism. Quantitative soundscape studies tend to rely on expensive and logistically demanding recording systems, limiting spatial coverage and long-term deployments. Recently, a growing number of low-cost underwater acoustic recorders have become available on the market. While these instruments offer new opportunities for dense spatial monitoring, they typically lack factory calibration and may exhibit significant inter-sensor variability. We deployed a small network of recorders in the Venice Lagoon, and implemented a dedicated calibration and cross-validation protocol using a reference-calibrated hydrophone. Contextual environmental and anthropogenic data, including vessel traffic, tidal phase, and meteorological conditions, was integrated to support interpretation of the recorded signal. Based on the collected data, we have conducted a preliminary characterization of the lagoon soundscape, identifying dominant temporal patterns and assessing the relative contributions of natural and anthropogenic sources. By improving data reliability and comparability, this approach supports more accurate and reproducible soundscape in shallow, human-impacted lagoon environments, providing a foundation for future ecological studies and management-oriented applications.

Session 5aAA

Architectural Acoustics: Acoustics Potpourri

Brandon Cudequest, Cochair
Threshold Acoustics, 141 W Jackson Blvd Ste. 2080, Chicago, IL 60604

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

Contributed Papers

9:45

5aAA1. Design and construction of a low-cost acoustic measurement space. Eric Rokni (Phys., High Point Univ., One University Parkway, High Point, NC 27268, erokni@highpoint.edu) and Claire Klein (Phys., High Point Univ., High Point, NC)

Quiet, nonreverberant spaces are essential for accurate acoustic measurements, yet such facilities are often cost-prohibitive for undergraduate institutions (>$100 000). This presentation describes the design and construction of an affordable chamber (<$3000) suitable for simple acoustic experiments. Students can be engaged in all aspects of the project from the conceptual design, woodworking, and acoustic characterization, providing them with hands-on experience with practical skills and acoustic measurement techniques. We will present a project timeline, general parts list, and characterization results for the completed chamber. The project is scalable to accommodate varying room sizes, budgets, and experimental requirements, making it an accessible solution for teaching and research in acoustics.

10:00

5aAA2. Sound absorption performance of layered micro-slit panels with baffled cavities. Ralph O. Peterson-Veatch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, roscarpv@gmail.com), Ziqi Chen, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This paper presents the design and experimental validation of absorptive broadband meta-surfaces; specifically, the development of layered micro-slit panels (MSPs) paired with baffled cavities (BCs) to achieve effective absorption across a broad frequency range while minimizing thickness. Currently, single-layer conventional MSP absorption is limited to a narrow frequency range. Additionally, absorption of low frequencies requires substantial posterior cavity depth. This contributes to a meta-material thickness that causes limitations in practical applications. This study enhances previously developed meta-structures. Theoretical models are leveraged to guide the fabrication of the broadband MSP + BP + BCs, which are then experimentally validated utilizing impedance tube measurements. Causal analysis is performed to evaluate and compare experimental and modeled results. Corrections are then reintroduced into the model to further improve performance. This work exhibits the possibility for practical meta-surfaces as well as emphasizes the value of causal analysis for robust design and experimental validation.

10:15

5aAA3. Analysis of listening room acoustic adaptation using reduced-scale modeling. Aleksandra A. Sawczuk (Mech. and Vibroacoustics, AGH Univ. of Krakow, Mickiewicza Av. 30, AGH Bldg. D1, Kraków 30-059, Poland, sawczuk.aa@gmail.com) and Bartlomiej Chojnacki (Mech. and Vibroacoustics, AGH Univ. of Krakow, Cracow, Poland)

Recent advances in acoustic scale modeling and measurement techniques have expanded the applicability of reduced-scale studies. However, the acoustics of small rooms remain insufficiently explored using such approaches. To address that gap, this paper presents measurement results obtained using 1:8 scale experimental setup for small-room acoustics. The methodology is validated through a direct comparison of acoustic parameters measured in the scaled model and in the corresponding full-scale reference room, with particular emphasis on temporal decay characteristics and modal behavior. The study includes an evaluation of early reflection control, reduction of dominant room modes using tailored low-frequency absorbers, and the performance of dedicated diffusive structures. Additionally, different sound sources are compared according to their frequency response and radiation directivity to assess their suitability for scale-model measurements. The analysis methods include decay curve evaluation, waterfall plots, and short-time Fourier transform analysis for detailed investigation of room resonances. The expected outcome of the project is an assessment of the accuracy with which acoustic parameters of small rooms can be determined using reduced-scale models, as well as the development of effective procedures for testing and optimizing acoustic treatment strategies.

10:30

5aAA4. Labyrinthine acoustic metamaterial for sound insulation with preserved airflow. Agata Maciuszek (Dept. of Mech. and Vibroacoustics, AGH Univ. of Cracow, Mickiewicza Av. 30, Cracow 30-059, Poland, maciuszek.agata24@gmail.com) and Klara Chojnacka (Dept. of Mech. and Vibroacoustics, AGH Univ. of Cracow, Krakow, Poland)

Recently developed acoustic metamaterials offer a way to transcend the limitations of the traditional materials: when it comes to size, weight and their airflow permeability, simultaneously maintaining high sound insulation values in the low frequency range. Manipulating the structure's geometrical properties allows us to tune their acoustic response to targeted frequency ranges, without a significant increase in overall dimensions. In this study, a labyrinthine acoustic metamaterial structure is considered as a solution for high sound insulation and undisturbed airflow. The effectiveness of the structure is based on Fano resonance, which enables high sound insulation resulting from the interaction between the labyrinth structure and a straight air duct. Firstly, the numerical model was created in COMSOL Multiphysics using the pcoustic module to examine the influence of dimensions of said structure on its sound insulation properties. Furthermore, it was analyzed how serial arrangement of several different resonant elements influences the effective range of the metamaterial. Taking into the account the results of the simulations, final structures were selected and the prototypes were made using 3-D printing. The structure effectiveness was measured in the impedance tube according to the ASTM E2611-09 standard in order to validate the model.

10:45

5aAA5. Using generative artificial intelligence to control immersive audiovisual environments for education and performance. Marcus Welch (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 1129 Sage Ave., Troy, NY 12180, welchm5@rpi.edu) and Jonas Braasch (School of Architecture/Future of Computing Inst., Rensselaer Polytechnic Inst., Troy, NY)

Augmented-reality acoustic experiences become more immersive when users do not need to stop to use a mouse, keyboard, or other control device to control the simulation. To that end, this work presents a device-less interface for the control of Rensselaer Polytechnic Institute's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab), which is equipped with a panoramic visual display and a 128-channel wave-field synthesis speaker array. We use beamforming methods and a deep-learning speech recognizer (OpenAI's Whisper) to effectively capture and transcribe user speech, prompting a chatbot (ChatGPT) to generate commands in protocols like Open Sound Control (OSC). An initial sticking point was overcome by implementing an interpreter layer (OpenInterpreter), which allows code to be run locally, completing the operation. The implementation of interactive conversation between the user and the space is planned, which will use a generative text-to-voice platform like ElevenLabs. This paradigm takes advantage of the effectiveness of large language models at coding small, self-contained scripts for deployment on specific hardware, alleviating the need for humans to predict a potential user's every need. Examples of usage are real-time acoustic design feedback, immersive language-learning environments, and reconstructions of historical performances.

11:00

5aAA6. A small reverberation chamber to measure sound transmission loss in 3-D-printed structures. Trigun Dinesh Maroo (Dept. of Systems Eng., George W Donaghey College of Eng. and Information Technol., University of Arkansas at Little Rock, 2801 S University Ave., Little Rock, AR 72204, dr.tmaroo@gmail.com) and Andrew Wright (Dept. of Systems Eng., George W Donaghey College of Eng. and Information Technol., Univ. of Arkansas at Little Rock, Little Rock, AR)

3-D-printed structures are useful in noise mitigation applications. Many acoustical designs can be generated by varying their infill density, infill pattern, shape, and material. However, their sound transmission loss (STL) characterization using existing standardized methods is challenging. The reverberation room method, which measures the random incidence STL, requires a large room (>50 m^3) and a minimum specimen dimension of 2.4 m (excluding thickness). This method is cost-prohibitive for small research institutions, and specimen sizes ≈ 0.02 m^2 are large for economical 3-D printers. The impedance tube method, which measures the normal incidence STL, cannot measure at high frequencies. A correlation between normal and random incidence STL, if any, is unknown. This paper presents the design and validation of a small reverberation chamber (0.49 m^3) for rapid experimental STL characterization of new specimens, including 3D-printed structures. A custom Python script is developed for signal processing and calculating the sound pressure levels. Solutions to the 3D-printing fabrication challenges associated with large specimen sizes of single and multi-materials are discussed. The STLs of single-material PLA and ABS specimens, and multi-material PLA + TS + TPU and ABS + TA + TPU specimens, are measured at 50%, 100% infills.

11:15

5aAA7. Examining musicians' performance experiences in a variable-acoustic, multi-use environment. Kathleen Ying-Ying Zhang (Music Res., McGill Univ., Dept. of Music Res., Montreal, QC H3A1E3, Canada, yingying.zhang@mail.mcgill.ca), Paul Geluso (Music and Performing Arts Professions, New York Univ., New York, NY), Richard King, and Wieslaw Woszczyk (Music Res., McGill Univ., Montreal, QC, Canada)

As a listener, the performing musician has particular perceptual needs that often differ from those of their audience. This research examines the experiences of musicians in a variable-acoustic, multi-use performance environment in order to better understand how they might perceive different acoustic settings (RT .64s and RT 1.70s) and room positions. Following a musician-guided pilot to help determine points of acoustic interest in the space, participants with formal musical training took part in this study. Asked to rate attributes taken from the 9-Factor Room Acoustic Quality Inventory (RAQI), along with the novel "participation" performer-centric room attribute, responses generally showed that instrument radiation, frequency response, and cognitive load impacted ratings. For instance, technical room issues, if insufficiently acoustically triggered by the instrumentalists, were more likely to go unnoticed. However, prior training may also have had an impact. While the RAQI contains technical terminology, it also contains attributes such as "intimacy" that performers responded to in a more intuitive manner. These attributes were more likely to receive attention, even regardless of instrumentation. Alongside ratings, discussions with subjects allowed for a mixed-methods approach to analyzing their responses.

11:30

5aAA8. Acoustics of the Muchukundeswara Temple. Shashank Aswathanarayana (Ctr. for Heritage Management, Ahmedabad Univ., Jack Child Hall, Rm. 202, American Univ., Washington, DC 20016, shashank@american.edu) and Braxton Boren (Performing Arts, American Univ., Washington, DC)

The Chola dynasty had a tremendous impact on religious art and architecture in South India. Over 100 temples in the Kaveri River delta from the 9th–10th centuries C.E. are classed as "early Chola temples." The period is also important as it was during this time that Tamil hymns began to be incorporated into Tamil liturgy. Historians, therefore, emphasize the need to examine the development of early Chola temples. Of the existing studies, the focus remains on the temples' exteriors, as the interiors are strikingly unadorned. Concurrently, the acoustics of South Asian religious architecture remains underdeveloped, leaving these temples largely unexamined. The Muchukundeswara Temple in Kodumbalur in present-day Tamil Nadu is an excellent example of an early Chola temple. This Shaivite temple was built in the 10th century C.E. and is largely unaltered aside from shrines added in the 14th–15th centuries C.E. It also assumes prominence as its location is associated with the Shaivite saint Idankazhi Nayanar. In this paper, we analyze the impulse responses and decay curves measured at the Muchukundeswara temple. Our analysis includes both standard acoustical parameters like T30, C80, and D50 and nonstandard parameters such as resonance quality and resonance width.

11:45

5aAA9. Eurocentricity is the Achilles Heel of acoustics: An acoustical comparison of performance spaces at union college. Alex B. Roginski (Phys., Union College, 807 Union St., Schenectady, NY 12308, aroginski42@gmail.com)

Contemporary acoustics has usually focused on Western classical music and European art standards. At the inception, this focus helped establish an accepted standard for acousticians, but many musical genres prefer different parameters for an ideal performance. Assessing performance spaces based on one style of music is a flawed choice; rather, one should assess the space's unique qualities and conclude which musical genres sound best. In this research, I compare and contrast the architectural acoustics of seven performance spaces on one college campus—three traditional musical auditoriums, one lecture space, two event halls, and a sound recording studio. While most spaces generally handle most varieties of sound well, some are predisposed to certain styles of music more than others, and many could be optimized to welcome more styles. My suggestions include the addition of acoustic paneling or fabrics to dampen noise, as well as a more open-minded approach to the field of acoustics that rejects contemporary Eurocentricity.

Session 5aPA

Physical Acoustics: General Physical Acoustics

Andrea P. Arguelles, Chair
Engineering Science and Mechanics, Penn State University, 212 Earth-Engr Sciences Bldg, University Park, PA 16802

Contributed Papers

7:55

5aPA1. Langevin radiation force and its connection with linear acoustics. Chirag A. Gokani (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 9767, Austin, TX 78766-9767, cgokani@arlut.utexas.edu), Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Is acoustic radiation force a consequence of linear or nonlinear acoustics? This question has been discussed for over a century. Regarding his attempt to calculate the radiation pressure in an adiabatic gas exerted by a plane progressive wave on a perfectly reflecting wall, Poynting wrote, "Lord Rayleigh kindly pulled me out of the pit into which I fell, pointing out that when we take into account second-order quantities the ordinary sound equation does not hold" [*Philos. Mag.*, **9**, 393–406 (1905)]. Rayleigh's result applies only to constrained fluids (e.g., plane waves in closed tubes), retaining the quadratic term in the state equation to conserve mass. Poynting's original result (which was developed further by Langevin) neglects this nonlinearity and is appropriate for unconstrained fluids, making it of greater practical importance. Still, traditional derivations of the Langevin force begin with the exact momentum equation, obscuring its linear origins. Here, the Langevin force is derived from the linearized continuity, state, and momentum equations, proving that "for the calculation of the average force correct up to terms of the second order… it is sufficient to find the solution of the linear scattering problem" [L. P. Gor'kov, *Sov. Phys. Dokl.*, **6**, 773–775 (1962)].

8:10

5aPA2. Contactless acoustic transport of matter in air using elongated Bessel-type fields. Dmitrii Nikolaev (Univ. of Helsinki, Gustaf Hällströmin katu 2, Helsinki 00560, Finland, dmitry.nikolaev@helsinki.fi), Denys Iablonskyi (Univ. of Helsinki, Helsinki, Finland), Alexander Martinez-Marchese (Univ. of Michigan, Ann Arbor, MI), Marika Sirkka (Univ. of Helsinki, Helsinki, France), Mikel Aldea (Public Univ. of Navarre, Pamplona, Spain), Chinedum Okwudire (Univ. of Michigan, Ann Arbor, MI), Edward Hæggström (Univ. of Helsinki, Helsinki, Finland), Asier Marzo (Public Univ. of Navarre, Pamplona, Spain), and Ari Salmi (Univ. of Helsinki, Helsinki, Finland)

Acoustic manipulation uses sound waves to exert forces on matter, enabling contactless movement, trapping, and structuring of particles. This approach provides a contamination-free tool for handling matter in applications ranging from chemistry and biology to advanced manufacturing. Despite significant progress, current state-of-the-art techniques for noncontact acoustic manipulation are generally restricted to the manipulation of individual particles or small clusters and are ineffective for transporting extended or dispersed matter such as gases or aerosols. To overcome this limitation, we present a method for continuous and contactless transport of matter in all its principal states along reconfigurable 3-D trajectories. The proposed method forms conduits using elongated Bessel-type acoustic fields generated by a phased array of transducers. This new family of fields can be formed along a desired path. Each mode is adapted to the state of matter and particle size (ranging from μm to mm) to enable efficient guidance. We experimentally demonstrate continuous transport of solids, liquids, gases, and ionized gases along dynamically reconfigurable paths. As shown in this study, this capability paves the way for containerless laboratory processes, precise and contamination-free reagent or energy delivery, and integration into advanced additive manufacturing platforms for controlled material deposition.

8:25

5aPA3. Roton-like dispersion in locally coupled waveguides. Muhammad Bilal Khan (Mech. Eng., Stevens Inst. of Technol. 1 Castle Point, Hoboken, NJ07030, mkhan42@stevens.edu), George Maroun, and Christopher Sugino (Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ)

Roton-like dispersion, defined by a local maximum (maxon) and minimum (roton) at finite wavenumbers, has recently been demonstrated in nonlocal acoustic and elastic metamaterials. Most prior realizations depend on complex 3-D structures or mode hybridization, limiting practical implementation. Here, we present a simple approach for generating roton-like and other exotic dispersions using locally coupled elastic waveguides whose effective dynamics emulate beyond-nearest-neighbor couplings. First, we present a generalized analytical framework to model the coupled-waveguide system, showing that the locally coupled system is equivalent to the nonlocal one. Next, we show that varying the coupling strengths and properties of the secondary waveguides tunes the coupled dispersion of the system, enabling a wide variety of nonlocal behaviors, such as roton-maxon dispersion and undulation. Analytical simulations show regions bounded by zero group velocity (ZGV) points exhibit exceptionally high modal densities. We experimentally validate the model predictions using a glide-symmetric metamaterial with two secondary waveguides. Steady-state and transient measurements confirm the presence of roton-like dispersion, high modal-density regions, and multiple coexisting bending modes. This work establishes an easily manufacturable platform for engineering nonlocal interactions in elastic systems, enabling practical control over exotic dispersions for applications in wave confinement, sensing, energy harvesting, and dispersion tailoring.

8:40

5aPA4. Acoustic frequency-up conversion effect by nonlinear thin plates. Alexis Mousseau, Soizic Terrien (Institut d'Acoustique—Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le Mans, France), and Vincent Tournat (Institut d'Acoustique—Graduate School, Laboratoire d'Acoustique de l'Université du Mans, UMR CNRS 6613, Laboratoire d'Acoustique de l'Université du Mans (LAUM), Le Mans 72000, France, vincent.tournat@univ-lemans.fr)

Thin plates are well-known nonlinear elements responsible for remarkable acoustic effects, as exemplified by musical instruments such as the kazoo or the gong. In this presentation, we investigate the efficiency and characteristics of transferring the acoustic power of an incident airborne wave to higher frequencies using thin clamped plates. The plates exhibit distinct nonlinear vibrational regimes depending on the amplitude and frequency of the excitation. At low excitation levels (typically below 85 dB),

the response is weakly nonlinear and dominated by harmonic generation, whereas higher levels lead to quasi-periodic and chaotic dynamical regimes. Because these nonlinear regimes govern the efficiency and bandwidth of acoustic frequency up-conversion, we systematically study how key physical and geometrical parameters influence their onset. A parametric study of plastic plates with thicknesses in the range 10–100 μm shows that the up-conversion effect is primarily governed by resonance at the first eigenfrequency, which depends not only on thickness but also on the static tension induced by clamping. Building on these results, we propose a multi-plate configuration that combines elements with different properties to achieve broadband acoustic frequency up-conversion. These findings open perspectives for the design of strongly nonlinear acoustic metamaterials enabling sound control beyond the limits of linear response.

8:55

5aPA5. Negative Doppler effect and negative transmission of an acoustical wave overtaken by a finite-size weak shock. Guillaume Musy (Institut Jean le Rond d'Alembert, Sorbonne Université, 4 Pl. Jussieu, Paris 75005, France, guillaume.musy@sorbonne-universite.fr), Ronan Delalande (PIMM, Ecole Nationale des Arts et Métiers, Paris, France), Mathieu Ducousso (PIMM, Ecole Nationale des Arts et Métiers, Magny les Hameaux, France), and François Coulouvrat (Institut Jean le Rond d'Alembert, Sorbonne Université, Paris, France)

We investigate the interaction in an inviscid fluid between a weak shock wave and an acoustic wave, which can be either contra- or co-propagative. In this last case, the shock wave, propagating supersonically overtakes the sound wave. Theory for a sustained shock reveals extraordinary transmission phenomena, such as a transmitted sound wave propagating toward the shock and not away from it, negative transmission (tangential directions of the incident and transmitted waves are opposite), and anomalous Doppler effect (transmitted frequency lower than the incident one). Such behaviors are similar to those observed in metamaterials. As a sustained shock is not realistic, this theory is extended to the case of a shock of finite size, with a post-shock flow progressively returning to rest. In this case, the problem is solved numerically by a finite-difference time domain algorithm combined with a proper management of the Rankine-Hugoniot relations. The algorithm is validated at 2-D by comparison with the sustained shock theory. The main effects predicted by sustained shock theory are shown to persist, especially in the co-propagative case with negative transmission and anomalous Doppler effect. [Work supported by ANR project LASERSHOWS contract Nr 24-CE51-3544.]

9:10

5aPA6. Modeling sound attenuation of near shore tall vegetation. Christian Di Nicolantonio (Mech. Eng., The Catholic Univ. of America, 7033 Highland Meadows Ct, Alexandria, VA 22315, cjdinicolantonio@gmail.com), Christopher Crognale, Ariel Wise (Mech. Eng., The Catholic Univ. of America, Washington, DC), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, John Judge, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

Tall grass and mangroves are a common feature of marine and riverine coastline. Understanding and modeling the acoustic attenuation introduced by this type of vegetation is necessary for predicting atmospheric acoustic transmission loss in a near shore environment. A set of impedance tube experiments on 3-D printed samples that emulate the geometry and spacing characteristic of a field of tall grass blades is used to estimate acoustic attenuation of this type of medium. Preliminary outdoor measurements were conducted on a field of marsh grass to validate the impedance tube results. The estimated acoustic attenuation of tall grass is used as input into a parabolic equation solver. In the solver, the acoustic response of a grass field is modelled by replacing the volume occupied by the vegetation with an equivalent fluid. The equivalent fluid is assigned a complex wavenumber with the imaginary part being the attenuation coefficient estimated from the experiments. Finally, simulation results are compared to outdoor measurements. [Work supported by ONR N00014-24-1-2400, N00014-24-1-2437, N00014-22-1-2492, and N00014-22-1-2410.]

9:25

5aPA7. Absorption of low-frequency sound by clouds: Model and laboratory experiment. Pavan K. Kirar (Institut Jean Le Rond d'Alembert, Sorbonne Université, 4 Pl. Jussieu, Paris, Île-de-France 75005, France, pavan-kumar.kirar@sorbonne-universite.fr), Benjamin Véron (Institut Jean Le Rond d'Alembert, Sorbonne Université, Paris, Île-de-France, France), and François Coulouvrat (Institut Jean Le Rond d'Alembert, Sorbonne Université, Paris, France)

Atmospheric sound propagation is strongly influenced by meteorological conditions, particularly through absorption mechanisms that are typically modeled for humid air without clouds. When sound propagates through clouds, additional physical processes arise that may significantly alter both absorption and dispersion. Existing theoretical models predict that, at low frequencies, sound attenuation in clouds is dominated by viscous drag associated with the oscillatory motion of droplets and by heat and mass transfer due to partial vaporization and condensation during acoustic compression and expansion. These mechanisms can lead to absorption levels several orders of magnitude higher than in cloud-free air, especially for thick clouds with high liquid water content. Despite these predictions, experimental validation is currently lacking. In this study, sound absorption in a droplet-laden medium is investigated experimentally using a controlled suspension of water droplets generated by atomization inside a closed resonant cavity. The acoustic absorption is quantified through measurements of the resonance quality factor. To isolate volume absorption from wall-related viscous losses, a differential approach is employed by comparing measurements obtained with and without droplets. The experimental setup, design considerations, and initial measurement results are presented. This work provides a first step toward experimental validation of cloud-induced acoustic absorption models.

9:40–10:00 Break

10:00

5aPA8. Acoustic analysis of vortex formation and thermal accumulation in urban street canyons. Claudia Bahamon Lopez (Eng., Univ. of the District of Columbia, 4200 Connecticut Ave NW, Washington, DC 20008, claudia.bahamonlopez@udc.edu) and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, acoustic tomography and numerical simulations are combined to extend the modeling of microclimate dynamics in urban street canyons during extreme heat-wave conditions. Building upon previous studies, the CFD framework is enhanced by incorporating additional acoustic propagation paths, expanding the computational domain to include multiple adjacent buildings, and simulating stronger heat-wave scenarios to improve thermo-acoustic signal sensitivity. The numerical simulations are based on computational fluid dynamics (CFD) modeling of temperature and wind profiles within an expanded street canyon configuration. The denser acoustic path network allows for a more complete tomographic representation of the temperature field, while intensified thermal forcing improves the identification of nonlaminar wind circulation and vortex formation within and above the canyons. The expanded domain further enables the analysis of airflow interactions across building arrays and localized recirculation zones. The combined experimental and numerical approach improves understanding of thermal accumulation, airflow recirculation, and UHI hot-spot formation during extreme heat events.

10:15

5aPA9. Real-time acoustic-visual source localization and tracking in urban environments. Chidera Igwebuike (Comput. Sci., Univ. of the District of Columbia, 2204 Dunrobin Dr., Bowie, MD 20721, chidera.igwebuike@udc.edu), Akemi Vasquez (Comput. Sci., Univ. of the District of Columbia, Washington, DC), Wagdy Mahmoud, and Max Denis (Univ. of the District of Columbia, Washington, DC)

Urban environments pose significant challenges for real-time target localization and tracking due to reverberation, occlusions, and frequent GPS degradation. Visual-only systems often suffer from delayed target acquisition when line-of-sight is obstructed, while acoustic sensing provides passive, low-latency directional cues but lacks spatial resolution for persistent

tracking. This work presents a real-time, edge-deployable multimodal framework that integrates acoustic sound source localization (SSL) with visual object detection and tracking to close this initialization gap. Acoustic cues are employed to actively steer a servo-mounted camera, reducing the visual search space and accelerating time-to-acquisition. Visual detections are performed using YOLOv8, while temporal consistency is maintained via BoT-SORT with camera motion compensation. An extended Kalman Filter (EKF) fuses acoustic DoA estimates and visual bounding box measurements to enable robust tracking under nonlinear motion and temporary visual occlusions.

10:30

5aPA10. Outdoor acoustic drone localization and tracking over a simple built environment. Filippo Vatteroni, Wagdy Mahmoud (Univ. of the District of Columbia, Washington, DC), and Max Denis (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, max_f_denis@hotmail.com)

In this work, acoustic localization and tracking of drones is investigated. Traditional sound source localization methods are employed, including time difference of arrival and beamforming angle of arrival methods, exploiting the acoustic characteristics of the drones to estimate their location. Of particular interest is the intrinsic harmonic sound emitted by the rotors of the drone. Kalman filtering is used for tracking and minimizing large position estimation errors. Outdoor measurements of various drones from a sparse number of microphones positioned in a simple built environment are studied. The evaluation metrics for drone localization and tracking are presented. The many factors that explain these results are discussed.

10:45

5aPA11. Advanced deep learning and machine learning approaches for vehicle detection using acoustic and seismic data. Abdoulaye Barry (SEAS, Univ. of the District of Columbia, 4200 Connecticut Ave NW, Washington, DC 20008, abdoulaye.barry@udc.edu) and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, algorithms for classifying ground vehicles using acoustic and seismic measurements are presented. Classification based on these modalities enables identification of vehicle types through sound and ground vibration characteristics and is critical for enhancing Warfighter situational awareness in hostile environments. Artificial intelligence and machine learning techniques are used to develop a robust framework that integrates acoustic and seismic data to distinguish between military and nonmilitary vehicles. Performance is compared across traditional single-input machine learning and deep learning approaches and a multiple-input vertical federated learning framework, with emphasis on vertical federated averaging. Results show that acoustic data alone consistently yields higher classification accuracy than seismic data alone for binary military versus nonmilitary classification. Integrating acoustic and seismic data produces the highest accuracy of 95% using Extreme Gradient Boosting (XGBoost). When the military class is expanded to include wheeled vehicles (amphibious assault vehicles) and tracked vehicles (Dragon Wagons), increasing label dimensionality, XGBoost accuracy drops to 73%. In contrast, convolutional neural network (CNN)-based models achieve accuracies above 85%. A federated averaging approach using CNNs as local classifiers attains 90% accuracy. These findings indicate that multimodal vehicle classification with higher class dimensionality requires more complex learning models.

11:00

5aPA12. Spatiotemporal dynamics of high-frequency combustion instability in a single-element liquid rocket engine combustor. Takahiro Seshime (Dept. of Mech. Eng., Tokyo Univ. of Sci., 6-3-1 Niijuku, Katsushika-ku, Tokyo 125-8585, Japan, 4524528@ed.tus.ac.jp), Takanori Haga (Japan Aerosp. Exploration Agency, 3-1-1 Yoshinodai, Chuo-ku, Sagamihara-si, Kanagawa, Japan), Hiroshi Gotoda (Dept. of Mech. Eng., Tokyo Univ. of Sci., Katsushika-ku, Tokyo, Japan), Yusuke Nabae (Dept. of Mech. Eng., Tokyo Univ. of Sci., 6-3-1 Niijuku, Katsushika-ku, Tokyo, Japan), and Ryoichi Kurose (Dept. of Mech. Eng. and Sci., Kyoto Univ., Kyoto daigaku-Katsura, Nishikyo-ku, Kyoto-shi, Kyoto, Japan)

High-frequency combustion instability is a self-excited phenomenon with large amplitude pressure oscillations with frequencies exceeding 1000 Hz. The onset of high-frequency combustion instability causes severe vibrations and can lead to structural damage in liquid rocket engines. We study the spatiotemporal dynamics of high-frequency combustion instability in a single-element liquid rocket engine combustor by a large-eddy simulation, focusing mainly on the flow structure and acoustic energy field. The LOX dense core exhibits periodic flapping due to the fluctuations in the momentum flux ratio in response to the pressure fluctuations during high-frequency combustion instability. The vortex dynamics originating from the shedding of the interface between H_2 and LOX is strongly associated with the pressure fluctuations in the recessed region of shear-coaxial injector. The vortical structures significantly affect the formation of acoustic energy sources.

11:15

5aPA13. Dynamic behavior of thermoacoustic instability in a dual-swirl combustor. Taisuke Yamamoto (Mech. Eng., Tokyo Univ. of Sci., 6-3-1 Niijuku, Katsushika-ku, Tokyo 125-8585, Japan, 4524563@ed.tus.ac.jp), Taira Sumiyoshi (Mech. Eng., Tokyo Univ. of Sci., Kastushika-ku, Tokyo, Japan), Yusuke Nabae (Mech. Eng., Tokyo Univ. of Sci., 6-3-1 Niijuku, Katsushika-ku, Tokyo, Japan), and Hiroshi Gotoda (Mech. Eng., Tokyo Univ. of Sci., Katsushika-ku, Tokyo, Japan)

Thermoacoustic instability is a self-excited phenomenon characterized by feedback coupling among fluctuations in acoustic pressure, heat release rate, and velocity field, which induces large amplitude acoustic oscillations in various combustors. In this study, we mainly investigate the nonlinear dynamics of combustion states in a dual-swirl combustor using complex-system approaches. Dynamic behavior of the acoustic pressure fluctuations under goes a significant transition from stable state to limit-cycle oscillations via intermittent behavior with increasing the equivalence ratio. The phase synchronized state between the acoustic pressure and heat release rate fluctuations occurs during limit cycle oscillations.

11:30

5aPA14. On-site measurements of characteristics of propagation of power transformers audible noise. Mats Bernesjo (Eng., Hitachi, 13695 Rider Trail N, Earth City, MO 63045, mats.s.bernesjo@hitachienergy.com) and Dr. Ramsis Girgis (Eng., Hitachi, Earth City, MO)

Power transformers are often installed near residential areas. Audible noise of these transformers propagates to these areas, causing annoyance, discomfort, sometimes resulting in sound levels that exceed levels allowed by noise ordinances. This paper presents noise measurements performed by the authors at different substations in North America with the objective of determining the sound propagation characteristics of transformer noise and its frequency components. This is in addition to examining the impact of sound walls around transformers, tank mounted radiators, and interaction between sound waves propagating from different transformers. One of the main findings is that distance attenuation is lowest for the 120 Hz frequency component and much higher for higher frequency components. As a result, transformer noise appearing at residential areas is dominated by this frequency. Hence, existing methods of calculating distance attenuation of transformer noise will need to be revised. The measurements also show that sound walls around transformers are effective even at large distances from the transformers. Finally, it was found that sound waves of the same frequency propagating from different transformers in same substation can add

or cancel each other resulting in a dip or a rise in the sound level at some points around the substation.

11:45

5aPA15. Synchronization-free acoustic sensor localization for GPS-denied array deployment. Hanieh Agharazi (NCPA, Univ. of MS, 145 Hill Dr., University, MS 38677, agharazi@olemiss.edu), Garth Frazier (NCPA, Univ. of MS, Oxford, MS), and Wayne Prather (NCPA, Univ. of MS, University, MS)

This presentation introduces a time synchronization-free extension of our acoustic positioning system (APS) for localizing sensors in GPS-denied environments using only local measurements from immediate neighbors. The motivation is to support array signal processing applications that require accurate sensor positions, while avoiding network-wide clock synchronization. Sensors with unknown locations exchange known acoustic probe signals. In the original APS, each neighbor estimates time-of-flight (ToF) via time-domain cross-correlation, which depends on synchronized timing. Here, we estimate propagation delay without synchronization using frequency-domain group delay. For each neighboring pair, we measure the phase response across frequency and estimate the delay from the slope of phase versus frequency. The reciprocal measurements in both directions cancel clock offset and drift. The estimated delays are converted to pairwise distances and relayed to a central node, forming a sparse local distance set over the network. We then apply our proven APS localization to reconstruct the 2-D configuration by solving a nonlinear least-squares problem. The nonlinear optimization algorithm refinement minimizes the mismatch between measured distances and model-predicted distances (and can jointly estimate wind effects when needed). The presentation details the reciprocal range, integration with the optimization pipeline, and expected accuracy under noisy, local-only measurements.

FRIDAY MORNING, 15 MAY 2026 SALON A, 9:00 A.M. TO 10:30 A.M.

Session 5aSA

Structural Acoustics and Vibration: Damping, Dissipation, and Energy Attenuation

Nathan Geib, Cochair
Mechanical Engineering, University of Michigan, 2350 Hayward St., Saint Joseph, MI 48109

William A. Willis, Cochair
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Contributed Papers

9:00

5aSA1. Towards understanding friction driven response in vibrating structures. Iyabo G. Lawal (Mech. Eng., California State Polytechnic Univ., Cal Poly Pomona, Pomona, CA 91768-4062, iglawal@cpp.edu)

Structure-borne noise represents the interplay between the vibration of structures and acoustics and intersects both machine performance and human comfort. Visco-elastic materials have been used in systems and structures that require a high degree of damping [1] provided by the visco-elastic core which has a higher shear-modulus that the elastic outer layers. Friction is also well known for its dissipation properties, however, its implementation in practical applications is limited due to the difficulty in modeling dynamic behavior at frictional interfaces. Dissipative behavior of frictional interfaces is generally unknown in real dynamic systems. One feature of friction contacts is the development of "stick-slip" motion caused by stuck and sliding points along the interface [2]. This motion is a form of wear, specifically fretting wear and causes damage to interfaces that changes their overall structural response. This study seeks to understand how stick-slip develops along a friction interface of an axially loaded structure. The extended Hamilton's principle will be used to develop an FEA solution to explore stick-slip induced vibration for a simple structure. Comparison with a Wrinkler foundation model will be used to quantify differences in the solutions. [1] Wang, H., Lee, H.P. and Du, C. Analysis and optimization of damping properties of constrained layer damping structures with multi-layers. *Shock and Vibration*. V., 2021, 11 pages. [2] Vingsbo, O., Söderberg, S., 1988. On fretting maps. *Wear* **126**, 131–147.

9:15

5aSA2. A novel fractional derivative model for rubber-like viscoelastic materials. Tamás Pritz (Budapest Univ. of Technol. and Economics, Műegyetem rkp. 3-9, Budapest 1111, Hungary, tampri@eik.bme.hu)

Fractional derivative models provide a powerful tool for describing the frequency-dependent dynamic elastic and damping behavior of viscoelastic polymeric materials used in sound and vibration control. Existing literature primarily focuses on modeling the damping peak of materials that occurs in the transition range of rubber-like polymeric materials. While advanced models, such as the four- or five-parameter fractional Zener model, effectively describe the damping peak, they are usually inadequate in the rubbery range of viscoelastic behavior. This paper introduces a novel fractional derivative model, characterized by five parameters, which is capable of describing the dynamic behavior of rubber-like materials across both the transition and rubbery ranges. The derivation of the constitutive stress–strain relationship is presented, followed by a comprehensive analysis of the model. The model behavior is demonstrated over a wide frequency spectrum. The physical meaning of the model parameters is clarified, and thermodynamic constraints are established to ensure physical admissibility. Finally, the proposed model is fitted to experimental data on rubber-like materials used for structure-borne sound insulation, demonstrating its accuracy and a wider range of applicability compared to existing fractional derivative models.

9:30

5aSA3. Estimating natural frequencies and decay rates of submerged targets from hydrophone data. Halim Polat (Mech. Eng., Boston Univ., Dept. of Mech. Eng., Boston, MA 02215, halopolo@bu.edu), James G. McDaniel (Mech. Eng., Boston Univ., Boston, MA), Irena Lucifredi (SOFAR Acoust., LLC, Woburn, MA), and Jeffrey Gilbert (Triton Systems, Inc., Chelmsford, MA)

Identification and classification of submerged targets using sonar rely on accurate determination of parameters that distinguish among different targets. These parameters must remain robust in the presence of ambient noise and variations in target orientation. Resonance scattering theory offers a principled route to parameter definition by showing that the fluid-loaded resonances of a target are encoded in hydrophone measurements made at a distance. In the resonance literature, these natural frequencies and associated decay rates are often described as a fingerprint of the target and are regarded here as the parameter set. This study is motivated by the use of impulsive acoustic sources that excite multiple resonances simultaneously. We develop and evaluate methodologies for estimating fluid-loaded natural frequencies and modal damping rates from hydrophone recordings following impulsive excitation. Time-domain and frequency-domain techniques are compared, with particular attention to the effects of additive noise and the spatial distribution of hydrophone sensors on estimation accuracy and robustness. Comparative performance assessments using simulated data identify strengths and limitations of each approach and provide practical guidance for sensor deployment and signal processing. [Work supported by the Office of Naval Research under Contract No. N68335-22-C-0502.]

9:45

5aSA4. An experimental study on the effect of cross-section on the thermoelastic damping of thin beams. Ruoxi Yang (Brigham Young Univ., N284 ESC, Provo, UT 84602, ruoxiy@byu.edu), Joshua T. Mills (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Peter K. Jensen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Micah Shepherd (Brigham Young Univ., Provo, UT)

Thermoelasticity has been well-studied for thin beams, where the length of the beam is large compared to its thickness and width. The original theory, proposed by Zener, includes direct dependence on the properties of the material and the thickness of the beam, but only indirect dependence on the cross-sectional area via the natural frequency of the beam. Zener's model shows no dependence on the cross-sectional shape. In this study, the effect of the cross-section on thermoelastic damping of thin beams will be considered in audio range frequencies. Specifically, the effect of varying cross-sectional area and cross-sectional shape will be investigated experimentally. Damping will be measured for aluminum beams, which are supported by thin strings inside a vacuum chamber in order to minimize acoustic radiation and boundary condition losses. The results show measurable differences for the different cross-section configurations, suggesting that cross-section may play a role in the thermoelastic damping of thin beams.

10:00

5aSA5. Transient attenuation by dual-chamber self-leveling pneumatic vibration isolators. Vyacheslav M. Ryaboy (Photonics Solutions, MKS Inc, 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

Pneumatic vibration isolators combine low stiffness with high load capacity, which makes them ideal for creating low-noise environments necessary for cutting-edge scientific experiments and precise manufacturing. However, the low stiffness makes them less suitable for applications involving transient events such as fast stage movements and robotic loading–unloading operations. Therefore, research on limiting performance and optimization of pneumatic isolators under transient loads is highly desirable. Whereas the dynamics of pneumatic isolators in stationary environments in the frequency domain have been thoroughly studied in the literature, the modeling and optimization of transient processes in the time domain have not attracted adequate attention. This paper presents linear models of dual-chamber pneumatic vibration isolators with self-leveling via mechanical feedback. Causality of the mechanical models is addressed. Time histories under typical initial disturbances are presented. The trade-off between the settling time and the vibration transmissibility is discussed. Analytical and numerical solutions for optimal internal damping levels and feedback gains, in the sense of fastest settling after disturbance, and stability conditions are derived.

10:15

5aSA6. Radiated sound analysis of acoustic black holes for interior noise reduction. Noah Robertson (Aerosp. Eng., Penn State Univ., 308 ECoRE Bldg., State College, PA 16802, npr7144@outlook.com), Nicholas Vlajic (Penn State Appl. Res. Lab., State College, PA), and Edward Smith (Aerosp. Eng., Penn State Univ., State College, PA)

The continued refinement of manufacturing and repair techniques in rotorcraft has led to an increase in lightweight, thin composite stiffened panels. Due to their light-weight nature, these materials can lead to higher levels of radiated sound in the complex acoustic environment of the rotorcraft interior. This research explores the use of panels embedded with acoustic black hole features as a passive noise reduction technique for a representative interior noise setting. A coupled structural-acoustic finite element model was constructed to explore the effects of acoustic black hole geometry and design changes. Vibration data and interior pressures were experimentally measured to make relative comparisons between an enclosure with and without acoustic black holes. The overall objective of this work is to determine the effectiveness of acoustic black holes in reducing radiated sound as well as demonstrate capabilities in designing the geometry of the black hole features to target specific structural vibration frequencies. This capability has particular application in interior noise environments with high tonal components, like blade passing frequencies or gearbox noise.

Session 5aSC

Speech Communication: Speech Perception & Listening Conditions (Poster Session)

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

Contributed Papers

5aSC1. Effects of vowel space size on temporal processing of two-syllable vowel sequences. Catherine L. Rogers (Commun. Sci. and Disord., Univ. of South Florida, University of South Florida, Dept of Comm Sci and, PCD1017, Tampa, FL 33620, crogers2@usf.edu)

This study investigated the effects of vowel-space size on the identification of two-syllable sequences. Large-space ("hid, head, hood" and "hod") and small-space ("hid, head, had" and "hud") syllable sets were recorded and processed to equalize duration, intensity, and fundamental frequency. Young, normal-hearing listeners heard two-syllable sequences and attempted to identify the syllables in order. Time between syllable onsets was adjusted to manipulate task difficulty. Performance across six onset-asynchrony steps was used to estimate 50%-correct thresholds. For data analyzed to date, the median SOA threshold for the small-space set was approximately double that of the large-space set, indicating that spectral area and/or distance between vowel pairs affects temporal processing of even easily identified syllables by young normal-hearing listeners, suggesting potentially greater impacts for clinical populations.

5aSC2. Dramatic declines in perception of time-compressed speech and other challenging listening conditions across the lifespan: Effects of age, general cognition, and hearing. McKenna Revis (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA, mckenna-revis@uiowa.edu), Sarah E. Colby (Univ. of Iowa, Ottawa), Samarium Knight (Dept. of Psych., Univ. of Illinois Urbana-Champaign, Champaign, IL), Sita Carraturo (Dept. of Otolaryngol., Univ. of Iowa, Iowa City, IA), and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Many older adults report social isolation. This may derive from hearing and cognitive declines, but the contribution of language processing is unclear. A critical language skill is the ability to cope with challenging listening conditions like noise, quiet speech, or time-compressed speech. We report results from an ongoing study of over 200 typically hearing adults aged 30–90. To test the ability to perceive time-compressed speech, participants heard sentences compressed to 30% of their duration and repeated them aloud. This was part of a larger battery testing cognition, language, speech perception, and patient-reported outcomes. We first asked if time-compressed speech perception declines with age. Age alone accounted for 60.2% of the variance in time-compressed speech ($N = 173$, $p < 0.0001$). For participants with hearing (PTA) and cognitive scores ($N = 124$), these factors accounted for 46% of the variance, and only hearing was significant ($p < 0.0001$). However, age predicted an additional 16.1% of the variance ($p < 0.0001$). Thus, age-related declines in time-compressed speech perception are not wholly the result of hearing and cognitive decline. Time-compressed speech performance was also correlated with speech reception threshold ($r = 0.570$) and speech in noise (0.504), even after accounting for age and hearing ($p = 0.0089$, $N = 117$). This suggests a core ability to recognize challenging speech.

5aSC3. Effects of mild cognitive stress on auditory discrimination in bilingual and monolingual adults. Ashley Marte (Communications & Performing Arts, City Univ. of New York—Kingsborough Community College, Brooklyn, NY), Aspen Shearer (Brooklyn College, Brooklyn, NY), Laura M. Muscalu (Psych., Ithaca College, Ithaca, NY), and Laura Spinu (Communications & Performing Arts, City Univ. of New York—Kingsborough Community College, 2001 Oriental Blvd., Rm. E329, Brooklyn, NY 11235, laura.spinu@kbcc.cuny.edu)

Cognitive stress can disrupt auditory attention and perceptual decision-making, yet individuals vary in how quickly performance recovers under stress conditions. This study examines whether bilingual language experience is associated with greater resilience and faster recovery in auditory processing after mild cognitive stress. Participants complete an online auditory discrimination task via the Pavlovia platform requiring fine-grained perceptual judgments under time constraints. Following a baseline block, participants perform a brief, time-pressured mental arithmetic task with performance feedback designed to induce mild cognitive stress. The auditory task is then repeated immediately after the stressor, and participants provide brief self-reports of perceived stress. Data collection is underway with a target sample of approximately 80 adults. Primary outcomes include changes in discrimination accuracy, reaction time, and response variability from pre- to post-stressor. We predict that although both bilingual and monolingual participants will show stress-related performance disruption, bilinguals will exhibit smaller decrements and faster recovery. Within bilinguals, higher proficiency and greater active language use are expected to predict stronger recovery profiles.

5aSC4. Great expectations: Evidence of "one shot" learning for speech perception. Anne Marie Crinnion, James Magnuson (Univ. of Connecticut, Storrs, CT), and Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

Prior research suggests that listeners integrate both lexical and phonetic cues over time through perceptual learning. This study tests whether a single exposure is sufficient to shift cue weighting. Such "one shot" learning has been observed in other domains but has not been systematically investigated for speech perception. Listeners ($n = 272$) heard a single exposure trial that contained either inconsistent lexical and phonetic cues (e.g., a clear /g/ in an *-iss* lexical context, where *giss* is not a word but *kiss* is) or consistent lexical and phonetic cues (e.g., a clear /k/ in an *-iss* lexical context). All listeners then completed a standard Ganong task consisting of 160 trials where listeners made phonetic decisions to items along word-nonword and nonword-word continua (i.e., *gift-kift* and *giss-kiss*). Aggregate performance for the 160 Ganong trials showed a weaker influence of lexical context on identification responses for those who heard inconsistent compared to consistent cues on the single expectation trial, indicating that one initial exposure is sufficient to reliably influence cue weighting. Consistent with learning over time, both groups

showed an attenuation of lexical influence across the 160 Ganong trials. These results suggest that failing to account for rapid, one-shot learning may mask rapid adaptation that yields patterns consistent with longer-term cue integration but is driven by different cue weighting processes.

5aSC5. Investigating speech intelligibility in auralized primary school classroom environments. Virginia Tardini (DIN—Dipartimento di Ingegneria Industriale, Univ. of Bologna, Viale Risorgimento 2, Bologna 40136, Italy, virginia.tardini2@unibo.it) and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Good acoustic quality is fundamental to supporting speech intelligibility in classrooms. Achieving this requires consideration of speech intelligibility parameters; however, their perceptual relevance under active classroom conditions remains insufficiently explored, particularly in primary school environments. This study examines the relationship between room acoustic characteristics and speech intelligibility using auralized simulations of a primary school classroom under active conditions. Several acoustic scenarios were created by varying the amount of sound-absorbing material within the same room. Each scenario was evaluated by considering both teaching orientation and listening positions to reflect realistic classroom use. Speech intelligibility was assessed through headphone-based listening tests conducted in a controlled laboratory setting with participants of varying ages, enabling an initial comparison across listener groups. The simulation framework reveals variations in speech intelligibility parameters across acoustic scenarios, while preliminary listening difficulty results suggest that these variations are perceptually relevant. Overall, this work presents an experimental approach for examining speech intelligibility and listening difficulty in realistic classroom scenarios, contributing to evidence-based acoustic design strategies for primary school learning environments.

5aSC6. Developmental trajectories in perception of gender diverse voices using face spectra. Tzu Pei Tsai (Speech, Lang. and Hearing Sci., Indiana Univ., 2451 E 10th St., Apt 1014, Bloomington, IN 47408, tsaitz@iu.edu), Tessa Bent (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Rachael F. Holt, Ryan Hale (Speech and Hearing Sci., Ohio State Univ., Columbus, OH), Rowan Kilgore (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Piper Reusch (Speech and Hearing Sci., Ohio State Univ., Columbus, OH), and Amelia Xanders (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

From early in life, children can discriminate between male and female faces and voices; one-year-olds can make associations between gendered facial and voice cues. However, little is known about this audiovisual integration process for non-cisgender talkers. Thus, we investigated how listeners map voice information from talkers representing five gender identities (cis men, cis women, trans men, trans women, nonbinary) onto five-step morphed face spectra (masculinized-to-feminized) from the Chicago Face Database. Ninety-nine adults and 178 children listened to sentences and selected the face that best matched the voice. Listeners associated cis men's voices with more masculinized faces, cis women's voices with more feminized faces, and voices of trans men, trans women, and nonbinary talkers with mid-spectra faces. Listeners younger than 7 had more variable and less distinct voice-face mapping compared to older listeners. By age 7, children showed adult-like performance patterns. One-third of the voice-face matching variance could be explained by talker average fundamental frequency. Children reach adult-like performance early on this audiovisual matching task, suggesting that they have some ability to form gendered representations of speakers, including those outside the binary, from speech in the early school-age years. [Work supported by Ohio State University Center for Cognitive and Brain Sciences.]

5aSC7. A novel paradigm for rapid determination of listener-specific phonetic boundaries. Shawn N. Cummings (Univ. of California, Irvine, 226 Prospect St., Newport beach, CA 06226, scummin2@uci.edu) and Xin Xie (Univ. of California Irvine, Irvine, CA)

The two-alternative forced choice (2AFC) task is one of the most popular methods in speech perception research, establishing basic effects across domains (Ganong, McGurk, perceptual learning, accent adaptation, etc.). However, interpretation of this task often assumes—but does not directly test—that a token is perceptually "maximally ambiguous" at the point of subjective equality (e.g., when categorized 50% of the time as "sh" and 50% as "s", for an /ʃ/-/s/ continuum). Furthermore, the task requires many repetitions of identical items to establish a reliable categorization curve. This is especially troublesome for studies of adaptation, where input may rapidly attenuate evidence of learning. We introduce a novel task wherein listeners directly control their input along an acoustic continuum by dragging a slider between labelled endpoints, and stop dragging at the point of maximal ambiguity. The task instructions are validated using a visual analogue of the same task (where a "ground truth" ambiguous shape exists between two endpoints). We benchmark the reliability of the new task with traditional 2AFC tasks using the same stimulus continua. Requiring as few as six trials, this new task can facilitate studies where minimizing experiment time or total input is desirable.

5aSC8. Perception of fricative consonants extracted from continuous speech. Lia Carroll (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD) and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., 3400 North Charles, Baltimore, MD 21218, colin.wilson@jhu.edu)

When listening to natural speech, listeners can integrate top-down information (such as lexical knowledge, contextual predictability, and talker characteristics) with bottom-up properties of the acoustic signal. We estimated the contribution of bottom-up properties in two identification experiments using word-internal fricative-vowel sequences ($N > 700$) that were extracted from the Buckeye multitalker corpus of spontaneous speech. Lab participants ($N = 24$) listened to stimuli presented in random order and identified each fricative as one of /f v θ ð s z ʃ h/. Fricatives, fricative-vowel combinations, and talkers were balanced within experimental lists and unpredictable. Identifications were accurate across participants (median = 0.67 [0.62–0.74]) and across fricatives (median = 0.72 [0.61–0.82]) with the exception of /ð/ (0.27). Online participants ($N = 32$) performed the same task with shorter lists; identifications were again accurate across participants (median = 0.67 [0.41–0.83]) and fricatives (median = 0.70 [0.56–0.88]) except for /ð/ (0.37). These results suggest that bottom-up properties provide substantial information about fricative identity in continuous speech regardless of context and talker, though they fall short of performance in a previous study that used fricative-vowel stimuli recorded in isolation (McMurray and Jongman, 2011; median = 0.93 [0.85–0.98] across participants).

5aSC9. Individual differences in speech categorization consistency in Cochlear implant users. Sita Carraturo (Dept. of Otolaryngol., Univ. of Iowa, 278 PBSB, Iowa City, IA 52242, sita-carraturo@uiowa.edu), Sarah E. Colby (Linguist, Univ. of Ottawa, Ottawa, ON, Canada), Frankie Gillenwater (Dept. of Otolaryngol., Univ. of Iowa, Iowa City, IA), and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Speech sound categorization is a fundamental component of speech perception. Recent work on speech categorization in typical hearing populations has shown that categorization consistency (i.e., how consistently an individual arrives at the same percept across different trials for the same token) is an important predictor of broader language outcomes: individuals who are more consistent also tend to perform better overall on language and are more successful in second language learning. Here, we evaluate the relationship between categorization consistency and speech perception outcomes in a population of cochlear implant (CI) users. We further conduct individual differences analyzes to identify factors that contribute to higher categorization consistency among these CI users. CI listeners ($N = 71$) with different device configurations categorized speech sounds in three /p/-/b/ and three /s/-/ʃ/ continua in a Visual Analog Scaling (VAS) task. Results show that higher categorization consistency predicted better speech perception outcomes (CNC: $r = 0.30$, $p < 0.01$; Az-Bio: $r = 0.34$, $p < 0.001$). It also predicted more efficient lexical processing in the Visual World Paradigm, controlling for device configuration. Furthermore, categorization consistency itself is predicted by better temporal and spectral resolution. Additional individual differences analyzes and the implications of these results are discussed.

5aSC10. Individual differences predicting memory and comprehension of medically related speech. Melissa Baese-Berk (Linguist, Univ. of Chicago, 1115 E 58th St., Rosenwald Hall Rm. 203, Chicago, IL 60657, mmbb@uchicago.edu), Tessa Bent, Tzu Pei Tsai, Sami Branson, Gabrielle Kilaras, Fiona Ko, Josie Hinton (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), Alexandra M. Kapadia, and Yilin Hao (Linguist, Univ. of Chicago, Chicago, IL)

Memory and comprehension of medicallyrelated speech is crucial for successful patient–provider interactions. One real-life concern is the impact of hospital noise on these two measures. Our previous research suggested that noise impacts memory and comprehension of medically related speech in hospital noise with large individual differences in performance. However, the factors predicting the listeners' abilities to remember or understand what they heard in noise were not determined. In this study, we investigate how standardized measures of memory, vocabulary, cognition, and hearing may relate to our experimental comprehension and memory measures. Older and younger adults completed a gamified perception study where they took the perspective of a patient during four medical scenarios, in either quiet or hospital noise. After the scenarios, they completed memory and comprehension tasks. Preliminary results suggest that the hearing and vocabulary measures did not reliably predict performance on any task. The Weschler memory test predicted memory for the medical scenario task in quiet, but not in noise. Further, the MoCA predicted comprehension scores, especially in noise. Together, these findings suggest that existing standardized measures may not best predict real world comprehension in ecologically valid scenarios.

5aSC11. Independence of temporal contrast effects across different timescales in speech perception. Liza Heim, Gretchen Jellison, Dijana Mitrovic (Speech Pathol. and Audiol., Marquette Univ., Milwaukee, WI), Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), and Christian E. Stilp (Speech Pathol. and Audiol., Marquette Univ., 230J Cramer Hall, Marquette University, Milwaukee, WI 53233, christian.stilp@marquette.edu)

Perception of speech sounds is shaped by the context of the earlier sounds, including speaking rate. Often, acoustic differences between earlier and later sounds are perceptually magnified. In temporal contrast effects (TCEs), a fast-rate precursor sentence can cause the following target word to sound longer (e.g., longer-VOT "tier"), and a slow-rate precursor sentence can cause the target word to sound shorter (e.g., shorter-VOT "deer"). The novel contribution of this study is the exploration of TCEs across different context durations to examine their consistency on a granular level. On each trial, listeners heard one of three contexts ("a", "the word", "this time I want you to click on the word") spoken at a fast or slow rate, then a target word to be identified as "deer" or "tier". TCEs were observed at all context durations but were surprisingly not consistent in magnitude. While speaking rate forms an important context for speech sound recognition, the degree of its influence may not be reliable across different timescales. This reveals important bounds on how context effects relate to one another. [Work supported by NIDCD.]

5aSC12. The magnitudes of spectral context effects in speech perception are consistent within frequency but not across frequency. Christian E. Stilp (Speech Pathol. and Audiol., Marquette Univ., 230J Cramer Hall, Marquette Univ., Milwaukee, WI 53233, christian.stilp@marquette.edu) and Matthew B. Winn (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Speech perception is shaped by spectral context effects, where spectral properties of earlier (context) sounds influence perception of later (target) sounds. While studies of a single context effect abound, more recent efforts explore relationships between context effects. The magnitudes of spectral context effects are consistent when measured in the same frequency region but not across different frequency regions. However, these findings are combined across multiple separate samples, which raises questions about replicability. Here, relationships within and across frequency were tested in a single normal-hearing sample. Listeners categorized target words as "dot", "got" (varying in third formant onset frequency), "sheet", or "seat" (varying in frication noise spectrum). On each trial, the preceding context sentence was filtered to amplify key frequencies (shifting target word perception toward adjacent frequencies via spectral contrast effects [SCEs]) or attenuate key frequencies (shifting target word perception toward attenuated frequencies via auditory enhancement effects [EEs]). Context effect magnitudes were highly consistent across SCEs and EEs in the same frequency region (dot-got: $r = 0.78$; sheet-seat: $r = 0.79$), but neither SCEs nor EEs were reliable across frequency regions ($r < 0.25$). Results solidify important bounds on the network of context effects in speech perception. [Work supported by NIDCD.]

5aSC13. Manipulation of fricatives for speech perception experiments. 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 Rachel M. Theodore (Univ. of Connecticut, Storrs, CT)

Perceptual categorization of fricative sounds "sh" and "s" has been used to examine perception of a variety of factors such as coarticulation and talker variability. However, the methods for creating stimuli in these experiments are marked by inconsistency, lack of grounding in vocal tract models, and lack of plausible mechanism of auditory processing for the key contrastive properties. In this presentation, we review the advantages and disadvantages of previous stimulus generation methods and, ultimately, present a new method for synthesizing fricatives that is constrained by and shaped by the vocal tract to which it is linked. The key principle is that the fricative spectrum contains peaks at fixed frequencies with varying amplitudes rather than varying frequencies. Those peaks are fixed to align with real or imputed vocal tract resonances in the adjacent vowel, resulting in natural-sounding sequences. Most importantly, the parameters vary on a psychoacoustic scale that is consistent regardless of vocal tract size or vowel context, enabling the same set of parameters across a wide variety of talkers. Finally, we present effort to increase reproducibility in reporting via a standardized method to describe the parameters and illustrate them for publication.

5aSC14. Abstract withdrawn.

5aSC15. Individual differences in binaural integration approach and integration consistency in Cochlear implant users. John B. Muegge (Univ. of Iowa, G60 Psychol. and Brain Sci. Bldg., Iowa City, IA 52242, john-muegge@uiowa.edu) and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

For Cochlear implant (CI) users (and other listeners with hearing loss), access to binaural hearing is associated with improved speech perception and quality of life outcomes relative to monoaural configurations. This benefit may be attributed to a number of reasons, but one unexplored factor is the strategy listeners employ when integrating speech information across ears—for instance, whether they average inputs together or ignore

information from a worse-performing ear. Recent work examined this in listeners with hearing aids, discovering a range of approaches. Here, we extend this to listeners with single sided deafness (SSD) treated with a CI, in particular examining whether these listeners have a tendency to ignore input from their implanted ear. We tested listeners with a speech categorization task once in either ear alone and again diotically, then characterized performance across these conditions as one of seven strategies (e.g., ignoring electric input and averaging inputs) via a gradient descent model-fitting technique. We then assessed the consistency with which each listener displayed these strategies across speech contrasts with a permutations analysis. Interestingly, SSD listeners demonstrate a wide range of approaches and vary in how consistently they apply these strategies, but rarely do they disregard input from their implant.

5aSC16. Effects of talker familiarity and cognitive load on children's speech-in-noise recognition. Mary M. Flaherty (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, maryflah@illinois.edu) and Daniel Fogerty (Speech and Hearing Sci., Univ. of Illinois Urbana-Champaign, Champaign, IL)

This ongoing study examines whether short-term talker familiarity improves children's speech recognition in noise and how this benefit interacts with cognitive demands. Children (5–17 years) with normal hearing complete digits-in-noise (DIN) tasks before and after a brief talker-familiarization phase. The DIN uses an adaptive procedure to estimate speech reception thresholds (SRTs) while children repeat three-digit sequences in speech-shaped noise. Two task variants, forward and backward recall, manipulate cognitive load. During familiarization, children listen to a three-minute household narrative spoken by one assigned talker while coloring a corresponding visual scene. Post-familiarization DIN SRTs are measured for both the familiar and unfamiliar talkers, counterbalanced across participants. To date, 29 children have been tested. Preliminary data suggest the expected effects of cognitive load, with backward recall harder than forward recall by roughly 4.5 dB, and variable familiarity effects. Early patterns indicate greater variability in familiarity benefit among younger children and smaller effects for older children. Findings will help inform the development of a mobile assessment that will determine the contribution of cognitive factors to speech-in-noise recognition across the lifespan, with attention to feasibility and task suitability for young children.

5aSC17. Testing the limits of accent adaptation in 15-month-old infants. Emily Shroads (Psych., Univ. of Toronto Mississauga, CCT 4110 University of Toronto Mississauga, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, emily.shroads@mail.utoronto.ca) and Elizabeth K. Johnson (Psych., Univ. of Toronto Mississauga, Mississauga, ON, Canada)

Studies examining accent adaptation in infancy repeatedly find it emerges only around 19 months (e.g., Best *et al.*, 2009; Mulak *et al.*, 2013). Yet standing in contrast to this body of work—and its support for theories that early phonological representations are underspecified and/or exemplar-based—is a single study finding 15-month-old infants demonstrate readily-observable accent adaptation in some contexts (van Heugten and Johnson, 2014). Here, we perform a much-needed replication and extend their task to probe the limitations of these early abilities. Following van Heugten and Johnson (2014), a group of 15-month-old Canadian English-learning infants heard a female Australian-accented talker read a story and were then tested for adaptation by their recognition of new words produced by the same talker. For another group of infants, we alter the identity of the talker reading the story (a new male talker) but not the test talker. Results from infants in the first group replicate the original study, supporting alternative views that abstract phonological representations are early-emerging. However, circumstances under which infants can leverage learning of accent information across talkers remain unclear. Further work in cross-talker adaptation has the potential to shed light on processes involved in accent adaptation, which remain poorly understood into adulthood.

5aSC18. Adaptation to distorted speech: The effects of hearing loss and hearing aids. Karen Banai (Univ. of Haifa, Mt Carmel, Haifa 3103301, Israel, kbanai@research.haifa.ac.il) and Limor Lavie (Univ. of Haifa, Haifa, Israel)

Adaptation, or rapid performance gains with repeated exposure, is one of the processes that supports the recognition of speech in challenging listening conditions. Age-related hearing loss is associated with substantial declines in speech recognition in challenging conditions. It is not clear whether age-related hearing loss relates to adaptation and whether adaptation is influenced by a hearing-aid. To this end, we characterized adaptation to time-compressed speech and speech in babble noise in a group of 57 participants (ages 60–90) that were tested both with and without hearing aids fitted by an audiologist just before the assessment. Significant adaptation occurred in both speech conditions. Speech in noise recognition accuracy improved by approximately 15% over the course of listening to 16 sentences with hearing aids but remained stable without the hearing aids (OR = 1.24). The rate of adaptation did not depend on hearing. Adaptation to time-compressed speech also occurred only with the hearing aids (improvement of ∼10%, OR = 1.32). Adaptation was stronger in listeners with poorer hearing (OR = 1.18), perhaps due to their poor starting accuracy. Together, these data suggest that amplification can contribute to speech adaptation even in inexperienced hearing aid users.

5aSC19. The effects of syntactic complexity on speech comprehension with informational masking. Ryan Gallagher (Commun. Sci. and Disord., Temple Univ., 1210 Polett Walk, Philadelphia, PA 19122, tup95618@temple.edu), Gayle DeDe, and Jing Shen (Commun. Sci. and Disord., Temple Univ., Philadelphia, PA)

Syntactic cues, such as word order, grammar, and sentence structure, support processing of linguistically complex speech. In everyday communication, background noise poses significant challenges when understanding linguistically complex speech. Yet, little is known about how syntactic cues interact with noise characteristics in affecting speech understanding. This knowledge gap hinders the development of clinical tests that can capture listeners' experience in real-world communication. This current experimental study examines the interaction of complex syntax and informational masking, and whether syntactic cues facilitate speech comprehension in noise. Speech stimuli were spoken complex sentences containing object-extracted relative clauses and subject-extracted relative clauses. Young adults completed a speech comprehension task by answering yes/no questions. The dependent measures were accuracy and reaction time. Results to date indicate that sentences with facilitative syntactic structures were easier to comprehend, and so was the noise with less informational masking. Regarding the interaction between syntax and noise, the comprehension accuracy data suggest listeners rely more on syntactic cues with stronger informational masking. Reaction time data suggest that listeners may be able to use syntax more for speeded processing with less informational masking. These findings support a complex interaction between perceptual and linguistic factors in speech perception. [Work supported by DRS at Temple University.]

5aSC20. Developing a naturalistic approach for an experimental question: Perception of acoustically ambiguous phonemes within semantically disambiguating discourse. Mel Mallard (Psychol. and Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, m.mallard.wustl@gmail.com) and Kristin J. Van Engen (Psychol. and Brain Sci., Washington Univ. in St. Louis, St. Louis, MO)

While acoustic details such as voice onset time can be informative as to the identity of a phoneme (e.g., GOAT versus COAT), these details are not always reliable. Fortunately, disambiguating cues can often be found elsewhere, such as the semantic context. Indeed, acoustically ambiguous phonemes can be perceptually biased toward a semantically congruent percept (e.g., "The girl milked the ?OAT," resulting in the percept GOAT). However, these findings come from well-controlled studies where participants repeatedly attend to specific contrasts produced in isolated sentences by a hyper-articulate speaker, which may limit generalizability to real-world listening tasks. Therefore, the current project aims to reconsider this research question in a more naturalistic listening scenario. A two-speaker referential communication task was used to elicit English discourse about visual

scenes. Subsequently, statements containing critical minimal-pair referents were replaced with experimental trials containing acoustic × semantic manipulations. Participants in the current study will be told to listen to the conversation and show they're paying attention by clicking referents in the on-screen image. Supposedly, their primary task will be to complete intermittent survey items about the speakers' social communication. In reality, we will use eyetracking to infer their phonological and lexical processing during the embedded trials.

5aSC21. The effect of clear speech on dual-task measures of listening effort in varied noise levels. Anushri Kartik-Narayan (Linguist, The Univ. of Texas at Austin, 305 E. 23rd St., Austin, TX 78712, anushri.kartik-narayan@utexas.edu) and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, Austin, TX)

Listener-oriented clear speech (CS) improves word recognition in noise compared to conversational speech (CO). Pupillometry studies suggest that this intelligibility benefit may result from reduced listening effort, reflecting differences in how cognitive resources are allocated during challenging listening tasks. Dual-task paradigms demonstrate validity at measuring listening effort (Brown, 2025); however, previous studies have not consistently shown a CS benefit in response times on secondary visual tasks (Meemann and Smiljanić, 2019). The current study evaluates whether a CS benefit on visual task response time, and thus listening effort, emerges across different noise levels: quiet, 0 dB signal-to-noise ratio (SNR), and −5 dB SNR. Native English listeners performed a word recognition in noise task and a visual Stroop task, first separately and then simultaneously. Preliminary results show improved word recognition and slower response time with increased noise levels for CS but not for CO. However, response time did not improve with CS at any noise level. Thus, it appears that CS does not provide a measurable benefit for dual-task measures of listening effort, suggesting that cognitive resources may not be more available for the visual task when listening to the more intelligible CS.

5aSC22. Auditory attention and speech perception in misophonia. Niah Nieuwenhuis (Speech, Lang., & Hearing Sci., Purdue Univ., 320 S Grant St., Apt 624, West Lafayette, IN 47906, nnn37@nau.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Misophonia is a disorder of decreased tolerance to specific sounds, called triggers. Previous research suggests that misophonia is grounded in fundamental differences in auditory attention that may also be implicated in speech perception. Speech perception in misophonia is extremely understudied and conveys mixed results. One hypothesis is that misophones maintain a state of hypervigilance or hyperarousal in conditions in which triggers are likely. Such heightened arousal could affect performance on complex, attention-dependent listening tasks like understanding speech in noise. However, it is not clear whether observed group-level differences are due to trait-level differences in auditory attention or context-dependent differences in arousal, or some combination. Here, we present preliminary results from an ongoing study of auditory attention and speech perception with misophones and controls. We evaluate listeners' auditory attention using the Auditory Attention Test (AAT) and perception of speech in multi-talker babble under conditions of typical versus heightened arousal (trigger threat). Self-reported and physiological measures of arousal and stress, including heart rate and electrodermal activity, are collected and analyzed alongside behavioral test scores. Findings will clarify whether misophonic "hypervigilance" reflects a clinical-level attentional alteration, how attentional differences affect speech perception, and could inform interventions targeting attentional control.

5aSC23. Native listeners more effectively utilize spatial release from masking when detecting speech in noise and reverberation. Yukina Sakamoto (Graduate School of Sci. and Technol., Sophia Univ., 7-1 Kioicho, Chiyoda-ku, Tokyo 102-8554, Japan, y-sakamoto-8r7@eagle.sophia.ac.jp), Taiki Matsubara (Graduate School of Sci. and Technol., Sophia Univ., Chiyoda-ku, Tokyo, Japan), Shinya Tsuji, Takayuki Arai (Information and Commun. Sci., Sophia Univ., Chiyoda-ku, Tokyo, Japan), Eri Osawa (Medical English Education Ctr., Showa Medical Univ., Shinagawa-ku, Tokyo, Japan), Yusuke Hioka, C. T. J. Hui (Mech. and Mechatronics Eng., Univ. of Auckland, Auckland, New Zealand), and Hinako Masuda (Seikei Univ., Musashino-shi, Tokyo, Japan)

This study examined how nativeness affects the performance of spatial release from masking (SRM) during a task detecting speech in noise under anechoic and reverberant environments. While previous research has primarily focused on speech intelligibility, the differences in SRM performance in terms of detecting speech especially between native and nonnative languages remain unclear. In the present study, native Japanese listeners performed a detection task with native (Japanese) and nonnative (English) target utterances. Experiments were conducted in anechoic and reverberant environments. The target speech was fixed at 0°, while the noise was presented from five azimuthal angles ranging from 0° to 180° in 45° steps. The results showed no significant difference between languages in the anechoic condition. However, a significant interaction was observed in the reverberant condition: the native language showed a significantly larger SRM effect than the nonnative language. Specifically, while SRM effect for the nonnative language was reduced due to reverberation, high SRM effect for the native language remained. These results suggest that the nativeness of the target speech is a critical factor for the SRM effect, specifically in reverberant environments. [Work was supported by Sophia University Special Grant for Academic Research (Research in Priority Areas).]

5aSC24. Dissociating category mapping and formant-cue sensitivity in second-language vowel perception and masking release in noise. Mingshuang Li (Dept. of Commun. Disord. and Sci., California State Univ., Northridge, CA 91330, mingshuang.li@csun.edu), Can Xu (Dept. of Speech, Lang., and Hearing Sci., Western Washington Univ., Bellingham, WA), and Chang Liu (Dept. of Speech, Lang., and Hearing Sci., Univ. of Texas at Austin, Austin, TX)

Nonnative listeners often obtain less masking release from fluctuating noise. To localize this vulnerability, we paired English vowel identification with vowel and nonspeech formant discrimination to dissociate vowel category mapping, sensitivity to formant cues in speech, and psychoacoustic sensitivity to formant-like spectral peaks. English-native listeners (EN) and Mandarin-native listeners with short U.S. residency (CNUS; mean 3.3 months) or long residency (CNUL; mean 59 months) completed (i) identification of English vowels and (ii) adaptive second-formant (F2) discrimination for English vowels and matched vowel-spectrum-shaped noises. Performance was measured in quiet, stationary noise, and 16-Hz temporally modulated noise at signal-to-noise ratios (SNRs) selected to yield comparably challenging vowel-perception conditions. Relative to EN, both Mandarin groups showed poorer vowel identification in quiet and temporally modulated noise, with reduced masking release, while identification in stationary noise was similar across groups. In contrast, F2 discrimination thresholds and masking release for both vowel and nonspeech stimuli did not differ by language background, and CNUS and CNUL performed similarly across all tasks. The dissociation suggests that reduced nonnative masking release primarily reflects less robust vowel category mapping rather than reduced formant-cue or spectral-peak sensitivity.

5aSC25. Can AXB perception tasks reliably assess children's phonetic imitation of adults? Eugene Wong (Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, wong0703@umn.edu) and Benjamin Munson (Univ. of Minnesota, Minneapolis, MN)

To assess phonetic imitation in an AXB task, listeners are presented with a target word (X) flanked by two words (A, B), with the goal of judging which of the flanking words is most similar to the target word. It is unclear whether this task is useful in assessing children's imitation of adult talkers. The spectral features in children's speech are globally more similar to those

of adult women than adult men. Hence, if the child's production was presented as a target word and the flanking words were produced by a man and a woman (as might be the case if one were assessing the effect of adult gender on children's phonetic imitation), then listeners might be biased to respond with the woman, even if the child had reliably imitated the man. The current study examines this using acoustically controlled productions. Listeners assessed the similarity of children's production of /s/-initial words to those of male and female adult talkers, where the /s/ was modified to contain a variant that exactly matches one of the two adults. Results will allow us to assess whether an AXB task can be used to assess gender effects on children's phonetic imitation.

5aSC26. Psychoacoustic distances and perceived vocal similarity. Vishwas Shetty (Elec. Eng., Univ. of California at Los Angeles, Los Angeles, CA), Yoonjeong Lee (Ming Hsieh Dept. of Elec. and Comput. Eng., USC Viterbi School of Eng., Los Angeles, CA), Abeer Alwan (Elec. Eng., Univ. of California at Los Angeles, Los Angeles, CA), and Jody Kreiman (Head/Neck Surgery & Linguist, UCLA, 1000 Veteran Ave., 31-19 Rehab Ctr., Los Angeles, CA 90095-1794, jkreiman@ucla.edu)

Studies of voice generally assume that perceptual voice spaces exist and that they reflect vocal similarity. Our previous analyzes of psychoacoustic variability in voice [e.g., Lee *et al*., *JASA* **146,** 1568–1579 (2019); Lee and Kreiman, *JASA* **151**, 3462–3472 (2022)] indicate that the lowest-order dimensions of the voice space, capturing roughly 30%–35% of total variability, reflect biologically relevant factors including physical size and arousal level. When speakers were projected into a space defined by these dimensions, they overlapped substantially. This suggests that these dimensions provide a binary distinction between "expected" voice qualities that can be ignored and those that command attention, rather than distinguishing talkers. Higher dimensions may reflect other factors, including information about individual voice quality. To test this hypothesis, we used a sort-and-rate task to gather dissimilarity judgments for sets of normal male and female voices. Across listeners, all pairs of speakers were compared, producing complete pairwise perceptual dissimilarity matrices. Multidimensional scaling analyzes were used to derive perceptual spaces from these dissimilarity ratings, and those spaces were compared to the lower (shared) and higher (individual?) dimensions of spaces derived from analyzes of psychoacoustic variability. Analyses will determine the extent to which shared versus individual vocal characteristics predict perceived similarity.

5aSC27. English speakers learn to use F1 as a place cue in perception of Arabic post-velar consonants. Annika Herlant (Linguist, The Univ. of North Carolina, Linguist, 3155, Chapel Hill, NC 27599, aherlant@unc.edu)

Native Arabic speakers use F1 to discriminate uvular and pharyngeal consonant contrasts (Alwan, 1989). However, F1 is not a primary place cue in English; listeners rely instead on F2 (Delattre *et al*., 1955). This study tests the revised Speech Learning Model (SLM-r) (Flege and Bohn, 2021) and Perceptual Assimilation Model of Second Language Speech Learning (PAM-L2) (Best and Tyler, 2007), which predict that learners can learn to discriminate new contrasts and will become more native-like over time. Native Arabic speakers, beginner Arabic learners (<4 months of study), and advanced learners (>1 year of study) listened to synthetic stimuli with manipulated F1 or F2 onsets, replicated from Alwan (1989), and provided forced-choice judgements of the consonant they heard (/ʕ, ʁ, ʔ/). Preliminary results (n = 24) show that all groups weigh F1 significantly more heavily than F2, demonstrating that English speakers can learn to disregard F2 and use F1. Results also suggest that advanced speakers have learned to use more native-like perception, as they pattern more like native speakers than do beginners. Interestingly, participants who learned French or Hindi–Urdu, which have post-velars, weighed F1 heavier (and more native-like) than learners with Spanish as an L2; this suggests an effect of L2 on L3.

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